

Configure Your VoiceExpress

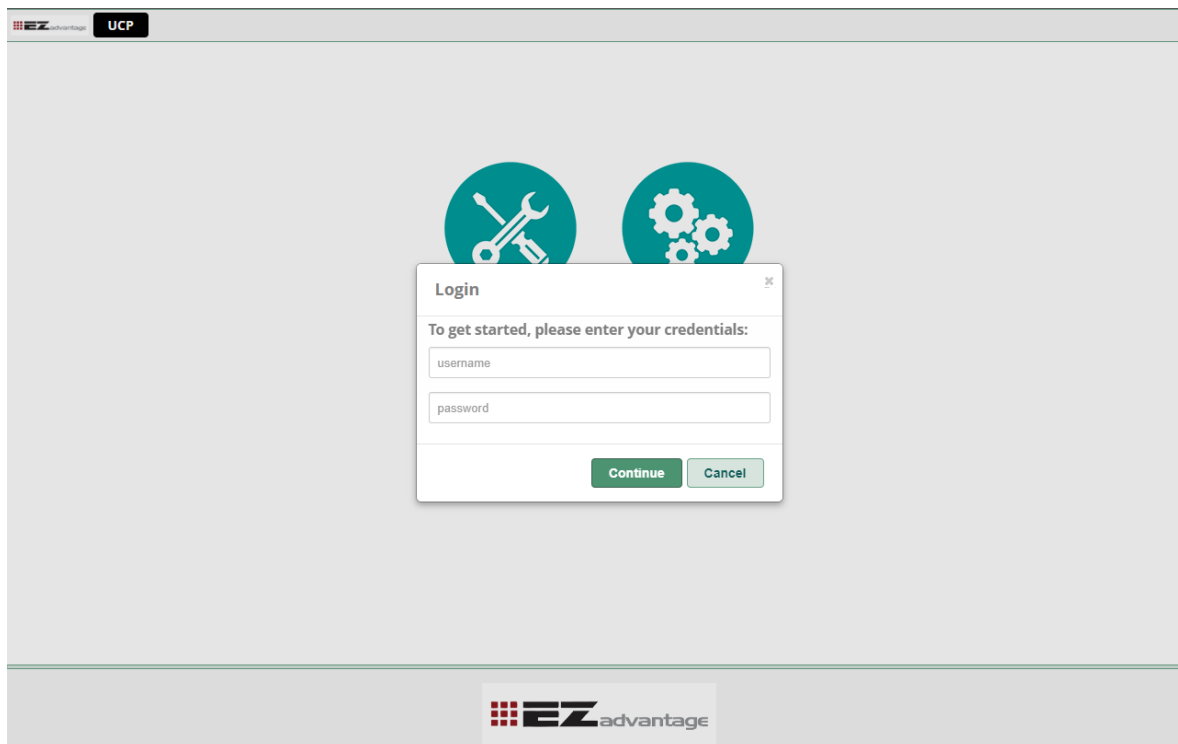
We are going to walk through a simple and typical setup of Voice Express. It would be impossible to teach you through a single wiki how to configure the many features of Voice Express, but following should allow a brand-new user to create a basic system setup. Please note, we try to keep these articles as up-to-date as possible, but your system may vary slightly from the procedures and screenshots shown here, based on the versions installed.

Configuration of the Voice Express is done using the various Voice Express Modules.

Login to the Voice Express Graphical User Interface ("GUI")

- Using another machine on your same network, open a web browser and enter the IP address of your Voice Express.
 - If you don't know the IP address of your Voice Express, go to the Linux console/command prompt. Login to the Linux console using the username "root" without quotes, and the root password you selected during installation. You will then be shown your IP address.

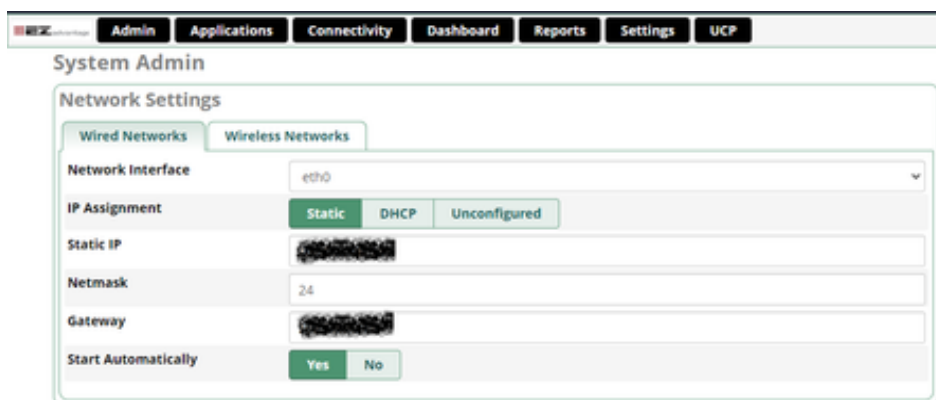
Interface	MAC Address	IP Addresses
eth0	00:0c:29:60:a5:c7	202.59.208.113 fe80::20c:29ff:fe60:a5c7



Sysadmin Module Setup

Set a Static IP Address and Configure DNS

- When Voice Express is first installed, it is configured to obtain an IP Address using DHCP. You'll need to assign your Voice Express a static IP address so that your phones will have a consistent internal IP address to use to contact it. The easiest way to give your Voice Express a static IP address is to configure your DHCP server to always assign your Voice Express the same IP address using a DHCP reservation. This ensures that if DHCP settings later change, such as changing DNS server settings, the Voice Express will get these new settings with all other DHCP clients. If that's not possible, you will need to configure your Voice Express to use a static IP and not use DHCP.
- From Voice Express GUI Click Admin, and then System Admin Module on the left hand side of the screen, and then Network Settings, on the right hand side of the screen. You should now see a screen that looks like this:




The screenshot shows the 'System Admin' interface with a navigation bar at the top containing 'Admin', 'Applications', 'Connectivity', 'Dashboard', 'Reports', 'Settings', and 'UCP'. Below the navigation bar is the 'System Admin' section, which includes a 'Network Settings' panel. The 'Network Settings' panel has two tabs: 'Wired Networks' (selected) and 'Wireless Networks'. The 'Wired Networks' tab shows a configuration form for a network interface named 'eth0'. The 'IP Assignment' section has three radio buttons: 'Static' (selected), 'DHCP', and 'Unconfigured'. Below this, there are input fields for 'Static IP', 'Netmask' (set to '24'), and 'Gateway'. The 'Start Automatically' section has two radio buttons: 'Yes' (selected) and 'No'.

- Change IP Protocol to "None" and then enter your desired static IP address. Be sure to also set your subnet mask (typically 255.255.255.0) and default gateway (usually 192.168.1.1).
- When you're done, click save settings.
- To continue, input the new IP address into your web browser. Return to the System Admin Module (follow the instructions described above) and manually set your DNS Servers
- Then go to the DNS section of the System Admin module and click DNS on the right hand side of the screen. You'll see a screen that looks like this:

System Admin

DNS

DNS Server list 

```
127.0.0.1
8.8.8.8
8.8.4.4
192.168.1.1
```

You can replace those two with your own, if you prefer.

When you're done, click "Submit"

Setting the Time Zone

- - Go to Admin→System Admin→Time Zone
 - Use the drop-down menus to select the appropriate time zone and then click Submit. Note the warning stating that you must reboot your system in order to complete the changes.

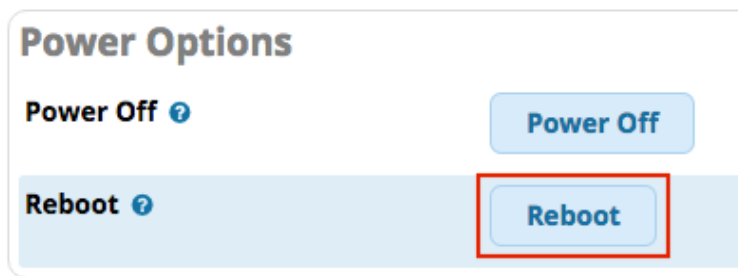
System Admin

Time Zone

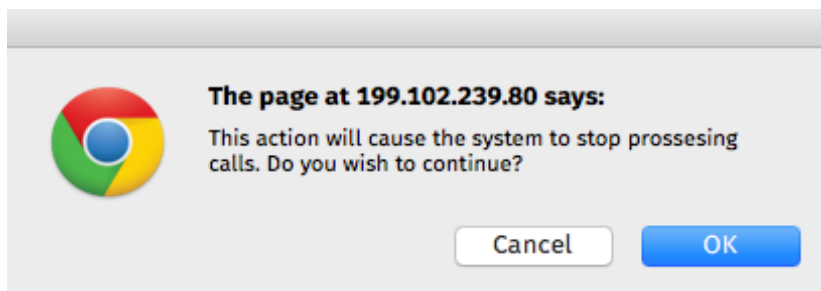
Warning: To complete timzeone changes, you must reboot your system.

Time Zone

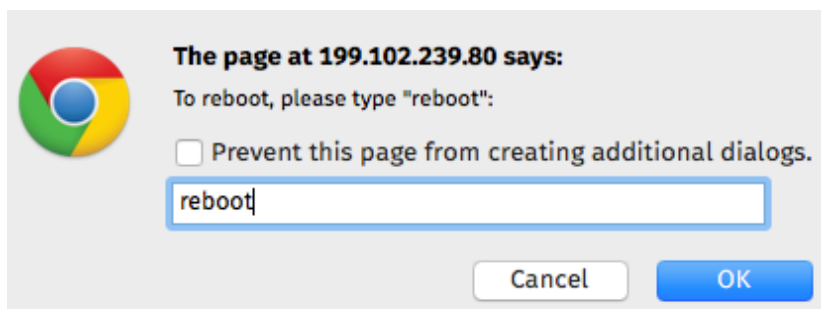
- Go to Admin → System Admin → Power Options
- Be very careful with this section! You will want to reboot, NOT power off.
- Click the Reboot button



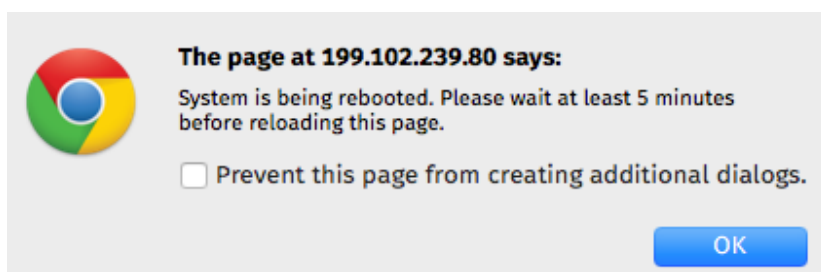
You'll be warned that rebooting will cause call processing to stop. Click OK if you are ready to continue.



Just to make sure you're really ready to reboot, a pop-up will ask you to type "reboot" in the field. Do this, then click OK.



Your system will now reboot. Click OK to close the information window, and wait at least 5 minutes before reloading the page.



Go to Admin → System Admin → Time Zone again, and you should see your newly chosen timezone displayed.

Configuring Intrusion Detection

- Go to Admin -> System Admin -> Intrusion Detection
- We recommend that you keep this service running, in order to detect, block, and notify you of attempts to compromise your system. You may check the status here and adjust your ban time, max retry, and find time if needed.
- Enter an e-mail address where you would like to receive intrusion detection notifications.
- Optionally, enter any IPs that you would like to whitelist.
- Click Submit. Your settings will be applied when the page reloads.

Status	running
Intrusion Detection	Stop Restart
Ban Time	<input type="text" value="1800"/>
Max Retry	<input type="text" value="8"/>
Find Time	<input type="text" value="600"/>
E-mail:	<input type="text" value="youremail@example.com"/>
Whitelist	<input type="text" value="127.0.0.1"/>
IP's that are currently banned.	
No Banned IP's	

Extensions Module - PJSIP Extension

Overview

This guide walks you through information related to PJSIP extensions.

In order to have access to creating PJSIP extensions

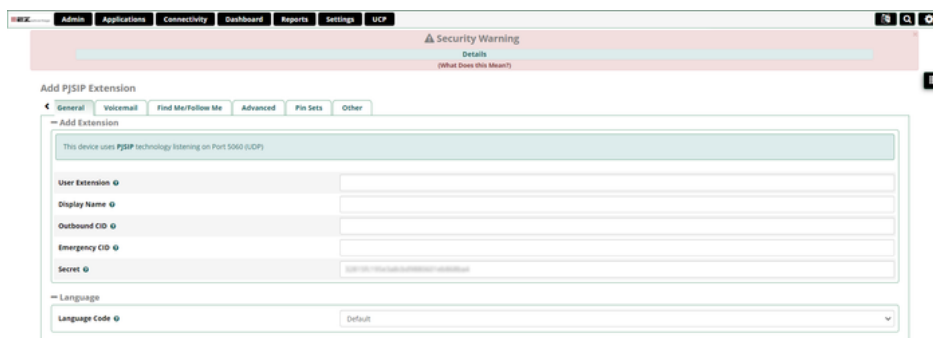
- From the top menu click Applications
- From the drop down click Extensions

Adding a PJSIP Extension

Click the Add New PJSIP Extension button.



General



Add Extension

User Extension

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

Display Name

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

Outbound CID

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

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

Secret

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

User Manager Settings

Voicemail

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

The screenshot shows the Asterisk Manager GUI interface for configuring a PJSIP extension's voicemail settings. The 'Voicemail' tab is selected, and the 'Enabled' checkbox is checked. The 'Voicemail Password' field is empty. Other options include 'Require From Same Extension', 'Disable (*) in Voicemail Menu', 'Email Address', 'Pager Email Address', 'Email Attachment', 'Play CID', 'Play Envelope', 'Delete Voicemail', 'VM Options', and 'VM Context'.

Field	Value
Enabled	Yes
Voicemail Password	
Require From Same Extension	Yes
Disable (*) in Voicemail Menu	Yes
Email Address	
Pager Email Address	
Email Attachment	Yes
Play CID	Yes
Play Envelope	Yes
Delete Voicemail	Yes
VM Options	
VM Context	default

Voicemail

Enabled

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

Voicemail Password

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

Require From Same Extension

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

Disable (*) in Voicemail Menu

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

Email Address

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

Pager Email Address

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

Email Attachment

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

Play CID

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

Play Envelope

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

Delete Voicemail

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

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

VM Settings

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

VM Context

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

VMX Locator™

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

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

VMX Locator™

Enabled Yes No

Use When: Unavailable Busy Temporary

Voicemail Instructions: Yes No

Press 0: Go To Operator

Press 1:

Press 2:

EZ advantage

Enabled

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

Use When

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

Voicemail Instructions

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

Press 0

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

Press 1

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

Press 2

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

Saving the Extension

- Click the Submit button
- Click the Apply Config button

Bulk Handler User Guide (Add bulk Extension)

Bulk Handler manages the bulk export or import of extensions, DIDs, user manager users, user manager groups, and contacts. You can export any of these as a CSV file. You can also upload a CSV file to save time versus having to enter each item individually. The module provides required/recommended headers for your CSV files.

Navigating to Bulk Handler

1. Log into your Voice Express GUI
2. In the top menu go to **Admin**
3. In the drop-down menu go to **Bulk Handler**

You will land in the "Export" section with "Extensions" selected by default.



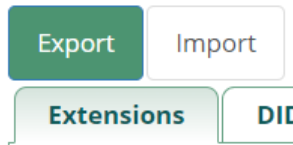
Export a CSV File

The **Export** section allows you to export a CSV file of Extensions, DIDs, User Manager Users, User Manager Groups, or Contacts. You can then make additions, removals, and changes to the file and import it if desired.

Exporting is also a way to create a "template" to ensure you are using all of the available headers if you plan to import data.

1. Click the **Export** button at the top if it is not already selected. (Dark background = selected)

Bulk Handler



2. Click the tab for the type of information you want to export.
In our example we'll export Extensions.



3. Click the **Export** button near "CSV File." Your browser will download a .csv file that you can open in a spreadsheet application.



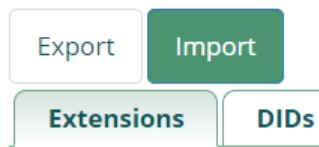
Import a CSV File

Tip

Perform an export first, if you would like to see all of the available headers and edit the file from there. You can also create a CSV file from scratch. The "Import" section will show you the minimum required/recommended headers.

1. Click the **Import** button at the top if it is not already selected. (Dark blue background = selected)

Bulk Handler



- 2.

Click the tab for the type of information you want to import. In our example, we'll be importing extensions, but the general process below applies to any kind of import.



3. If you are creating a CSV file from scratch, note the **Required/Recommended Headers:**

If you are importing...

Required/Recommended Headers

Extensions

```
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),
```

DIDs

```
description (Description),
extension (Incoming DID),
cidnum (Caller ID Number),
destination (The context, extension, priority to go to when this DID
is matched. Example: app-daynight,0,1),
fax_enable (Fax Enabled),
fax_detection (Type of fax detection to use (e.g. SIP or DAHDI)),
fax_detectionwait (How long to wait and try to detect fax),
fax_destination (Where to send the faxes),
```

User Manager Users

```
username (Login Name),
password (Password - plaintext),
default_extension (Primary Linked Extension),
description (Description),
fname (First Name),
lname (Last Name),
displayname (Display Name),
```

User Manager Groups

```
groupname (Group Name),
description (Description),
```

Contacts

```
groupname (Name of group for contact. If group does not exist, it
will be created.),
grouptype (Type of group for contact.),
```

```
displayname (Display Name),
fname (First Name),
lname (Last Name),
title (Title),
company (Company),
address (Address),
userman_username (User Manager username this contact should point to.
Internal contacts only.),
phone_1_number (Phone number. External contacts only.),
phone_1_type (Type of phone number. External contacts only.),
phone_1_extension (Extension. External contacts only.),
phone_1_flags (Comma-delimited list of flags. (Example: sms,fax)
External contacts only.),
phone_2_number (Phone number. External contacts only.),
phone_2_type (Type of phone number. External contacts only.),
phone_2_extension (Extension. External contacts only.),
phone_2_flags (Comma-delimited list of flags. (Example: sms,fax)
External contacts only.),
phone_3_number (Phone number. External contacts only.),
phone_3_type (Type of phone number. External contacts only.),
phone_3_extension (Extension. External contacts only.),
phone_3_flags (Comma-delimited list of flags. (Example: sms,fax)
External contacts only.),
email_1 (E-mail address. External contacts only.),
email_2 (E-mail address. External contacts only.),
email_3 (E-mail address. External contacts only.),
```

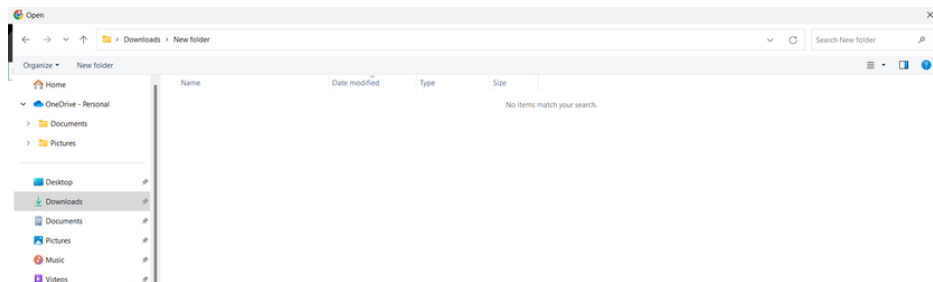
4. Click the **Browse** button.

CSV File 

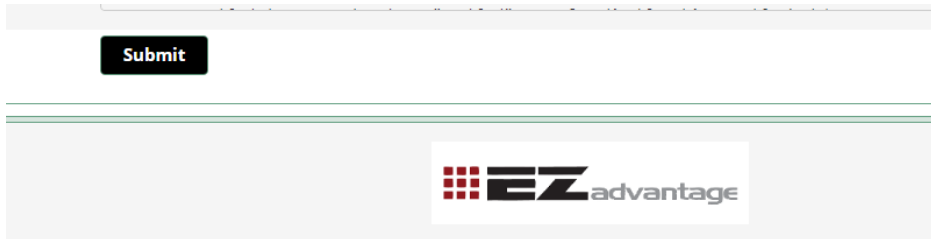
Browse

File to Import

5. Select your .csv file from your computer. Your selected filename will appear next to the browse button.



6. Click the **Submit** button at the bottom. You're not done yet -- this will take you to a data validation page.



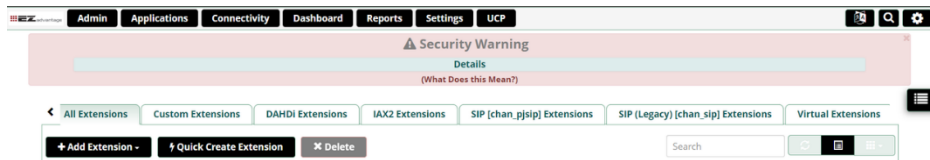
7. At the top, use the **Yes/No** toggle button to indicate whether you want to replace and update all of your existing data with the contents from the CSV file.
8. The contents of your CSV file are displayed in the table, but only a few main headers are shown.
9. Optional: You can edit the details of each item at this time before completing the import. You will be able to edit all headers, not just the ones shown in the table.
 1. Click the edit button for any item you want to edit.
 2. A window will pop up for this item. Note that you can scroll in the window to see all fields.
 3. Click the **Close** button if you are not making any changes. Otherwise, make your changes in the appropriate fields and click the **Save changes** button.
10. Wait while the import is performed. The status bar near the top will show the progress of your import. The rows of the table will also turn green as each item is imported.
11. Click the **Finished** button when done.

Click the Apply Config button.

Extensions - Editing or Deleting

Overview

Existing extensions appear in a table on the Extensions Module landing page. You can click on an action for the extension to edit or delete it.



- From the top menu Click Applications
- In the drop down click Extensions

Viewing/Editing an Extension

- Click on the pencil icon :edit: for the extension you wish to view.
- Make the desired changes. For explanations of each type of extension,
- Click the Submit button.
- Click the Apply Config button.

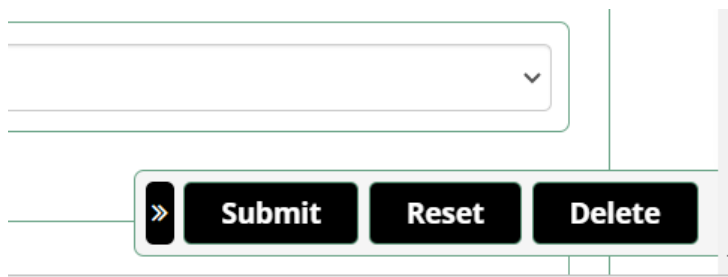
Deleting an Extension

From the extensions list:

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

Alternatively, if already viewing the extension:

- Click the Delete button.



- In the alert window that appears, click the OK button to confirm the deletion.
- Click the Apply Config button.

Outbound Routes

Overview

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

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

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

Important

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

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



















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

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

Outbound Routes

This page is used to manage your outbound routing.

+ Add Outbound Route

Name	Outbound CID	Attributes	Actions
+ E911-Leave-First		   	 
+ SIPStation-Out		   	 
+ SIPStation-INT		   	 

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

Name

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

Outbound CID

The outbound caller ID for this route.

Attributes

- Green = Yes
- Gray = No

= Emergency Route

= Intra-Company Route

= Password-Protected

= Time Group Assigned

Actions



= View or Edit



= Delete

Adding an Outbound Route

Click the Add Outbound Route button.

Route Settings Tab

Route Name

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

Route CID

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

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

Override Extension

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

Route Password

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

Route Type

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

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

Music On Hold

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

Time Group

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

Route Position

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

Trunk Sequence for Matched Routes

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

Route Position

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

Trunk Sequence for Matched Routes

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

Select one or more trunks from the drop-down menus. You can also change the order of trunks by dragging and dropping the routes using the arrow icon



. The top route will be tried first, followed by the next route down, and so forth.

Optional Destination on Congestion

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

Dial Patterns Tab

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

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

The format is:

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

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

PATTERN	DESCRIPTION
X	Any whole number from 0-9
Z	Any whole number from 1-9

N	Any whole number from 2-9
[###]	Any whole number in the brackets, example [123] is 1 OR 2 OR 3. Note that multiple numbers can be separated by commas and ranges of numbers can be specified with a dash ([1.3.6-8]) would match the numbers 1,3,6,7 and 8.
.(dot)	It matches one or more characters and (acts as a wildcard)

Prepend

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

Prefix

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

Match Pattern

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

CallerID

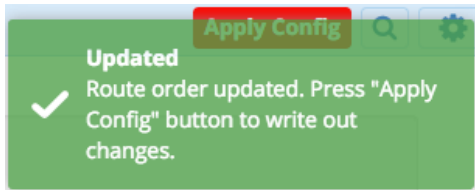
If caller ID is supplied, the dialed number will only match the prefix + match pattern if the caller ID being transmitted matches this. When extensions make outbound calls, the caller ID will be their extension number and NOT their outbound CID. The above special matching sequences can be used for caller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

Changing the Order of Outbound Routes

Remember, the system searches for a matching dial pattern by starting with the top route and working its way down. If a match is found, the system does not continue going down the list looking for a "better" route. Therefore, route order is important, especially if there is some overlap. For example, the number 555551212 will match both a dial pattern of 55555XXXX and NXXNXXXXXX.

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

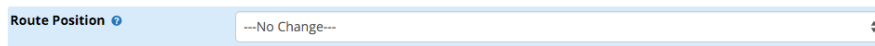
- Simply drag and drop. Hold the mouse button down over the arrow symbol or the route's name, and drag the entire line to a new position in the list.
- A pop-up notification will appear to let you know you've changed the order.



Click the Apply Config button to apply the changes.

B) To change the order of a route when creating/editing it:

In the Route Settings tab, you can use the Route Position tab to select a new position for a route.



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

Editing or Deleting an Outbound Route

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

Inbound Route

Overview

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

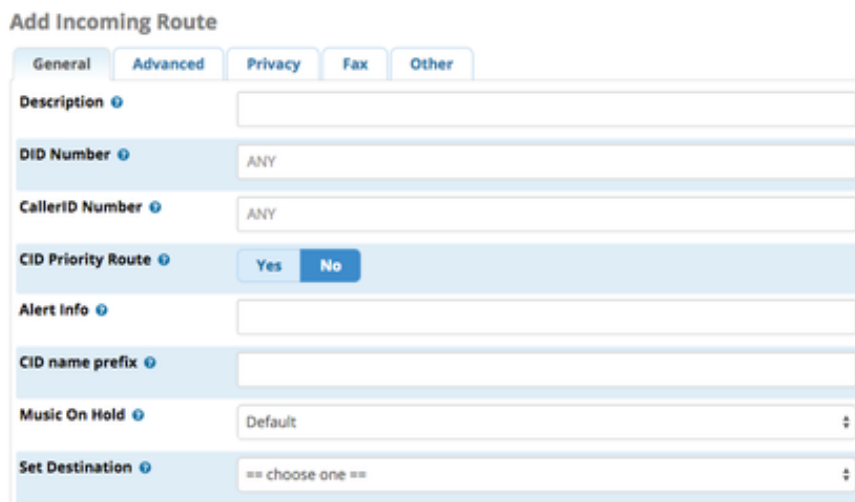
Settings depend on installed modules. You may have more settings than are shown here, or settings may be missing.

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

Adding an Inbound Route

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

General



The screenshot shows the 'Add Incoming Route' configuration form with the 'General' tab selected. The form contains the following fields and options:

- Description:** A text input field.
- DID Number:** A dropdown menu with 'ANY' selected.
- CallerID Number:** A text input field with 'ANY' entered.
- CID Priority Route:** Radio buttons for 'Yes' and 'No', with 'No' selected.
- Alert Info:** A text input field.
- CID name prefix:** A text input field.
- Music On Hold:** A dropdown menu with 'Default' selected.
- Set Destination:** A dropdown menu with '== choose one ==' selected.

Description

Enter a unique description for the route.

DID (Direct Inward Dialing) Number

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

CID (Caller ID) Number

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

CID Priority Route

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

The default priority levels are matched in the following sequence:

With CID Priority Route disabled:

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

With CID Priority Route enabled:

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

Alert Info

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

CID name prefix

This allows text to be prepended to the caller ID name information from the call. This is often used to identify where a call came from. For example, a number dedicated for sales might be prefixed with "Sales:." A call from John Doe would display as, "Sales:John Doe."

Music On Hold

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

Set Destination

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

Advanced

Add Incoming Route

General	Advanced	Privacy	Fax	Other
Signal RINGING ⓘ	<input type="radio"/> Yes <input checked="" type="radio"/> No			
Reject Reverse Charges ⓘ	<input type="radio"/> Yes <input checked="" type="radio"/> No			
Pause Before Answer ⓘ	<input type="text"/>			

Signal RINGING

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

Reject Reverse Charges

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

Pause Before Answer

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

Privacy

Add Incoming Route

General	Advanced	Privacy	Fax	Other
Privacy Manager				
<input type="radio"/> Yes <input checked="" type="radio"/> No				
Max attempts				
<input type="text" value="3"/>				
Min Length				
<input type="text" value="10"/>				

Privacy Manager

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

Max attempts

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

Min Length

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

Fax

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

Add Incoming Route

General	Advanced	Privacy	Fax	Other
Detect Faxes				
<input checked="" type="radio"/> No <input type="radio"/> Yes				

Detect Faxes

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

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

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

Fax Detection type

Type of fax detection to use.

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

Fax Detection Time

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

Fax Destination


Where to send the faxes.



Other



Add Incoming Route


General Advanced Privacy Fax Other



Note that the meaning of these options has changed. Please read the wiki for further information on these changes.

Call Recording  Force Yes **Don't Care** No Never

CID Lookup Source  None 

Language  Default 

Enable Superfecta Lookup  Yes **No**

Superfecta Scheme  ALL 

Call Recording

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

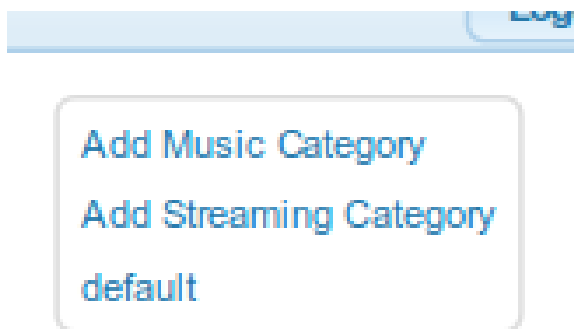
Music on Hold User Guide

Overview

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

Adding Music Categories

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



You will be brought to the following

On Hold Music

Add Music Category

Category Name:

[Submit Changes](#)

Category Name

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

Adding Static Music

In the right side navigation menu click the category you wish to add music to.



Upload a WAV or MP3 File

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

You will click Choose File

You will then be prompted to specify the location on your local computer where the sound file is located. Depending on your browser, you may be able to drag and drop the file directly to the "Choose File" box.

Volume Adjustment

Here you can increase or decrease the volume of this specific MoH file before uploading it. Once the file has been uploaded, you cannot make changes to the volume without deleting the file and uploading it again with the adjusted levels.

Upload

Press this button to actually upload your MoH file. If you're uploading an MP3 file, you will need to wait for the page to completely load after submitting because the system is converting the MP3 to WAV format.

Disable/Enable Random Play

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

MoH List

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

Setting Up a Ring Group

A ring group is a list of multiple numbers that you would like to ring when a call is received. This gives several phones the opportunity to answer a call. You can choose from various ring strategies to control the order in which the system rings different phones.

- Go to Applications -> Ring Groups
- Click Add Ring Group.

Enter a Ring-Group Number. This will be the number a user may dial to ring the phones in the ring group. Be sure it does not match any existing extensions or feature codes. The default is 600.

Enter a Ring Group Description. This can be a "friendly name" and spaces are acceptable.

Choose a Ring Strategy. The default is "ringall." Other options include ringall-prim, hunt, hunt-prim, memoryhunt, memoryhunt-prim, firstavailable, firstnotonphone, and random. Click on the pop-up tooltip question mark icon for a full description of each of these strategies.

Set the Ring Time, in seconds (default=20, maximum=300 seconds).

Add extensions and/or outside numbers to the Extension List, one per line. You can include an extension on a remote system, or an external number, by suffixing the number with a "#" symbol. You can use the Agent Quick Select menu to quickly select existing extensions. Your ring group will call the numbers in the extension list.

In our example, we've added two internal extensions and an external phone number.

At the bottom of the page, set a destination to which to route the inbound call in case no one answers. For example, you may wish to route it to a voicemail box.

- Click the Submit button and then click the Apply Config button.
 - In order for the ring group to work, you'll need to give callers the option to dial it (i.e. from a company directory or IVR), or you'll need to route inbound calls to a DID or extension directly to the ring group. These options are controlled within the Inbound Routes module and elsewhere in other modules.

For example, using the Inbound Routes module, we can set the destination of our DID 9203831234 to be "Ring Groups" and select the ring group we just created. If we click "Submit" and then "Apply Config," all calls coming into 9203831234 will now ring the numbers in our ring group.

Setting Up a Queue

You can set up automatic call distribution with the Queues module. A queue differs from a ring group because it allows advanced call routing options and escalation rules. This wiki gives you a brief overview of some basic settings, but there are many more. To learn more about queue settings, you can use the pop-up tooltips in the module.

- Go to Applications -> Queues
- Click the Add Queue button. You should now be viewing the General Settings tab.
- Enter a Queue Number. This number will be used to dial into the queue or to transfer callers to the queue. Be sure it does not match any existing extensions or feature codes.

Enter a Queue Name that will help you identify this Queue. This can be a "friendly name," and spaces are allowed.

Optionally, set a Queue Password (numbers only). This would require agents to enter a password before being allowed to log into the queue.

Choose whether to generate device hints, confirm calls, prefix the caller ID, give a wait time prefix, or use Alert Info. See the pop-up tooltips for more info.

Choose how you would like to handle agent restrictions. If you set Restrict Dynamic Agents to Yes, you can prevent unauthorized users from logging into the queue. Later, you will define a list of users who are allowed to log in. If this is set to No, any user who knows the queue number (and password, if set) would be able to log in as a dynamic agent. Next to Agent Restrictions, choose how you would like the queue to handle Call Forwarding / Follow-Me settings.

You can configure several more options at the bottom of the General Settings section. These include autofill, skip busy agents, queue weight, music on hold class, join announcement, call recording, and whether to mark calls answered elsewhere.

At the bottom of the General Settings section, choose a Fail Over Destination. This is where callers will be sent in case the queue is over capacity, the caller reaches wait time limits with no one answering the call, etc.

- Click on the Queue Agents tab. Here, you will create lists of agents who can answer queue calls. The Agent Quick Select menus help you find existing extensions. You can manually enter extensions or outside telephone numbers.
 - Static Agents are assumed to always be available to answer queue calls. They do not need to log in or log out of the queue.
 - Dynamic Agents are additional agents who can choose to log into or log out of the queue. They are not automatically available. They must log in to the queue in order to participate. To log in, a dynamic agent would dial the queue number plus "*" (for example, 123* if the queue number is 123). To log out, they would dial the queue number plus "***" (for example, 123**).

- Click on the Queue Agents tab. Here, you will create lists of agents who can answer queue calls. The Agent Quick Select menus help you find existing extensions. You can manually enter extensions or outside telephone numbers.
 - Static Agents are assumed to always be available to answer queue calls. They do not need to log in or log out of the queue.
 - Dynamic Agents are additional agents who can choose to log into or log out of the queue. They are not automatically available. They must log in to the queue in order to participate. To log in, a dynamic agent would dial the queue number plus "*" (for example, 123* if the queue number is 123). To log out, they would dial the queue number plus "***" (for example, 123**).

Click on the Timing & Agent Options tab to configure several settings related to wait time, timeout, retry, agent pause, and more. The default values are shown below.

Click on the Capacity Options tab to define how calls are sent into the queue vs. sent to an alternate destination. If you would like to limit the queue to a certain maximum number of callers, you can set that here. ("0" means "unlimited.") Join and Leave options control the behavior of the queue when it appears that agents might not be able to take the call.

Click on the Caller Announcements tab to set up announcements that callers will hear. You can choose whether the system should tell callers their position in line and/or their estimated hold time. You can also give callers an option to use an IVR break out menu or request a queue callback.

Click the Submit button when finished, then click the Apply Config button.

Creating System Recordings

Modules such as Interactive Voice Response (IVR) are able to use custom system recordings in addition to default recordings. You can use the System Recordings module to create and save custom system recordings.

There are two ways you can create a system recording: by speaking over the phone, or by uploading an audio file from your computer.

After you have created a recording, you can update it by uploading a replacement audio file or by making a new recording over the phone. You can make updates in the System Recordings module and/or enable users to dial a specific feature code that will allow them to re-record over the phone.

Creating a System Recording Over the Phone

- Go to Admin -> System Recordings
- Step 1 (a): Enter the extension number of the phone that you want to use to make your recording, and click Go.

Step 1 (b): The page will reload and display instructions that give you a code to dial. Dial this code from the same extension you entered above. You will hear a beep. Begin speaking after the beep, and press # when finished. You will be prompted to listen to your recording to or re-record it. When you are satisfied with your recording, simply hang up.

Step 2: Now, dial *99 to verify your recording. You will hear the tone play again, followed by your message. You can re-record from here if you would like, or simply hang up again.

Step 3: When you are satisfied with the recording, enter a name for the recording (no spaces) and click Save.

The page will reload and you will see a message letting you know your recording has been saved.

Note: There is no Apply Config button in this case. You should see your new recording show up in the list of recordings.

Uploading an Audio File from your Computer

- Go to Admin -> System Recordings

- Step 1: Choose a sound file from your computer. The file must be an Asterisk-supported format. Click Choose File, select a file from your computer, then click Upload.
 - A window will pop up asking you to wait until the page reloads. Don't let this confuse you; please click OK right away, and then wait for the page to reload.

After a successful file upload, you will see a message confirming the upload:

Step 2: Dial *99 to verify your recording. You will hear the tone play again, followed by your message. Note: if you are not satisfied with the recording, you should repeat step 1 and upload a new file from your computer.

Step 3: The Name this Recording field will automatically populate with the name of your file. You can accept the automatically generated name or enter a new name. Click Save when you are ready to save the recording.

The page will reload and you will see a message confirming that your recording has been saved.

Note: There is no Apply Config button in this case. You should see your new recording show up in the list of recordings.

Enabling Users to Re-Record Existing System Recordings Over the Phone

You can enable a link to a feature code that will allow users to re-record a system recording over the phone. When users dial the specific feature code, they can re-record the message directly from their phone without visiting the System Recordings module. This gives users the opportunity to change recordings without contacting a system administrator.

- Go to Admin -> System Recordings
- Click on the name of the desired recording in the list.
- Check the box next to Link to Feature Code. The system will display the feature code that you can use to re-record this particular recording.

In our example, the feature code is *294. A unique feature code will be generated for each of your recordings.

Click Save and then click Apply Config. You will now be able to dial the feature code in order to listen to and re-record the system recording. After dialing the feature code, you will hear a beep followed by a playback of the current recording. You will then be given the option to re-record the message.

Setting Up an Interactive Voice Response (IVR) System

An Interactive Voice Response (IVR) system gives a caller the option to choose from several destinations. A caller uses a touch-tone (DTMF) telephone to choose from menu options.

Before configuring your IVR, you will need to set up system recordings that will give instructions to the caller. You will also need to set up the destinations you plan to use with the IVR, such as extensions, ring groups, queues, voicemail boxes, directories, other IVRs, etc.

Your IVR system can be simple, with just a few options, or it can be complex with multiple options and even multiple IVRs. As a best practice, we recommend giving callers only a few options at a time, and using multiple IVRs to “nest” or “layer” various menus. For example, a single IVR might lead to three different destinations with #4 being another IVR, which then gives a caller more choices.

Always be sure that your message to callers matches the options that are set up in the IVR. If you change your IVR options, don't forget to change your recording!

An IVR may contain "hidden" options. In other words, your recorded announcement would not tell a caller that the option exists, but a caller could dial the option. For example, you could set up a "hidden" option for VIP users to receive queue priority or for employees to dial their voicemail from an outside number.

In case the caller does not enter anything, or makes an invalid entry, the IVR can send the caller to an alternate destination (the "invalid destination"). Always set up a destination for the caller to reach in case he/she reaches the maximum number of invalid entries or fails to make an entry.

- Go to Applications -> IVR
- Click the Add IVR button
- Under IVR General Options, enter an IVR Name and IVR Description (both can contain spaces).

Use the Announcement drop-down menu to choose the system recording you would like to play to callers when they arrive at the IVR. The default option is “None.” A setting of “None” would not be advised, since it would not explain any options to the caller. (Please see visit the System Recordings module at Admin -> System Recordings if you need to set up recordings.)

- Use the Direct Dial drop-down menu to choose whether you would like callers to be able to direct-dial an extension from your IVR. It is disabled by default. Enabling

direct-dial saves your callers time if they already know a person's extension and do not wish to navigate through an IVR. You can enable direct-dialing for all extensions or restrict it to a certain directory.

- Disabled: Users can only dial the options you define in your IVR. Anything else is considered an invalid entry.
- Enabled: Users can dial your IVR options as well as any valid extension on the system.
- Specific Directory (Name of Directory): If you have created a directory, you will see it here in the menu. If you select it, you will limit callers to dialing your IVR options and any extensions included in the directory. Callers will not be able to direct-dial extensions that are not included in the directory.

Set the number of seconds you want to be considered a Timeout. If no DTMF tones are heard during this time period, the system will take the actions you specify in your timeout settings. Other timeout options are found further down the page.

- Configure your invalid entry settings. If the caller dials something that isn't one of the choices in your IVR, or tries to dial an extension for which direct dialing is disabled, the system considers it an invalid entry.
 - Invalid Retries: Number of chances you give a caller to make a correct entry before sending them to the invalid destination.
 - Invalid Retry Recording: Message played to the caller letting them know they have made an invalid entry, and asking them to try again. You can set up custom recordings if desired.
 - Append Announcement to Invalid: Whether to replay the main IVR announcement to a caller who makes an invalid entry.
 - Return on Invalid: Return to the parent IVR if it was called from an IVR.
 - Invalid Recording: Message played to the caller informing them that they have made too many invalid entries and will be sent to an alternate destination.
 - Invalid Destination: Where to send callers who have reached the maximum number of invalid retries. You are required to set an Invalid Destination. This is somewhat of an "if all else fails" option for customers who may be having difficulty navigating the system. For example, you may wish to send callers to an operator or a voicemail box.

- Configure your timeout options.
 - Timeout Retries: Number of times to retry when no DTMF is heard.
 - Timeout Retry Recording: Message to play to a caller when no DTMF is heard, prompting caller to try again.
 - Append Announcement on Timeout: Whether to replay the IVR announcement when no DTMF is heard.
 - Return on Timeout: Whether to return to a parent IVR if it was called from an IVR.

- Timeout Recording: Message to play to a caller when no DTMF is heard and caller has reached the maximum number of timeout retries.
- Timeout Destination: Where to send a caller who has reached the maximum number of timeout retries.
- Return to IVR after VM: If the caller reaches a voicemail box after being sent to a timeout destination, whether to return the caller to the IVR.

Set up your IVR entries. These are the options that callers can dial. The Ext field is the number the caller needs to dial in order to reach the specified destination. For example, if your IVR announcement tells callers to "Press 1 for sales," you would enter "1" in this field. Choose the Destination from the drop-down menu.

Click the blue plus sign to add additional entries.

- When done, click the Submit button and then click the Apply Config button.
 - There is one final step you must take in order to make your IVR work for you. You will need to either set the IVR as a destination on an inbound route, or set it as a destination in one of your other modules (i.e. Call Flow Control, Time Conditions, etc.). This will enable callers to reach your IVR.

Setting Up a Time Group

A Time Group defines periods of time that a module such as Time Conditions or Outbound Routes will use. For example, you might create a time group that defines your open business hours.

Before setting up a time group, be sure you have set the correct system time zone by visiting Admin → System Admin → Time Zone. If you need to adjust your time zone, follow the instructions given earlier in this wiki. Times in the time group are local to your system time zone. They use a 24-hour clock.

- Go to Applications → Time Groups.
- Check that the server time displayed is correct. If not, adjust the time zone as explained at the beginning of this wiki. You may need to reboot your system.
- Click Add Time Group.
- Enter a Description for your time group. In our example, we will be creating a list of times that our business is open.

Define your set(s) of start and finish times, days of the week, days of the month, and months as necessary. If you need to define more than one set of times, click Add Time.

In our example, we have first selected Monday through Friday, 8 AM to 5 PM. Then we clicked Add Time and added a second selection, Saturdays 8 AM to noon.

- After you have made your time selections, click the Submit button and then click the Apply Config button.
- Your new time group will show up in the time group list.

Excluding Specific Time Periods, Such as Holidays

If you need to define an exception to your time group, such as a holiday, the trick is to leave the “Time to start” and “Time to finish” drop-down menus blank. This tells the system you do not want to include any time period within this particular day, week, month, etc.

For example, here is how you would exclude Christmas Day from your time group:

As another example, if you would like to exclude Thanksgiving, you can set that up with some simple rules. Since Thanksgiving in the U.S. is always the fourth Thursday of November, it can only fall between November 22nd and 28th. You need to define these dates and let the system know that you want to look for a Thursday. You would leave the times blank in order to define this as an exception to your regular time periods.

You can define as many of these types of these all-day “exceptions” as you wish.

Always click the Submit button to save your settings, followed by the Apply Config button when finished.

Setting Up a Time Condition

A time condition is a destination that checks the current time against a time group, and then routes calls to one of two destinations based upon the result. You need to create a time group first in order for a time condition to work. Otherwise your time condition wouldn't have anything to check. Please refer to the instructions earlier in this wiki. When you have defined your time group, now you can configure your time conditions.

A common use of this feature is to direct callers to a different destination during open business hours vs. closed hours. You could also use a time condition to route calls to the front desk during daily lunch periods, for example.

Your time condition will only affect call routing if you define it as a destination within another module, such as inbound routes, extensions, IVR, etc.

- Go to Applications → Time Conditions
- Click Add Time Condition
- Enter a Time Condition name. (Can contain spaces.)

If you would like to require users to enter a PIN before they can override the time condition, enter it in the Override Code PIN field.

Select the appropriate Time Group from the drop-down menu. The time condition you are creating will be linked to this time group.

Set the desired destinations. One will be used if the current time matches the time group, and the other will be used if there is no match. You cannot leave either of these blank.

In our example, we are sending callers to our IVR during our open business hours. After-hours, we are sending callers to a voicemail box.

Click the Submit button and then click the Apply Config button.

Setting a Time Condition as an Inbound Route Destination

You can route inbound calls to a time condition in order to send them to different destinations based on time of day. First, you must set up a time group and a time condition as described above. Once this is set up, you can route inbound calls to the time condition by doing the following:

- Go to Connectivity→Inbound Routes
- Click the edit button next to an existing inbound route.
- At the bottom, next to Set Destination, select Time Condition from the top drop-down menu, then select the name of your time condition from the bottom drop-down menu.
- Click the Submit button and then click the Apply Config button.
- Inbound calls will now flow through the time condition. Remember, a time condition requires a time group to be set up first, in order to work. You also need to set two destinations in the time conditions module in order to define where a caller will be sent.

Setting Up Conferences

The Conferences module allows you to set up a conference room where multiple callers can join in a conversation. The module will create an extension number for internal callers to dial to join a conference call. Other modules such as inbound routes, time conditions, IVR, etc. can also use your conference as a destination, allowing outside callers to reach the conference room.

- Go to Applications → Conferences
- Click the Add button.
- Enter a Conference Number that callers will dial to reach the conference. This cannot match any existing extensions or feature codes.

Enter a Conference Name that will help you identify it (can contain spaces).

- Optional: Set PINs for users and/or admins.
 - User PIN: Users would need to dial this after reaching the conference room, in order to join.
 - Admin PIN: The admin would dial this in order to be identified as the conference leader. An Admin PIN is mandatory if you enable the Leader Wait option.

You can set a variety of conference options. Please see the pop-up tooltips for details.

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

Direct-Dialing Into a Conference

External Callers:

If you would like external callers to be able to reach the conference room by dialing a phone number, you can set up the inbound route for one of your DIDs to go directly to the conference.

- Go to Connectivity → Inbound Routes
- Click the edit button for the DID you wish to route to the conference.
- Next to Set Destination, choose Conferences and select your conference room.

Internal Callers:

Internal extensions can reach your conference room by dialing the conference room number you have set up.

- Click Save Model on the left. You should be taken back to your phone template page.
- Click and Rebuild Configs and then reboot your phone. Your BLF will indicate whether the conference room is in use. You can press the button to dial into the conference.

Adding a Conference to an IVR

You can make it easy for your callers to reach the conference room through an IVR. Simply add your conference as a destination in the IVR. These instructions assume you have already created an IVR. If you need to create one, If you have an IVR set up with the the Enable Direct Dial: Enabled option set up, outside callers would be able to dial the conference number from within the IVR.

You can also configure an IVR option that would allow callers to dial a specific number (such as one digit) to access the conference:

- Go to Applications → IVR
- Click the edit button next to the IVR you want to edit.
- Under Ext, enter the digit(s) that the caller should press in order to reach the conference. Under Destination, choose Conferences and select your conference room. In our example, we've entered "5" as the digit to be pressed, and we've selected our conference room. "12345" is the conference room number.

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

Wake Up Calls

Overview

The Wake Up Calls module can be used to schedule a reminder or wake up call to any valid destination. To schedule a call, dial the feature code assigned in the Voice Express Feature Codes module (default is *68), or use the form in the GUI interface. You can also enable an operator mode that allows operators to schedule wake up calls for other extensions or destinations.

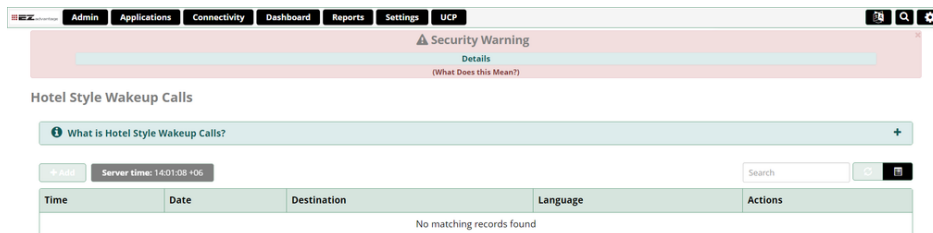
Logging in

- On the top menu click **Applications**
- In the drop down menu choose **Wake Up Calls**

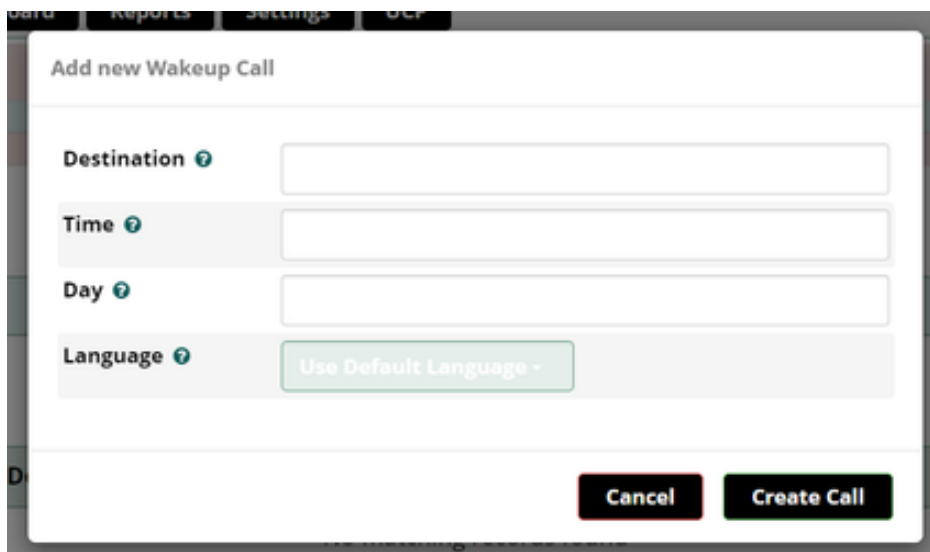
Usage

Adding a Call from the GUI

Click the **Add** button



A window will pop up where you can enter the destination and select a time and date.



- **Destination:** Enter the extension to be called.

- **Time:** Click in this field to display a drop-down menu of times. Select a time. Note: the menu only lists choices for each half-hour, but you can type in the Time field to change it to something different if needed (for example, 8:05am).

Day: Click in this field to display a calendar. Select a date.

Click the **Save Changes** button. There is no Apply Config button. Your new wakeup call will show up in the list, and the call will be made at the scheduled time.

Settings

Click the **Settings** tab.

Operator Mode

Yes / No - Whether to enable Operator Mode. Operator mode allows certain extensions to request wake up calls on behalf of any valid internal or external destination. If Operator Mode is disabled, wake up calls are only made to the Caller ID of the user who requests them.

Max Destination Length

Maximum length of a destination. This controls the maximum number of digits an operator can send a wakeup call to. Set to 10 or 11 to allow wake up calls to outside numbers. Otherwise, set to the number of digits used by your internal extensions (for example, if you have four-digit extensions, set this to 4).

Operator Extensions

Caller IDs that may act as an operator when Operator Mode is enabled above. Enter one per line. Operator extensions are allowed to create wake up calls for any valid destination. Numbers can be extension numbers, full caller ID numbers, or Asterisk dialing patterns.

Ring Time

How long to ring the destination, in seconds. Consider setting lower than the voicemail threshold in order to prevent the wake up call from going to voicemail.

Retry Time

The number of seconds to wait between retries. If a wake up call is not answered, another call will be sent after this number of seconds, until reaching the "max retries" threshold set below.

Max Retries

The maximum number of times the system should attempt to deliver the wake up call when there is no answer. Zero retries means only one call will be placed.

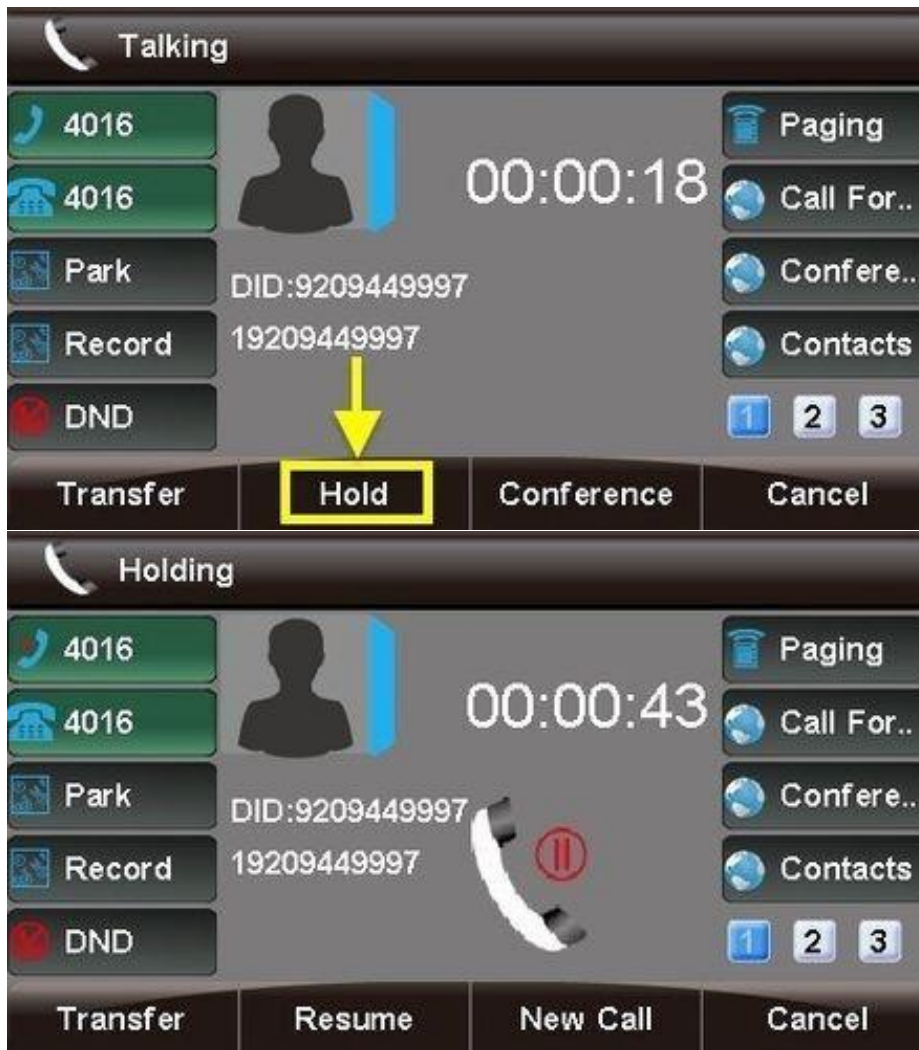
Wake Up Caller ID

How the wake up call will show up on the user's caller ID. Format: <#####>. You can also use the format: "hidden" <#####> to hide the CallerID sent out over Digital lines if supported (E1/T1/J1/BRI/SIP/IAX). Some systems require quote marks around the textual caller ID. Include the "" if required by your system.

Caller on Hold

To Place a Caller on Hold:

Press the (hold) button or press the Hold soft key.



To Resume a Call:

Press the (hold) button again or press the Resume soft key.

