Configuring your PBX

We are going to walk through a simple and typical setup of FreePBX. It would be impossible to teach you through a single wiki how to configure the over 1000 features of FreePBX, 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.

This setup assumes you have purchased the (System Building Basic) commercial module bundle to make setting up phones and your PBX much easier. It also assumes you are using SIP trunks. In our example we will use SIPStation for our trunking. The SIPStation module built into FreePBX Administration makes setting up FreePBX a breeze, as it does most of the work for us.

Configuration of the PBX is done using the various FreePBX Modules. The modules are divided into several categories at the top of the GUI. Once you're at a specific module's page, you can hover your mouse over the question mark icon

near each entry to view more information on what the item does.

For detailed instructions on each module, start here: Standard Modules.

- Login to the PBX Graphical User Interface ("GUI")
- Register your PBX
- Configure Asterisk SIP Settings
- Sysadmin Module Setup
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    - Configure E-Mail Settings
        - Configure Email Server
        - Configure E-Mail Alert Destination Addresses
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    - Setting a Hard Drive Failure / Fill-Up Notification E-Mail Address
    - Configuring Intrusion Detection
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    - Setting Up a Line Key for an Extension
    - Mapping the Phone to an Extension
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            - What if There is No Inventory?
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        - View the Search Results
            - How to Find "Vanity" Toll-Free Numbers
            - What if There is No Inventory?
        - Selecting Toll-Free Numbers
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- Setting Up Inbound Routes
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    - Setting Up an Inbound Route to an Extension Manually
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    - Setting Up an International Outbound Route Manually
    - Changing the Order of Routes

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- Setting Up a Queue
- Setting Up a Company Directory
- Creating System Recordings
    - Creating a System Recording Over the Phone
    - Uploading an Audio File from your Computer
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- Setting Up an Interactive Voice Response (IVR) System
- Setting Up a Time Group
    - Excluding Specific Time Periods, Such as Holidays
- Setting Up a Time Condition
    - Setting Up a BLF for a Time Condition
    - Overriding a Time Condition
    - Setting a Time Condition as an Inbound Route Destination
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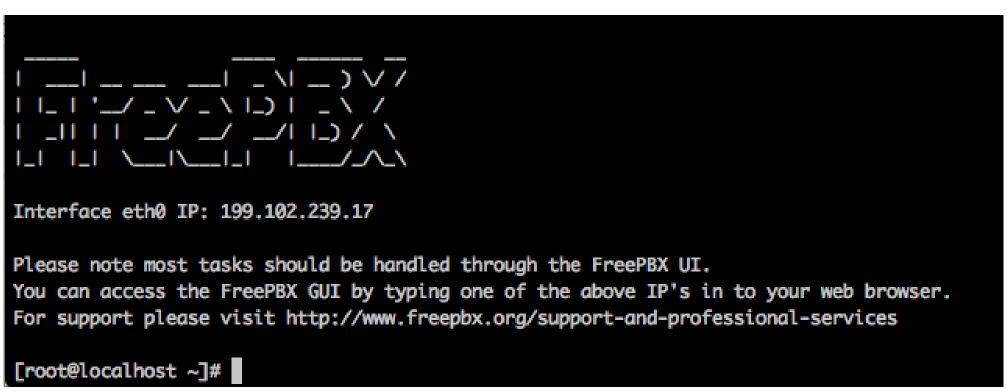
    - Adding a New User and Setting Up UCP Privileges
- Setting Up Conferences
    - Direct-Dialing Into a Conference
    - Setting up a BLF for a Conference
    - Adding a Conference to an IVR
    - Giving a User Access to Conferencing in UCP
- Setting Up Paging
    - Changing the Auto-Answer Sound
    - Paging Pro Features
    - Setting up Outbound Route Notification
- Setting Up Parking
    - Setting Up a BLF for a Parking Slot
- Setting Up Follow Me
    - Configuring Follow Me in UCP
    - Setting Up a BLF for Follow Me
- Setting Up Backups
    - Creating a Backup
    - Running a Backup Manually
- Other Modules
- Consider the Paid Modules

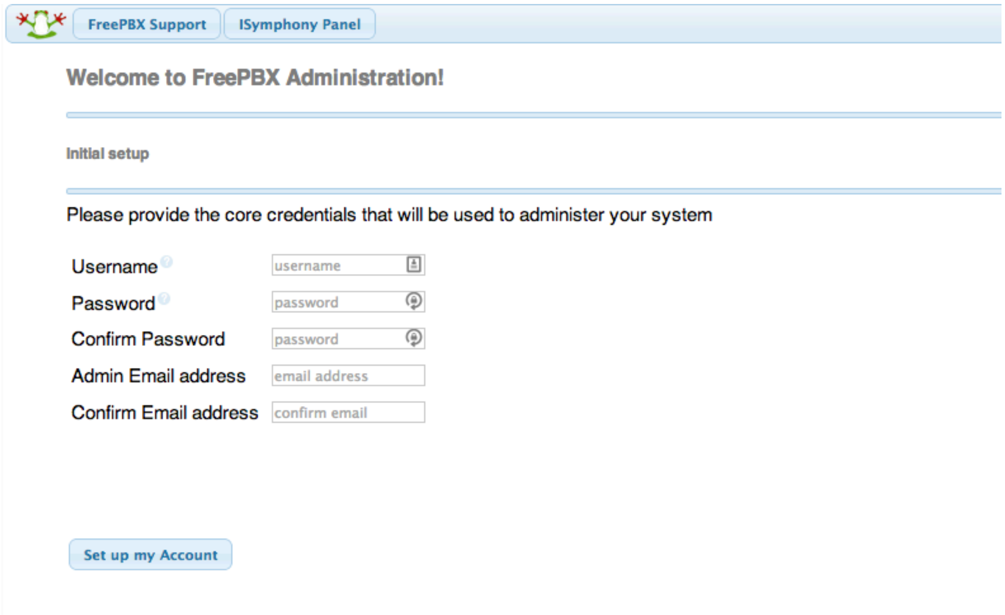
Login to the PBX Graphical User Interface ("GUI")

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

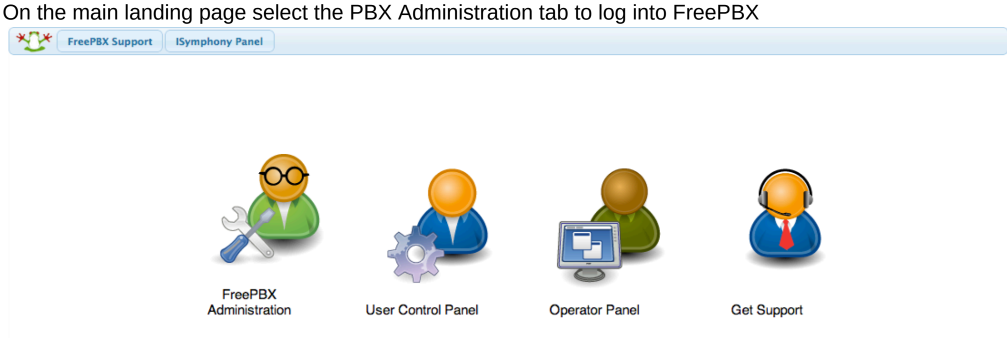
    - If you don't know the IP address of your PBX, go to the Linux console/command prompt. Login to the Linux console using the username "root" without quotes, and the root password you selected during installation. You will then be shown your IP address.



- You will now be asked to create a user to log into FreePBX GUI. Press Create Account when done.



- On the main landing page select the PBX Administration tab to log into FreePBX

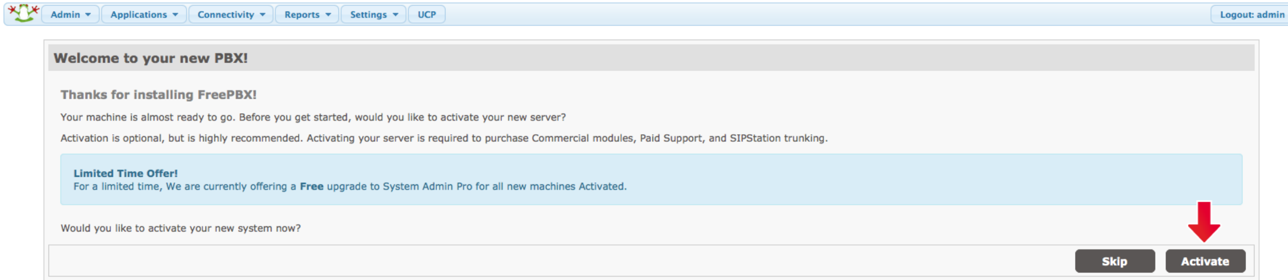


- Login with the username and password you just created.

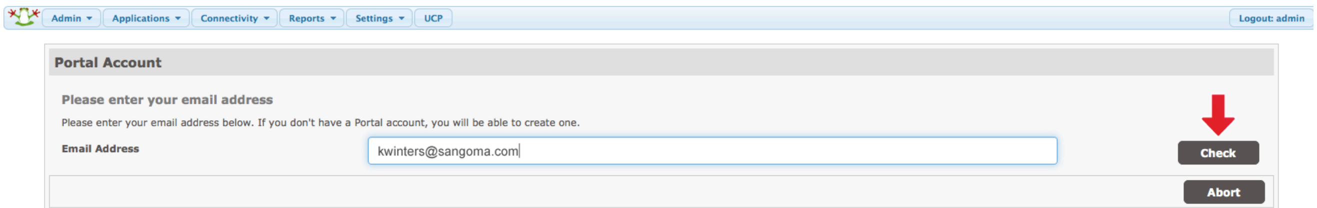
Note: These passwords do not change the Root password used to login to the Linux command prompt! They are only used for access to the web interface.

Register your PBX

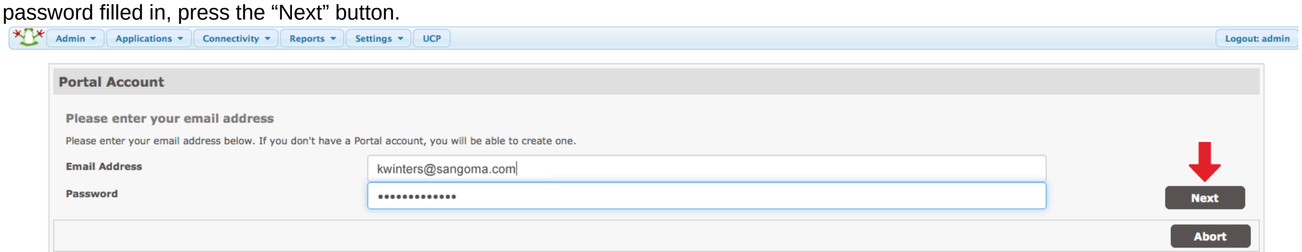
- From here, you'll be welcomed to your FreePBX and asked if you would like to activate your system. Activation is highly recommended, as it's required to purchase Commercial Modules, Paid Support and SIPStation Trunking. To activate now, press the "Activate" button.



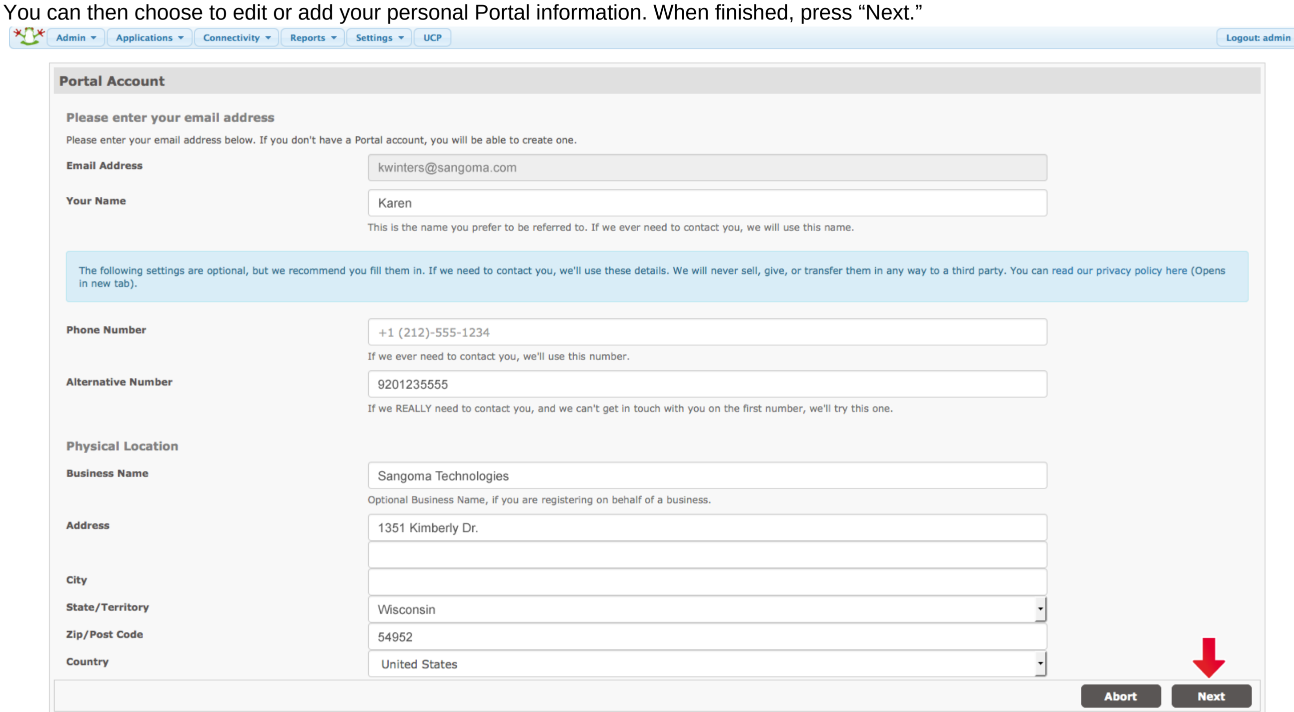
- Enter in your email address that you use for the FreePBX Portal. Once you've entered your address, press the "Check" button.
  - If you do not have an account enter the email address you would like to use for logging into the FreePBX Portal and you will be directed to create an account.



- The system will detect this address is tied to an existing Portal account and prompt you to enter your password. Once you have your password filled in, press the "Next" button.

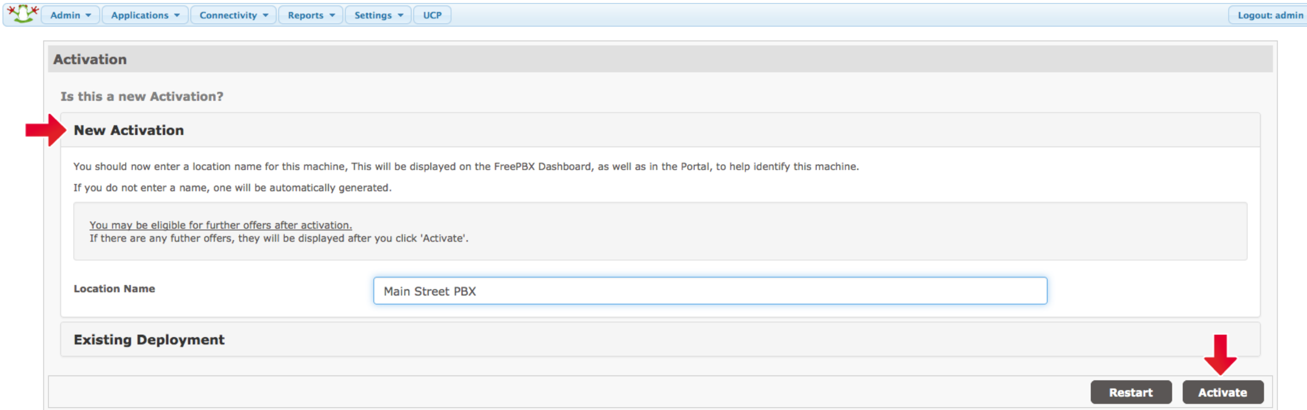


- You can then choose to edit or add your personal Portal information. When finished, press "Next."

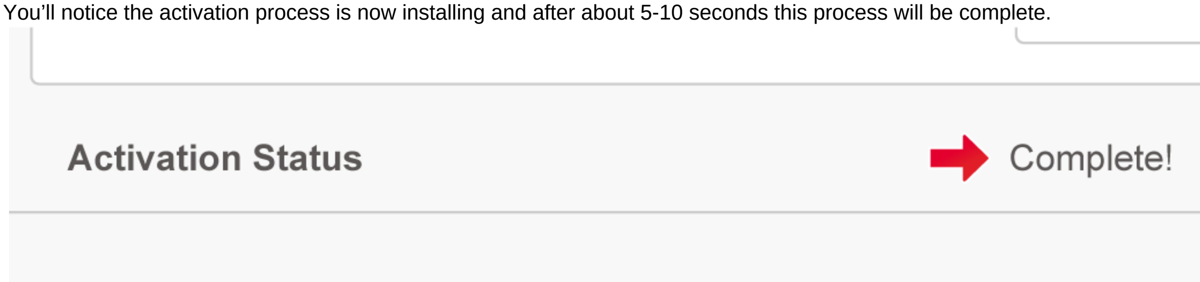


- From here, you'll be asked if this is a new activation. Since you don't have an existing Deployment ID, then you'll be looking at the first option, New Activation. All you need to do is give this deployment a "friendly" name to help you differentiate between your different systems when using the Portal to buy products. Then press the "Activate" button.
  - If you have a Deployment ID that is not tied to another PBX you can pick the Existing Deployment Option and provide that

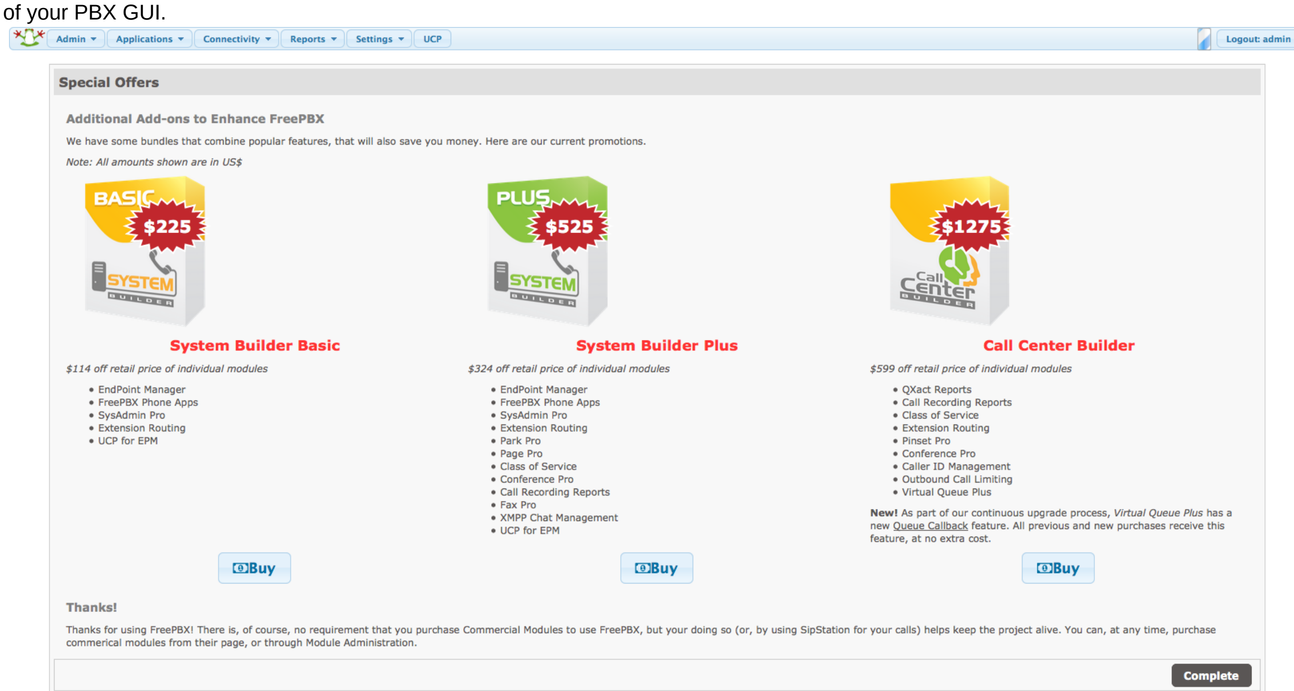
Deployment ID



- You'll notice the activation process is now installing and after about 5-10 seconds this process will be complete.



- You will now be able to take part in special offers. We offer bundles that combine some of our most popular features that will save you money. Click on one of the "Buy" buttons to purchase your bundle of choice or press "Complete" to advance to the System Status screen of your PBX GUI.



Configure Asterisk SIP Settings

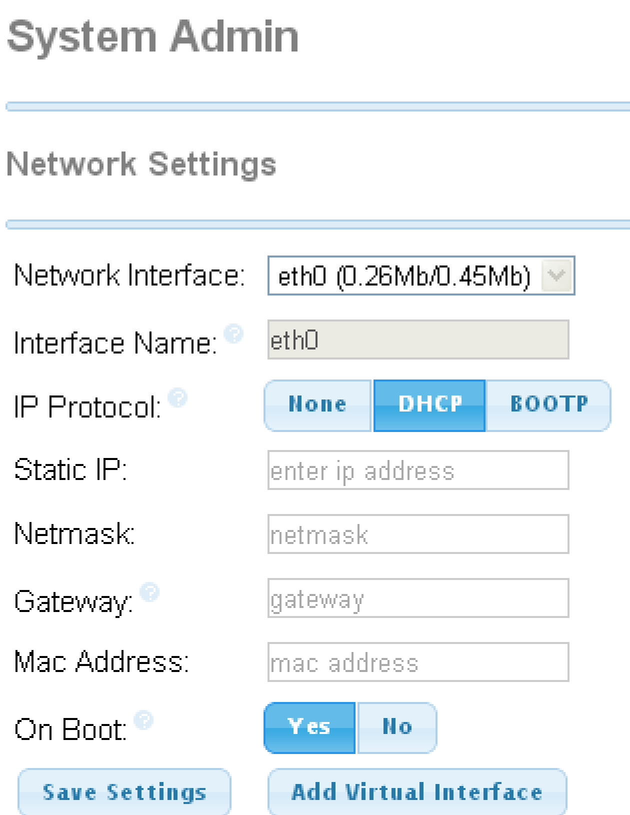
- Next, configure the Asterisk SIP Settings Module by following these instructions. The most important section, which you must configure in order to avoid one-way audio problems, is the "IP Configuration" section.

- Make sure you define your external IP and local subnets

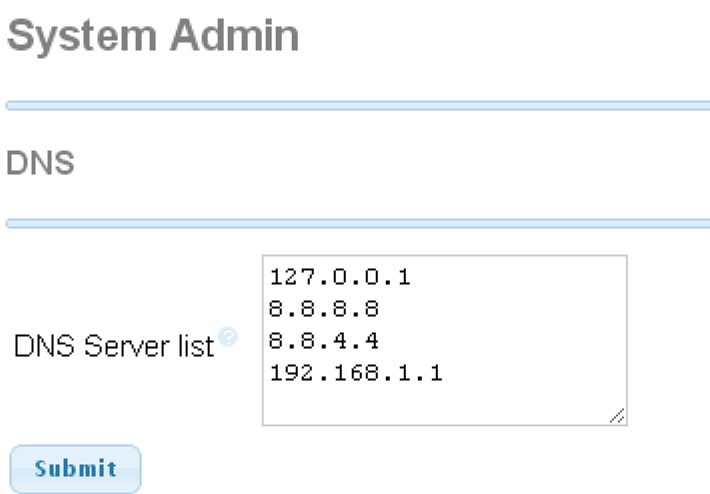
Sysadmin Module Setup

Set a Static IP Address and Configure DNS

- When FreePBX is first installed, it is configured to obtain an IP Address using DHCP. You'll need to assign your PBX 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 PBX a static IP address is to configure your DHCP server to always assign your PBX the same IP address using a DHCP reservation. This ensures that if DHCP settings later change, such as changing DNS server settings, the PBX will get these new settings with all other DHCP clients.   If that's not possible, you will need to configure your PBX to use a static IP and not use DHCP.
- From FreePBX GUI Click Admin, and then System Admin Module - Standard Version 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:



- 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:



The Distro installs DNSMASQ to ensure that your system maintains DNS even when the internet is down, and DNSMASQ won't work unless 127.0.0.1 is listed as your first DNS Server. The last two are Google's DNS servers. You can replace those two with your own, if you prefer.

- When you're done, click "Submit"

Configure E-Mail Settings

Configure Email Server

- View the System Admin Module - Standard Version module guide here to setup and manage your email relay servers.

Configure E-Mail Alert Destination Addresses

FreePBX has a number of components that will send out e-mail alerts on various conditions.

Check the following areas to verify or set the desired e-mail destination address of the phone system administrator who will receive the alerts:

- Admin > System Admin>Notifications Settings
    - From Address: Enter the From address the PBX will use for the notifications on this page, e.g. pbx@mydomain.com
    - Storage Notifications: (enter the desired e-mail address)
    - Intrusion Detection Notifications: (enter the desired e-mail address)
    - Abnormal Call Volume Notifications: (enter the desired e-mail address)
    - Submit

- Settings > Advanced Settings
    - Backup Module
        - Email "From:" Address: Enter the From address the Backup module will use, e.g. pbx@mydomain.com
        - click the green check mark to the right of the field
            - Apply Config

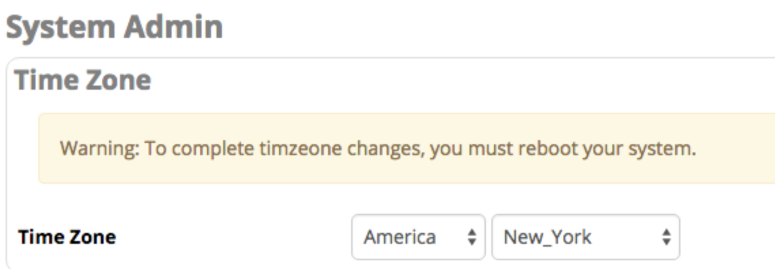
- Settings > Fax Configuration
    - Fax Options > Fax Presentation Options
        - Outgoing Email address: Enter the From address the fax module will use when sending received faxes, e.g. pbx@mydomain.com
    - Fax Options > Fax Feature Code Options
        - Email address: Enter the default destination e-mail address for received faxes so they can be manually routed by a human being, e.g. mainreception@mydomain.org
    - Submit
        - Apply Config

- Settings > Voicemail Admin
  - Settings
    - fromstring: Enter the From address the voicemail module will use when sending out voicemail alerts, e.g. pbx@mydomain.com
    - pagerfromstring: Enter the From address the voicemail module will use when sending out voicemail alerts, e.g. pbx@mydomain.com
    - serveremail: Set the same as "fromstring"; not sure how this is different
  - Submit
    - Apply Config

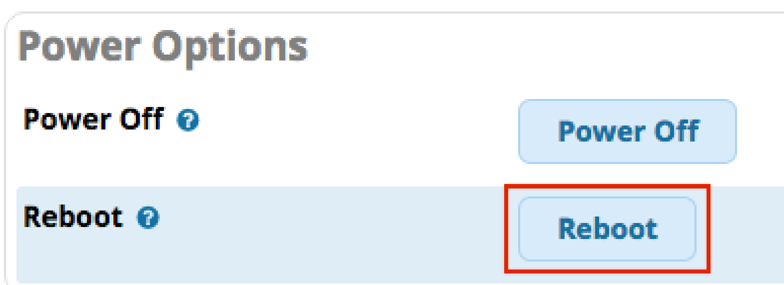
There is additional configuration needed for some of the above components but the above is just the e-mail From and To configuration.

Setting the Time Zone

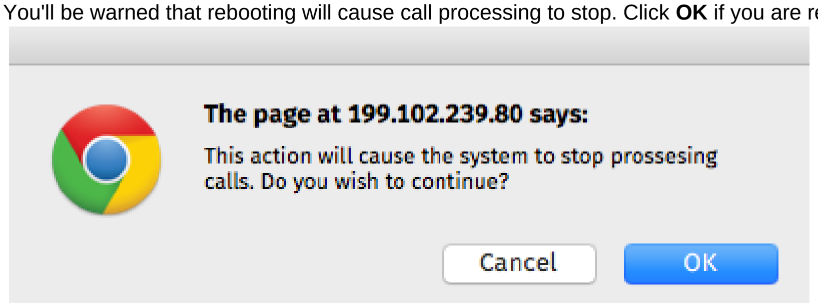
- Go to **AdminSystem AdminTime Zone**
- Use the drop-down menus to select the appropriate time zone and then click **Submit**. Note the warning stating that you must reboot your system in order to complete the changes.



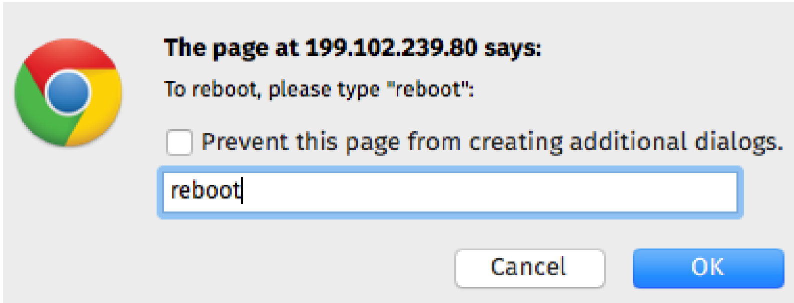
- Go to **Admin System Admin Power Options**
- *Be very careful with this section! You will want to reboot, NOT power off.*
- Click the **Reboot** button



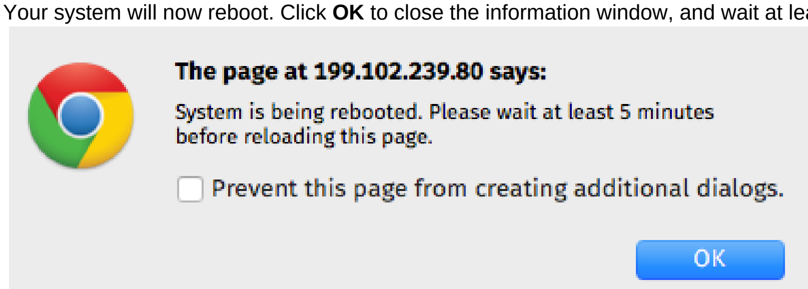
- You'll be warned that rebooting will cause call processing to stop. Click **OK** if you are ready to continue.



- 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.

Setting a Hard Drive Failure / Fill-Up Notification E-Mail Address

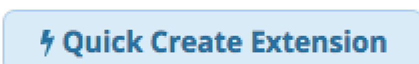
- Go to **Admin** -> **System Admin** -> **Storage**
- Enter an e-mail address where you would like to receive hard drive failure / fill-up notifications, and click **Submit**.

Configuring Intrusion Detection

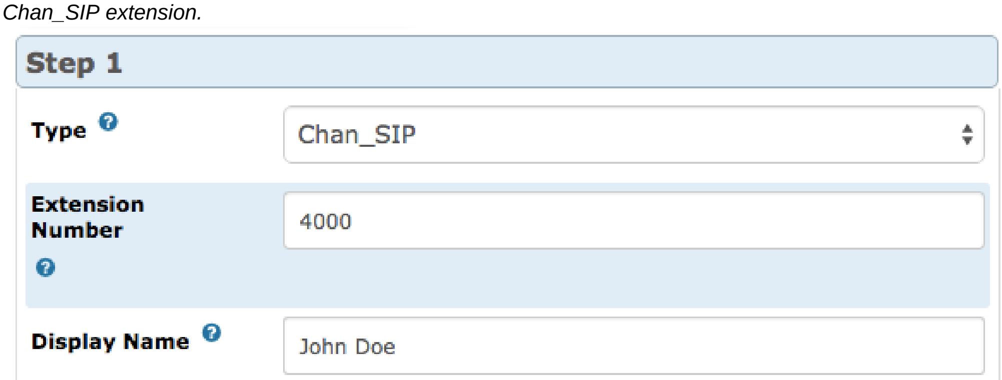
- Go to **Admin** -> **System Admin** -> **Intrusion Detection**
- We recommend that you keep this service running, in order to detect, block, and notify you of attempts to compromise your system. You may check the status here and adjust your ban time, max retry, and find time if needed.
- Enter an e-mail 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.

Creating Extensions

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



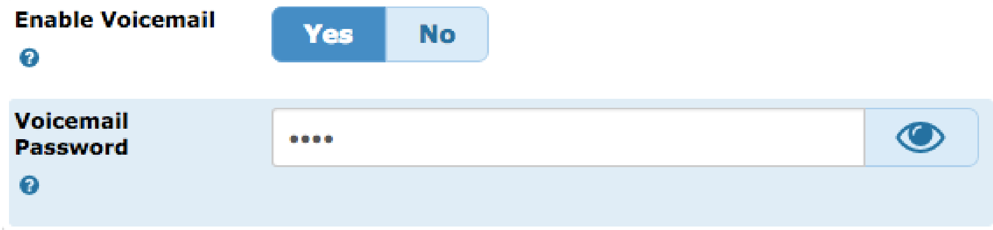
- Select the desired type of new extension, enter the extension number, and enter the display name. *In our example we are creating a new Chan_SIP extension.*



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

- Click **Next** to go to step 2.
- Optional: If you would like to enable voicemail now, do the following:
  - Click the **Yes** button next to "Enabled." This will make the password field available.
  - Enter an initial password (digits only). We recommend initially setting this to the extension number, because the first time the user dials *98, they will be prompted to set up their voicemail box and change this password. Users can also change their

passwords later by dialing *97 and changing voicemail settings.



- Click the **Finish** button.
- Click the **X** button (

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

) to edit, or the trash can icon (

) to delete.

Please see our Extensions Module wiki for more information.

Configuring a Phone Using EndPoint Manager (EPM)

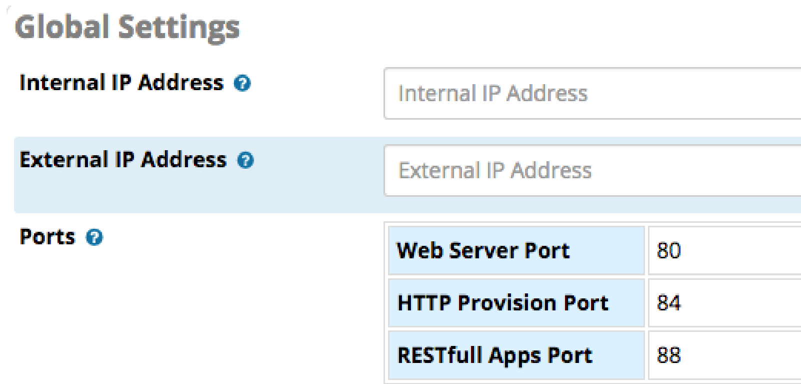
EndPoint Manager is a commercial module allowing you to configure over 220 devices and change settings on a per-template basis from the GUI. EndPoint Manager helps you quickly and easily configure a phone.

These instructions will walk you through basic configuration, including setting an IP address, creating a phone template, setting up a line key, mapping the phone to an extension, installing firmware, and applying phone config files. Keep in mind that you will always have to download a new config file to the phone in order to see any of your changes.

- Go to **Settings EndPoint Manager**
- Click on **Global Settings** in the menu at the upper right corner of the page.

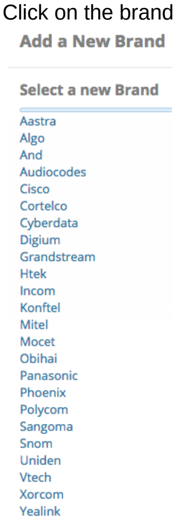
Setting the IP Address and Viewing the HTTP Provision Port

- Enter the IP address of your PBX into the **Internal IP Address** field. If you will have phones connecting to the PBX on an External IP or FQDN, enter that into the **External IP Address** field.
- Take note of the HTTP provision port shown, as some phones such as Digium only support HTTP or FTP provisioning.
- Click **Save Global** when done.

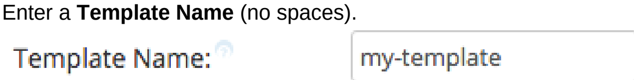


Creating a Template for the Phone

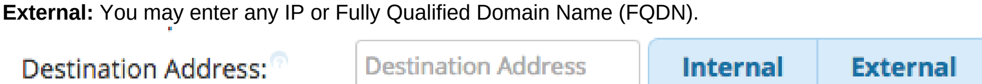
- Click **Add Brand** in the EndPoint menu at the upper right corner of the screen.
- Click on the brand of your phone. You will be taken to a page where you can begin creating a new template.



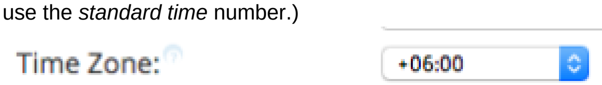
- Enter a **Template Name** (no spaces).



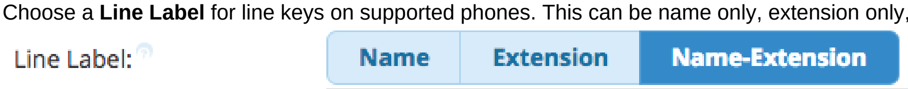
- Choose an **Internal** or **External Destination Address.**
    - **Internal:** The system will pull the IP address you defined earlier in Global Settings.
    - **External:** You may enter any IP or Fully Qualified Domain Name (FQDN).



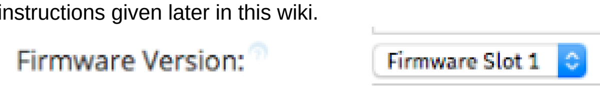
- Select the **Time Zone** for the phone. Choose the time offset from GMT, or select GMT. (Do not adjust for daylight savings time; use the *standard time* number.)



- Choose a **Line Label** for line keys on supported phones. This can be name only, extension only, or name and extension.



- For **Firmware Version**, select **Firmware Slot 1** from the drop-down menu. Note: you will need to install firmware by following the instructions given later in this wiki.



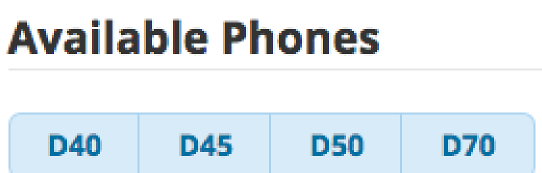
- Check your **Provision Server Address.** It should match the Destination Address set earlier.
- For **Provision Server Protocol**, click on **HTTP**, **FTP**, or **TFTP**, depending on the options you see available, and which protocol is correct for your phone.

- Click the **Save Template** button when finished. The page should reload and you should see a list of available phones. Next you'll configure some buttons on the phone.

Setting Up a Line Key for an Extension

- After you've saved your template, you should now see your available phone(s) at the bottom of the page. (To navigate to this page if you are not there already, go to **Settings EndPoint Manager Brands:** [Name of Brand] **Manage**: [Name of Template].)
- Click on the blue button for the phone you'd like to work on.

This example shows Digium phones. Your list may vary. We are working with the D70 in our example.

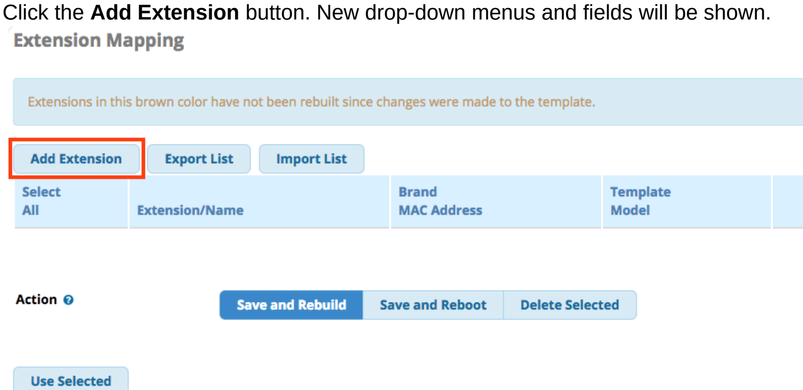


- In the **Line Keys** section, select **Type: Line**. The Label and Value fields will disappear. Select **Account: Account 1**.

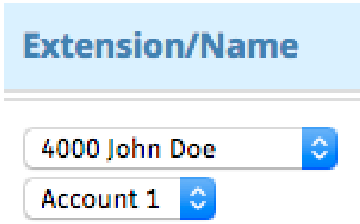
- Click the **Save Model** button on the left side of the page when finished.

Mapping the Phone to an Extension

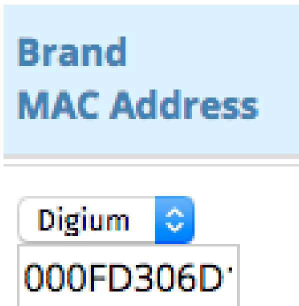
- Click on **Extension Mapping** in menu at the right side of the page.
- Click the **Add Extension** button. New drop-down menus and fields will be shown.



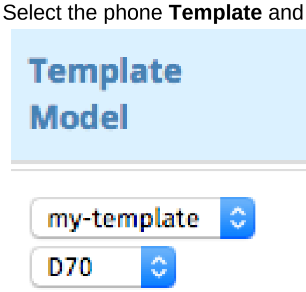
- Under **Extension/Name**, select the extension and account you would like to associate with this phone.



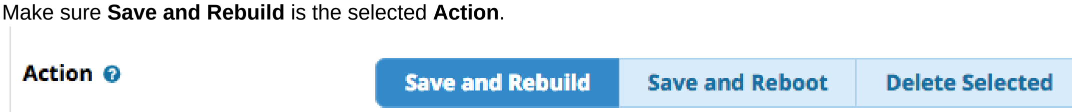
- Select your phone's **Brand** from the drop-down menu and enter its **MAC Address**. The MAC address is typically found on a label on the back of the phone.



- Select the phone **Template** and the **Model** of the phone.



- Make sure **Save and Rebuild** is the selected **Action**.

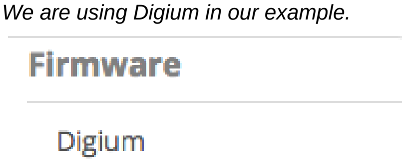


- Click the **Use Selected** button and wait for the page to reload.
- Your newly mapped extension should show up with a green background.

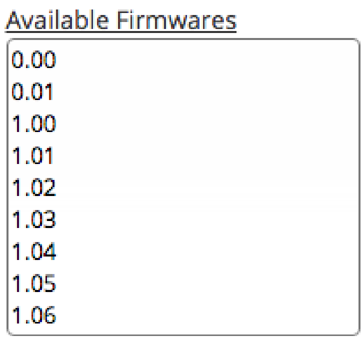
Select All	Extension/Name	Brand MAC Address	Template Model	
☐	4000 John Doe Account1	Digium 000FD306D15A	my-template D70	

Installing Firmware

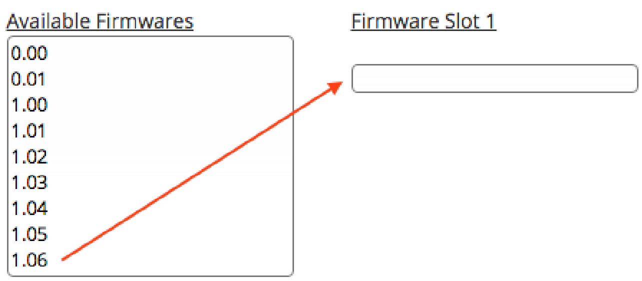
- Click **Firmware Management** in the menu at the right side of the page.
- Your phone's brand should appear under **Firmware**. Click the phone brand.
  We are using Digium in our example.

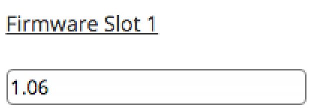


- You should now see a list of **Available Firmwares.**



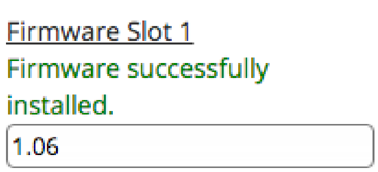
- Drag the latest firmware to Firmware Slot 1.





D40
1_4_2_0_63880
D45
1_4_2_0_63880
D50
1_4_2_0_63880
D70
1_4_2_0_63880

- Click the **Submit** button. The page will reload.
- Wait a few seconds, then reload the page. (Confirm form submission if prompted.) You should see a green message confirming the firmware has been installed.



Downloading Configuration Files to the Phone

Now we need to point the phone to the IP address and HTTP port of the PBX. The IP address and HTTP port were set up earlier in Global Settings. Setup may vary between phone models. Please refer to your specific phone's instructions.

The following instructions apply to the Digium D70:

- Press the phone’s **Settings** button

- Select **Fetch Configuration File from URL**
- Enter the **IP address** of your PBX
- Enter the **HTTP port** for provisioning
- Press **Go**
- Your phone should reboot, and you should now see the extension line key that was set up earlier.

Please see our End Point Manager wiki for more information.

Setting Up SIPStation Trunks

With SIPStation unlimited SIP trunks, you can be making and receiving calls from your PBX in just a few minutes. The FreePBX SIPSTATION module helps you set up SIP trunks easily and automatically. When you set up SIPStation trunks, a few basic inbound and outbound routes are automatically set up for you. We will discuss inbound and outbound routes later.

If you are a new SIPStation customer, our Free Trial program will let you try SIPStation before you buy. The Free Trial includes one trunk and one DID. Setup is quick and easy in the SIPSTATION module within FreePBX.

Setting Up a SIPStation Free Trial

SIPStation 21-day Free Trials are available in the lower 48 U.S. states and Canada. You won't need to give us a credit card, and you can get started immediately, as your trial service is provisioned instantly!

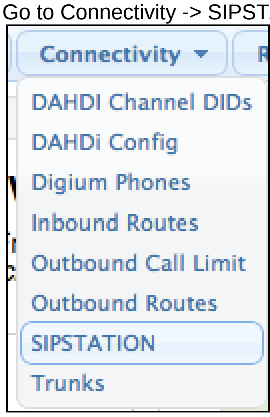
SIPStation Free Trials are Available Through Your PBX GUI

With FreePBX version 12 and above, in just a few clicks we can have you up and running with your PBX connected to our world class services. You just pick a phone number and we'll do the rest! There is no cost or obligation. You'll see just how easy it is to connect your phone system to our award-winning platform. There is no risk. After your evaluation, you can convert your account to a regular account (no contract period required), or end the trial and go your own direction. You can keep your Free Trial phone number (DID) when you convert to a regular account.

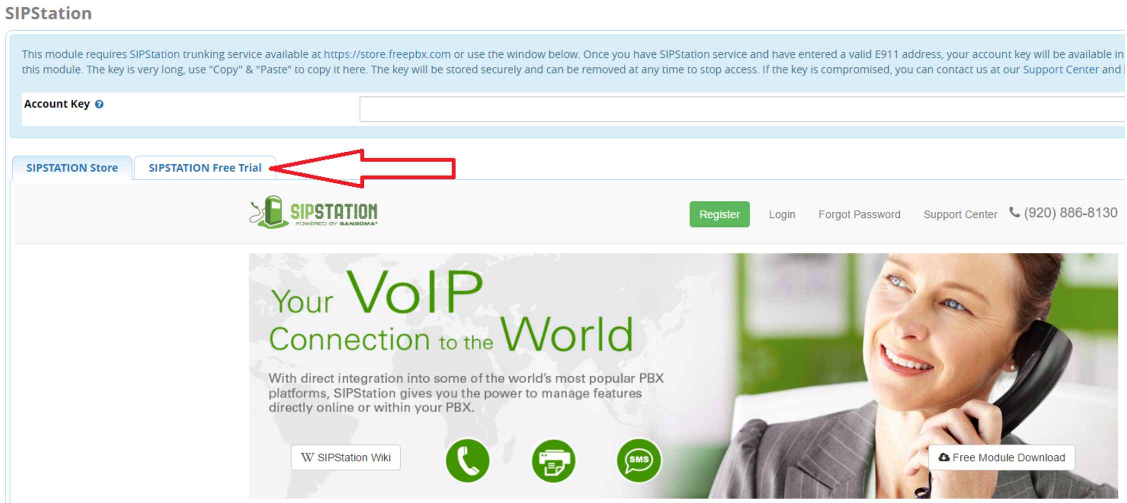
E911
E911 is not available with SIPStation Free Trial accounts. You will need an alternate means of calling 911 in an emergency.

How to Use the SIPSTATION Module in Your PBX GUI to Sign Up for a Free SIPStation Trial

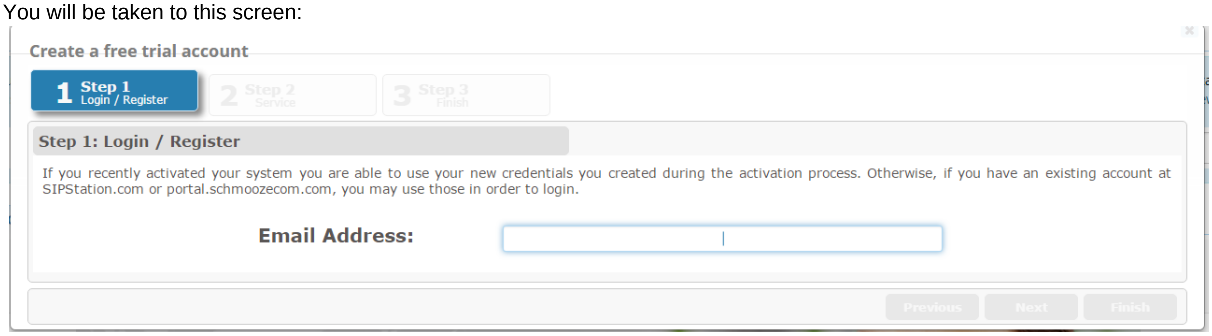
- Go to Connectivity -> SIPSTATION



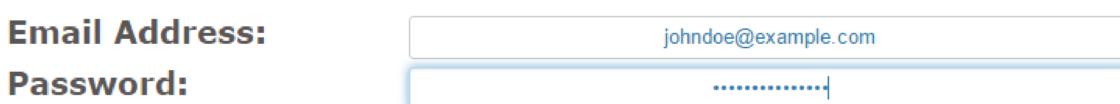
- Click on the "SIPSTATION Free Trial" tab.



- You will be taken to this screen:

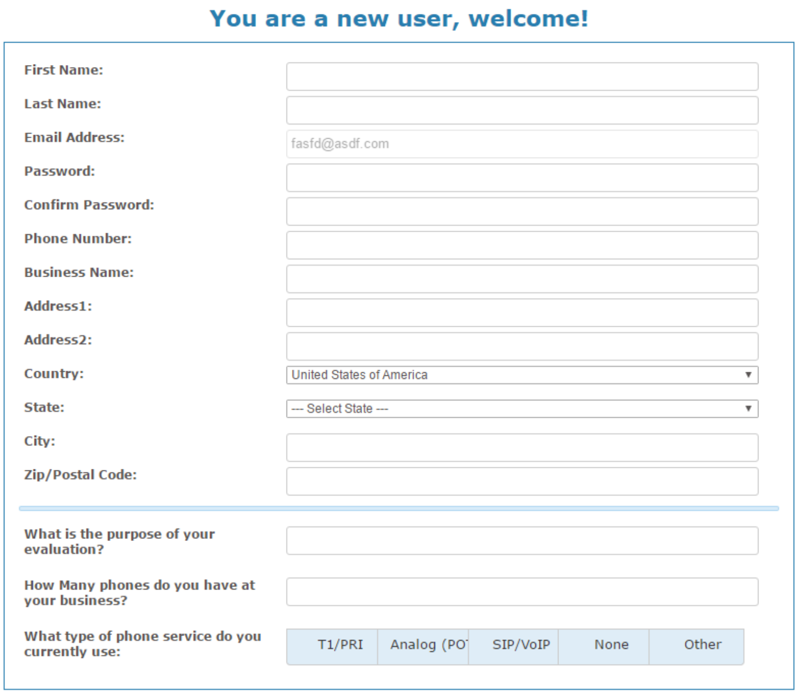


- **Step 1 - Login / Register:** You will need to log in with an existing account or register a new one.
  - **If you already have a SIPStation or FreePBX Portal account:**
    - Enter your Email Address and Password.



    - Click "Next"

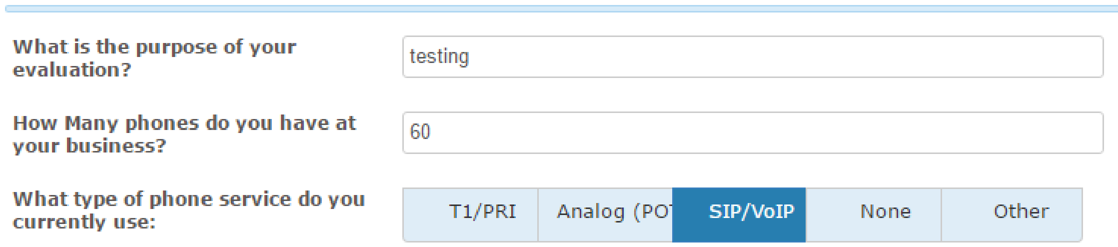
  - **If you need to create a new account enter in your email address and the new user screen as seen below:**
    -



- Enter your registration information and choose a password.

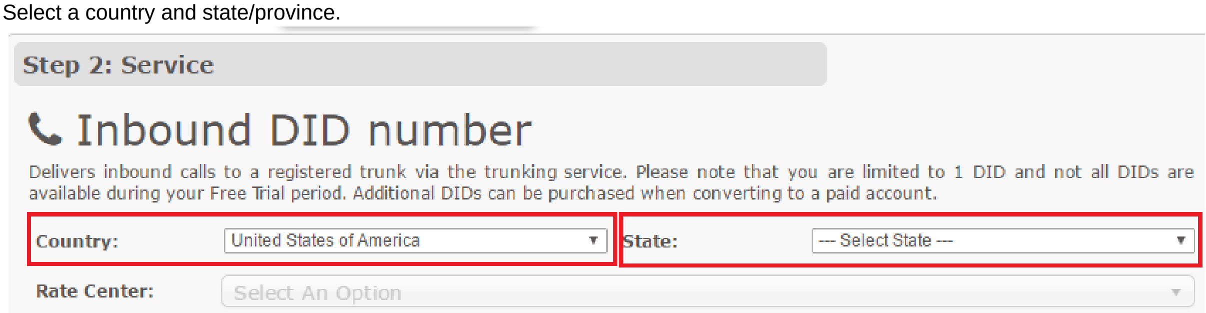
- Enter your address.

- Fill in the remaining information

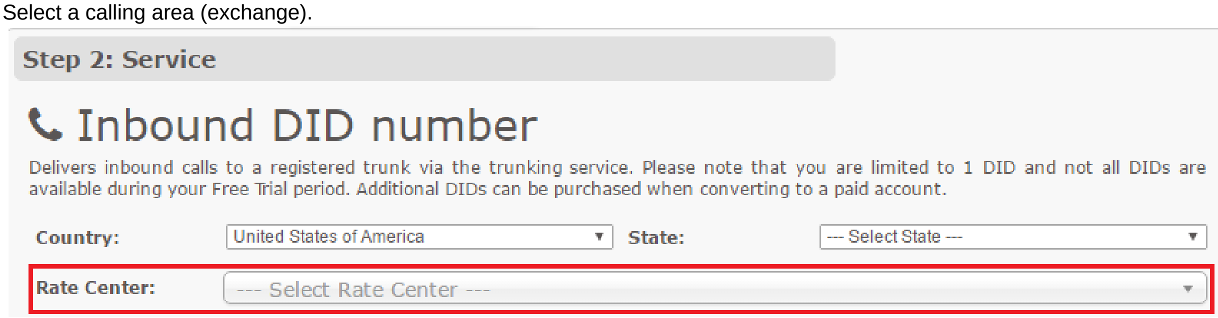


- Click "Next" when finished.

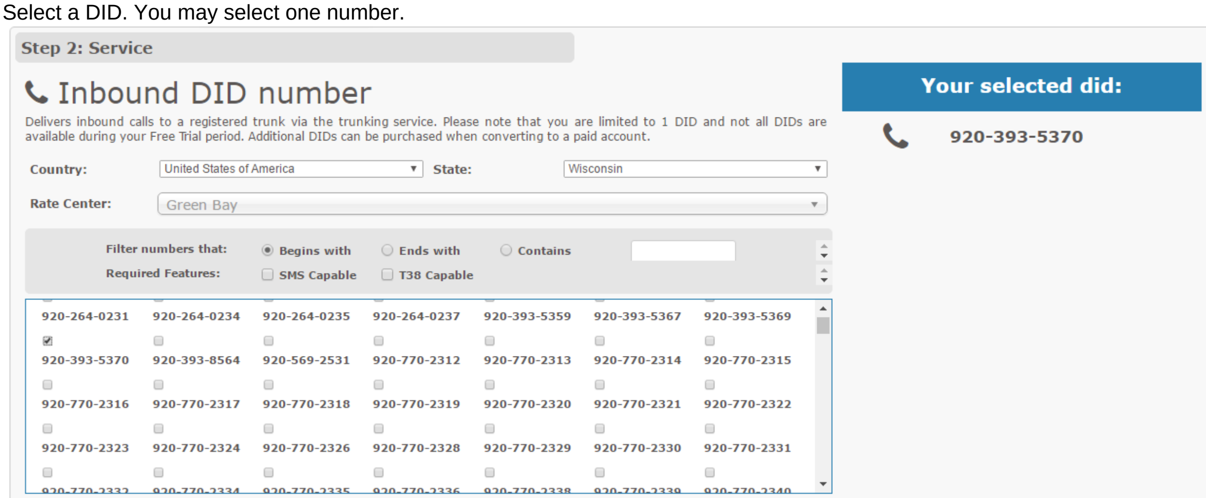
- **Step 2 - Service:** Choose a DID (Direct Inward Dial) number.
  - Select a country and state/province.



  - Select a calling area (exchange).

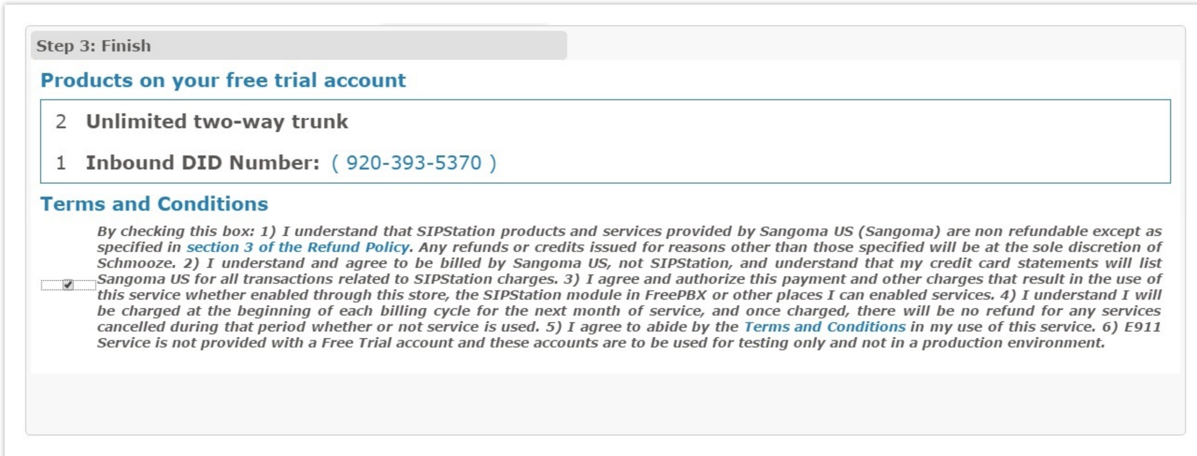


  - Select a DID. You may select one number.

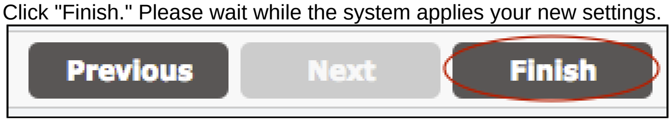


  - Click "Next."

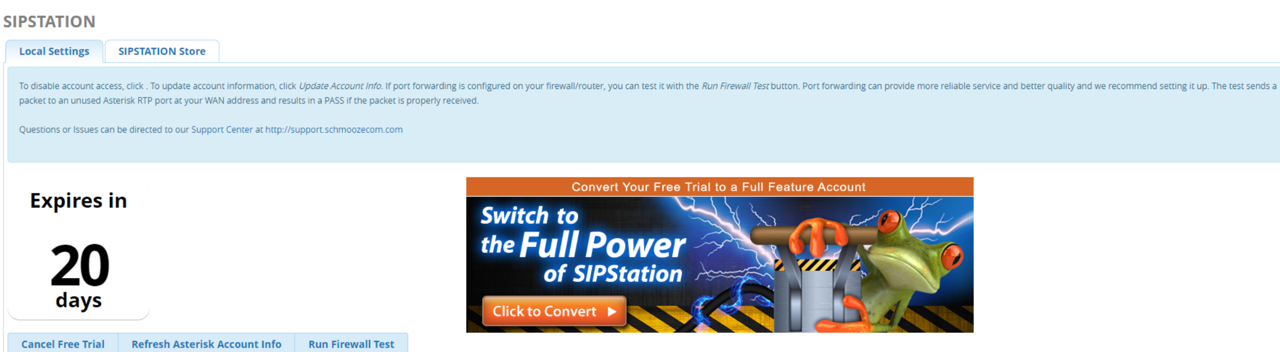
- **Step 3 - Finish:** You will see the products included with your SIPStation Free Trial account. You must agree to the Terms and Conditions in order to continue.
  - Check the box next to the Terms and Conditions.



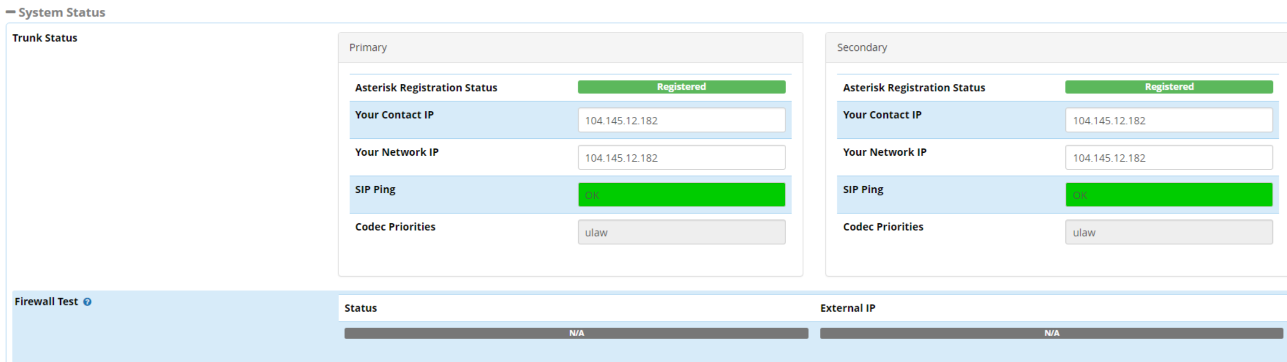
  - Click "Finish." Please wait while the system applies your new settings.



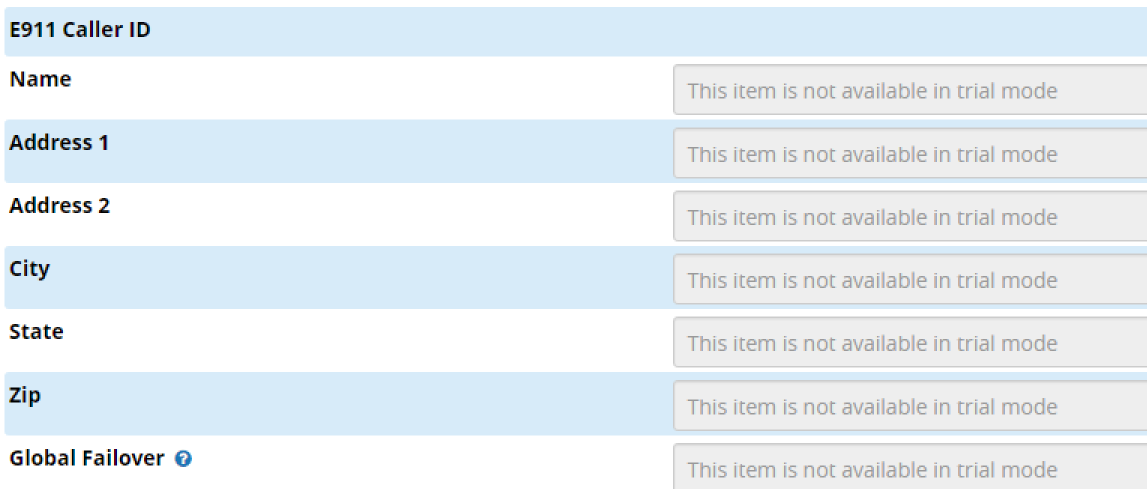
- After your free trial has been set up successfully, you will see new information in your SIPSTATION module. The tabs at the top are now called "Local Settings" and SIPSTATION Store, with no SIPSTATION Free Trial tab. The "Account Key" area has been replaced with instructions about how to cancel your free trial, update account information, or test your firewall. This area also displays the number of days remaining for the free trial.



- Your SIPSTATION System Status is displayed:

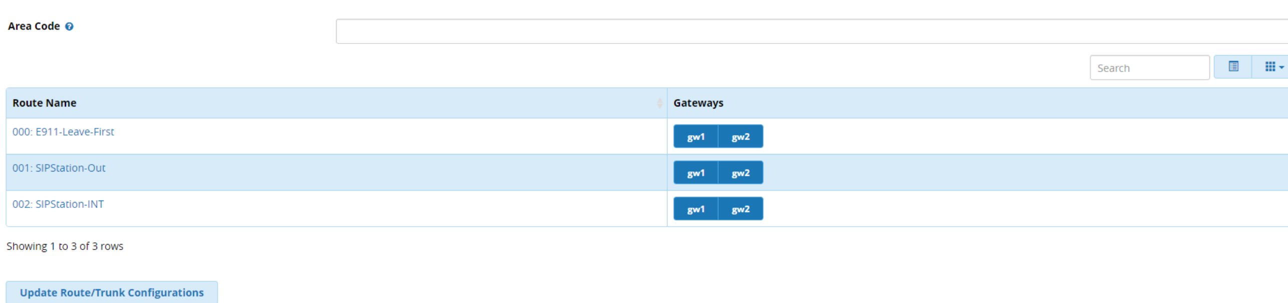


- Note that E911 options are greyed-out in trial mode. E911 is not available with SIPStation Free Trial accounts.



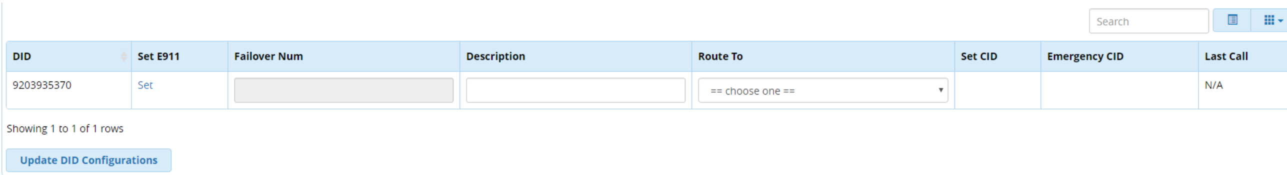
- Your SIPStation account settings are shown:

- Your route and trunk configuration is shown:



- Your DID configuration is shown.
The default inbound destination for your new DID is "Sipstation - DID Verification." In order to receive calls, you will need to change this to

a different destination, such as an extension.



Please see the Configuring your PBX wiki for a complete guide to getting your PBX system up and running. This wiki will help you create various call destinations and configure popular options.

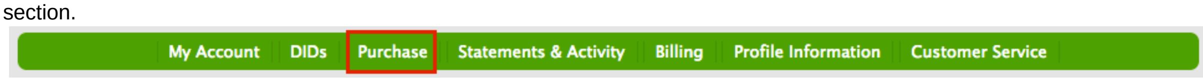
Purchasing SIPStation Trunks and DIDs

After you've created a SIPStation account at http://www.sipstation.com, it's easy to purchase new trunks, local DIDs, and toll-free numbers. Simply make your selections, add the items to your shopping cart, and complete the checkout process. Detailed instructions are found below.

- Logging into the SIPStation Store
- Adding Trunks
- Adding Local DIDs
  - Choose a Rate Center
  - View the Search Results
    - How to Filter by Specific Digits
    - How to Filter by SMS and/or T38 Capability
    - How to Find Consecutive DIDs
    - What if There is No Inventory?
  - Selecting DIDs
- Adding Toll-Free Numbers
  - Choose a Toll-Free Prefix
  - View the Search Results
    - How to Find "Vanity" Toll-Free Numbers
    - What if There is No Inventory?
  - Selecting Toll-Free Numbers
- Checkout Process
- Special-Ordering Local DIDs and Toll-Free Numbers

Logging into the SIPStation Store

- Before you can purchase any DIDs, toll-free numbers, or trunks, or make any special orders, you will need a SIPStation account. If you need to create an account, please see our Creating a SIPStation Account wiki for instructions.
- Log into your account at the SIPStation Store.
- If you are already logged in but are not seeing the store section, click **Purchase** in the green navigation menu to navigate to the store section.

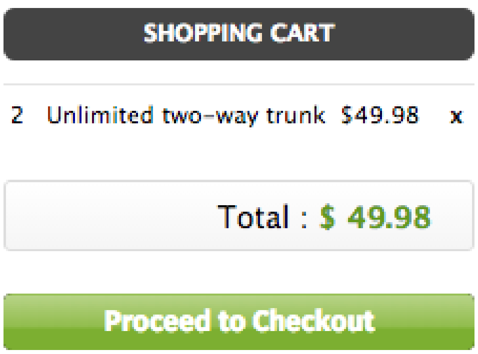


Adding Trunks

- Under "High Volume Voice Trunks," move the slider to select the number of trunks (concurrent call paths) you want to purchase. Click on the square at the left and drag it along the green bar to change the number of trunks selected.

- The total quantity of new trunks will be displayed in the green box at the upper right, along with the monthly fee for the trunk(s).

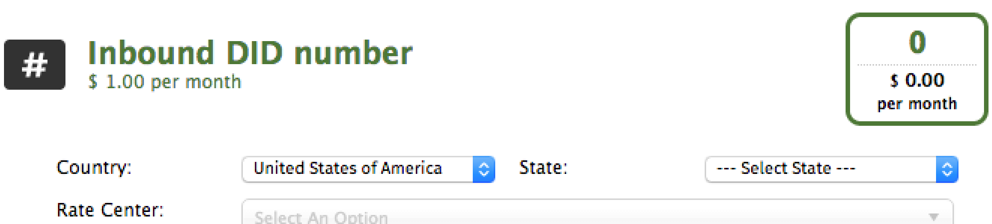
- The new trunks will be added to your shopping cart. If you need to remove a product, you can click the small **X** button.



- At this point you can continue shopping or click the **Proceed to Checkout** button to proceed to the checkout process, described later in this wiki.

Adding Local DIDs

- You can search for new local DIDs under the **Inbound DID number** section.



Choose a Rate Center

- A "rate center" is a geographic area whose numbers are considered to be local to each other instead of long distance. Rate centers vary in size, ranging from one small community to several large cities combined. A rate center may contain several area code-prefix combinations (NPA-NXXs).
    - Select a **Country** (US or Canada)

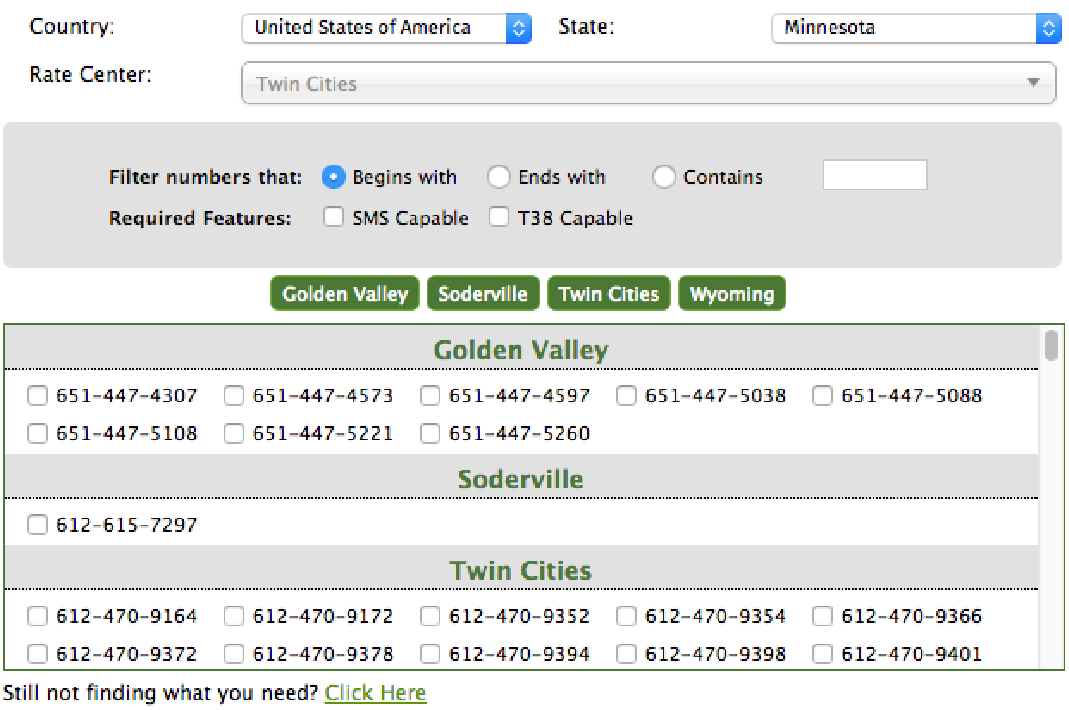
- Select a **State** or **Province**
- Select a **Rate Center**.

> **Tip**
> If you are unable to find the city you are looking for, take a look at the list of rate centers to see if your city might be included under a different name. For example, Minneapolis and St. Paul, MN are both part of the "Twin Cities" rate center.

View the Search Results

- Available DIDs will be shown below your selected rate center. In some rate centers, the search results will be divided up into specific areas, which will be listed alphabetically. You can click on the green buttons to jump to the DIDs for these different areas, or you can scroll down within the search results to view all available DIDs. Numbers are listed in ascending numerical order.

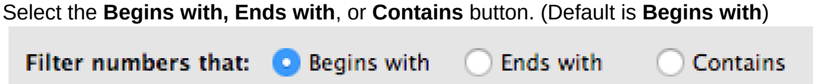
- *For example, here we ran a search for the Twin Cities rate center and are finding four different areas available within this rate center.*



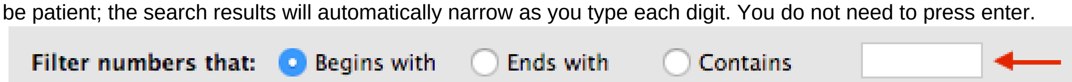
How to Filter by Specific Digits

- Are you looking for specific digits at the beginning, end, or middle of your phone number? If so, you can enter those digits to filter your search results.

- Select the **Begins with, Ends with**, or **Contains** button. (Default is **Begins with**)



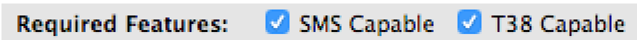
- Type your desired digit(s) into the field next to these buttons. Do not enter any special characters such as parenthesis or dashes. Please be patient; the search results will automatically narrow as you type each digit. You do not need to press enter.



- If you need to undo your specific search, you can delete what was entered in the field, and the list should return to its original search results.

How to Filter by SMS and/or T38 Capability

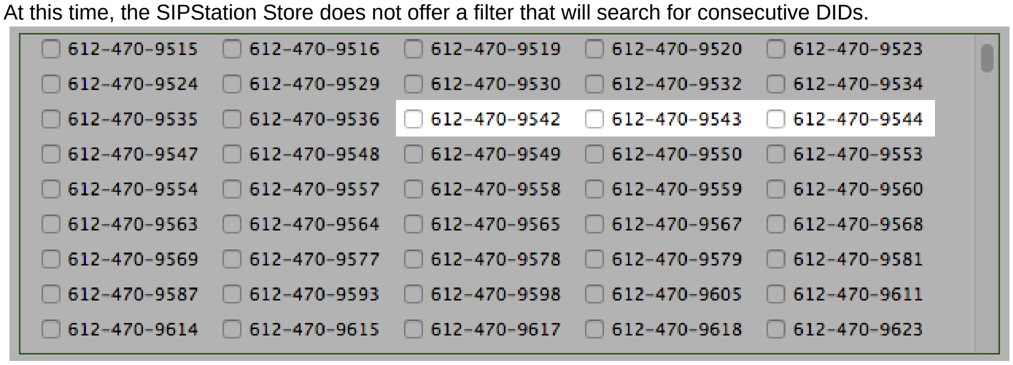
- Do you need SMS (text messaging) and/or T38 (fax) capability with your new DID? If so, check the appropriate boxes to filter your search results.



How to Find Consecutive DIDs

- Since available numbers are listed in sequence, in ascending order, you can browse the list to search for consecutive DIDs. Simply scroll down the list to see if you can find any sequential numbers.

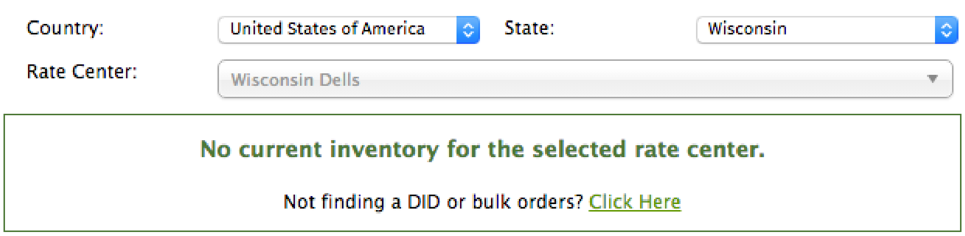
- At this time, the SIPStation Store does not offer a filter that will search for consecutive DIDs.



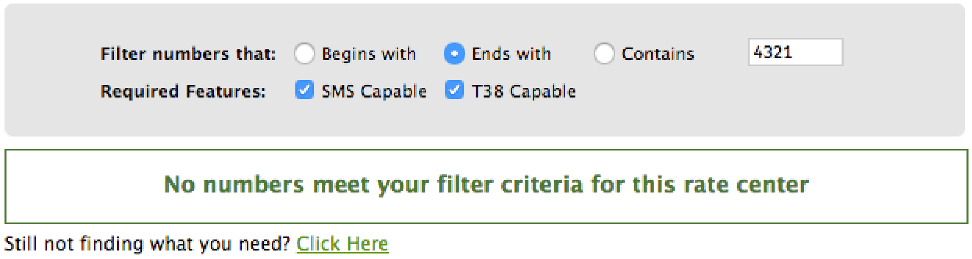
- There is no special fee for consecutive DIDs when you find them in the SIPStation Store. If you would like us to help you with finding and ordering consecutive DIDs, we can help. You can use the special order process to make this request, and there would be a $2-per-DID setup fee for this service.

What if There is No Inventory?

- If you are seeing the message "**No current inventory for the selected rate center**," it means we do not have new numbers immediately available for that area. You can check back again later to see whether new numbers become available. Or, you can use our special-ordering process, and a member of our team will search for and order numbers for you. The special-order process may involve backordering numbers and isn't guaranteed to be successful, but in many cases, we will be able to fulfill your request.

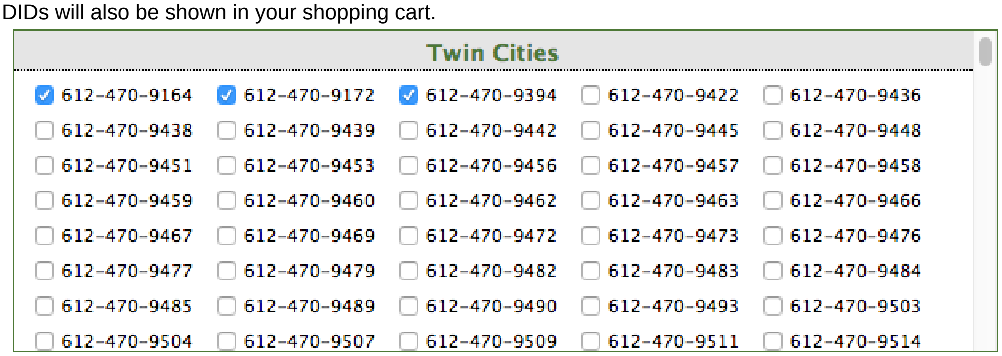


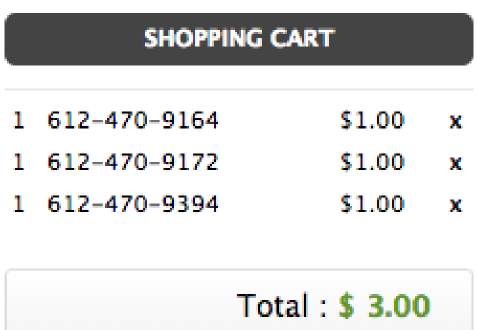
- If you are seeing the message "**No numbers meet your filter criteria for this rate center**," you can try selecting different filter options, for example by changing your search for specific digits, or by removing the filter for SMS and/or T38 capability.



Selecting DIDs

- After you've found some available numbers that meet your search criteria, it's easy to add the numbers to your shopping cart. Simply check the checkbox next to each DID you would like to order. The total quantity and monthly fee will be shown in the green box, and the DIDs will also be shown in your shopping cart.

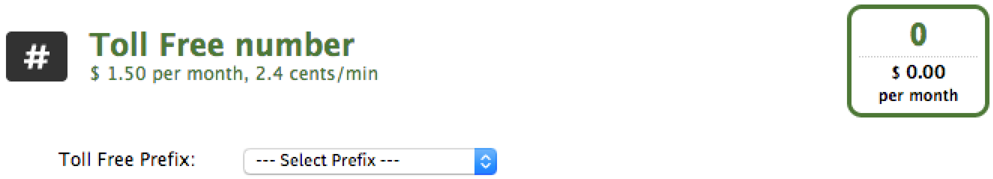




- You can click the **X** next to an item in your shopping cart if you need to remove it.
- At this point you can continue shopping or click the **Proceed to Checkout** button to proceed to the checkout process, described later in this wiki.

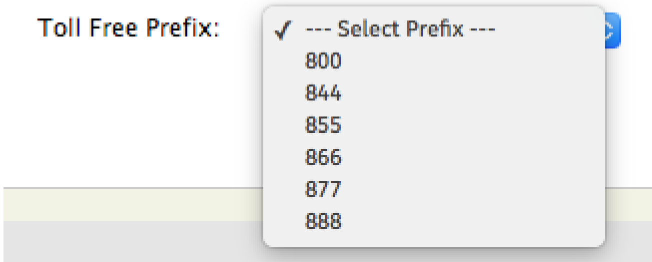
Adding Toll-Free Numbers

- Toll-free numbers begin with 800, 844, 855, 866, 877, or 888. When a caller calls your toll-free number, you pay for the call. You can search for new toll-free numbers under the **Toll Free number** section.



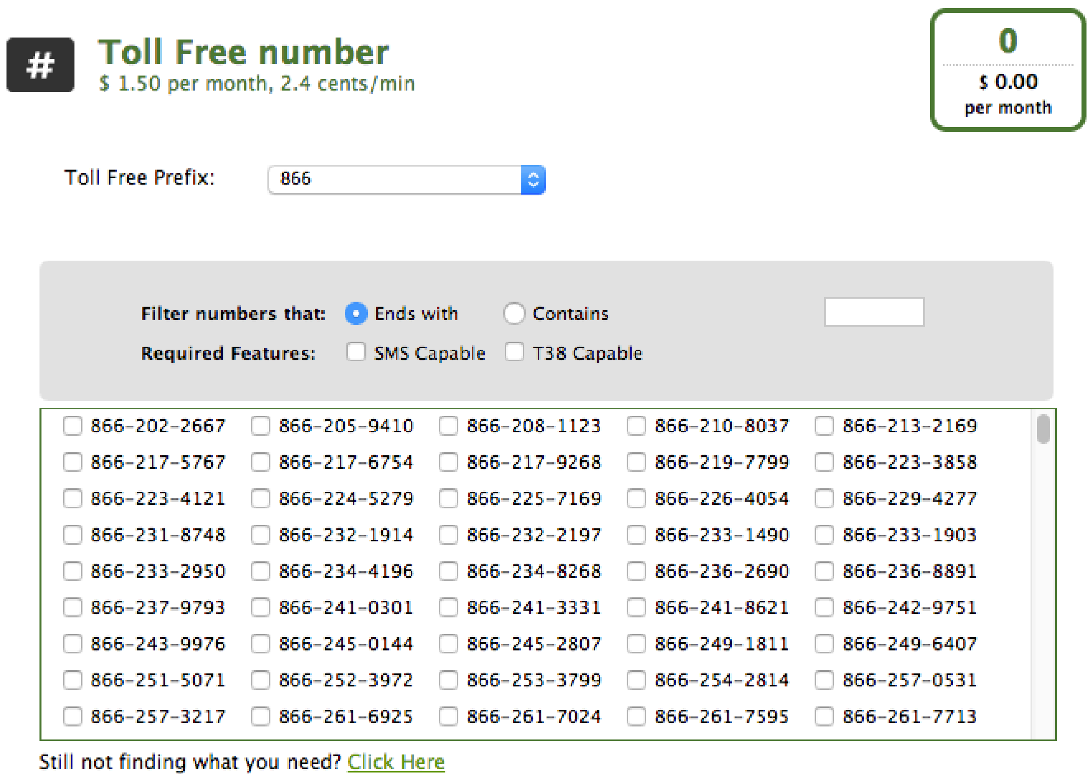
Choose a Toll-Free Prefix

- Select a **Toll Free Prefix** from the drop-down menu.



View the Search Results

- Available toll-free numbers will be shown below your selected prefix. Numbers are listed in ascending numerical order.



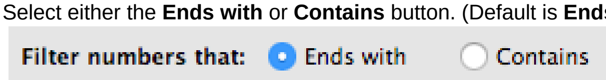
How to Find "Vanity" Toll-Free Numbers

- A "vanity" number is an easy-to-remember telephone number such as 1-800-FLOWERS or 1-800-FREE-411. A vanity number benefits your business by helping customers remember how to reach you.
- At this time, we do not offer the capability to filter toll-free search results by letter, but you can search by number. On phone keypads,

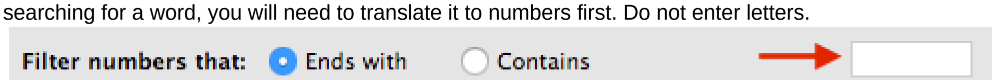
numbers translate to the following letters:

- **2**: ABC
- **3**: DEF
- **4**: GHI
- **5**: JKL
- **6**: MNO
- **7**: PQRS
- **8**: TUV
- **9**: WXYZ

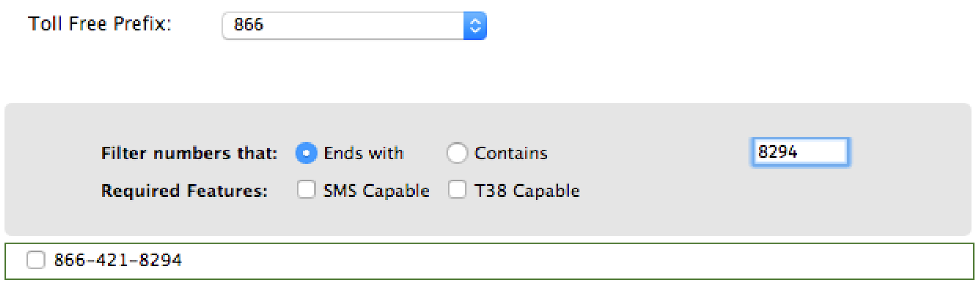
- Choose your desired **Toll Free Prefix** from the drop-down menu.
- Select either the **Ends with** or **Contains** button. (Default is **Ends with**)



- Type your desired digit(s) into the field next to these buttons. Do not enter any special characters such as parenthesis or dashes. Please be patient; the search results will automatically narrow as you type each digit. You do not need to press enter. Remember, if you are searching for a word, you will need to translate it to numbers first. Do not enter letters.



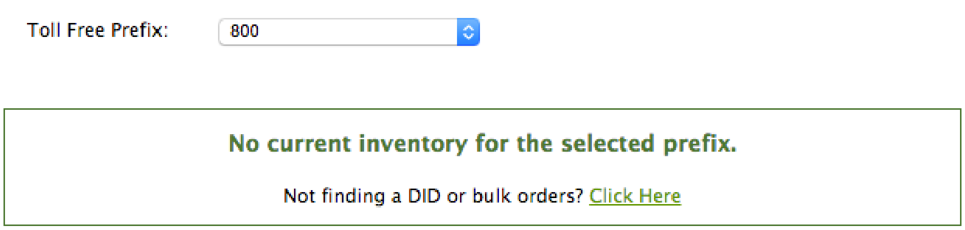
For example, let's say we are looking for an 866-number that ends in "TAXI." We'll select 866 for the prefix, then select "Ends with" for the filter. We'll enter "8294" in the search field, because TAXI=8294. We're in luck! There is a number meeting that search criteria.



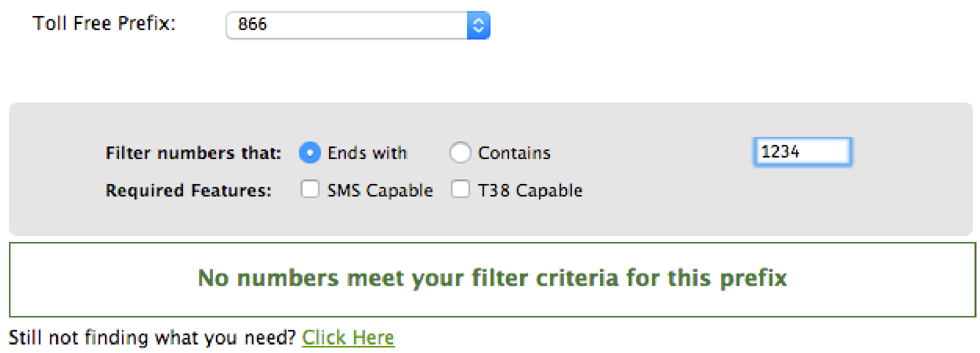
- If you need to undo your specific search, you can delete what was entered in the field, and the list should return to its original search results.

What if There is No Inventory?

- If you are seeing the message "**No current inventory for the selected prefix**," it means we do not have new numbers immediately available in that toll-free prefix. You can check back again later to see whether new numbers become available. Please note, we are not able to backorder toll-free numbers. If you use our special order form by clicking the "Click Here" link, we can only help you search our existing inventory.

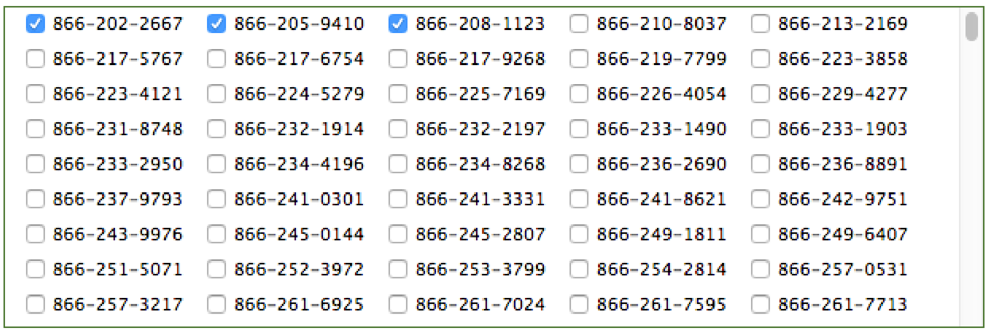


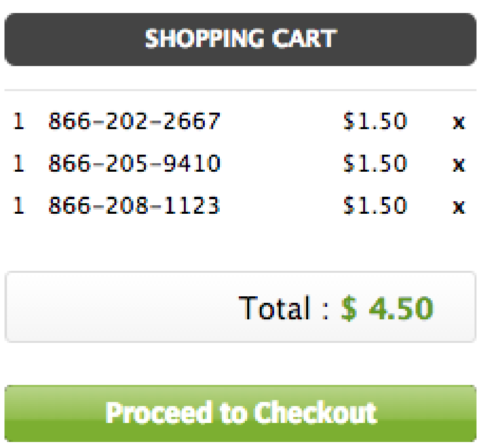
- If you are seeing the message "**No numbers meet your filter criteria for this prefix**," you can try entering different digits or removing digits. Also, since toll-free numbers typically do not support SMS or T38, you can try de-selecting these options.



Selecting Toll-Free Numbers

- After you've found some available numbers that meet your search criteria, it's easy to add the numbers to your shopping cart. Simply check the checkbox next to each toll-free number you would like to order. The total quantity and monthly fee will be shown in the green box, and the numbers will also be shown in your shopping cart.



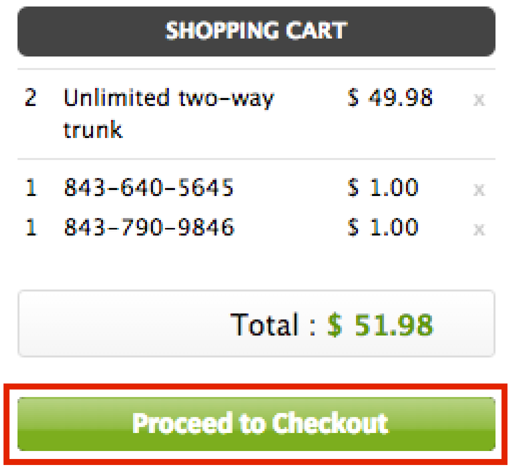


- You can click the **X** next to an item in your shopping cart if you need to remove it.
- At this point you can continue shopping or click the **Proceed to Checkout** button to proceed to the checkout process, described later in this wiki.

Checkout Process

You must continue to the next step and click **Proceed to Checkout,** or the DID(s) and toll-free number(s) in your shopping cart will not be saved to your account, meaning you will lose the ability to use those numbers if another user selects them before you check out. If you need to remove any items from your shopping cart, you can click on the gray **X** next to the item.

- Add services such as trunks, local DIDs, and toll-free numbers to your shopping cart using the processes described earlier in this wiki.
- Click on the green **Proceed to Checkout** button when ready to make your purchase.



- Review your Order Summary. If needed, click **Add / Edit Items** to go back to the store, where you can remove items from your cart or select additional items to add.

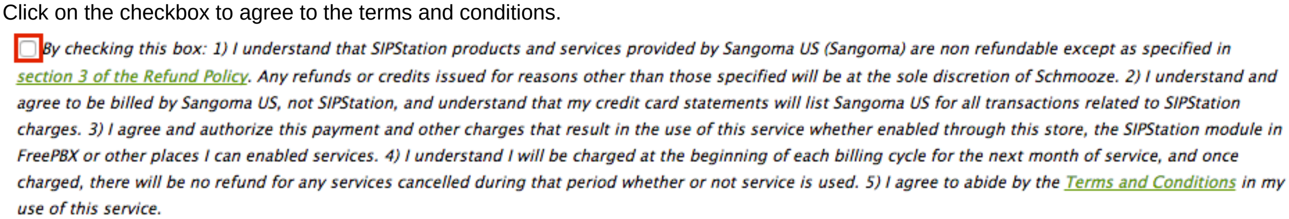
Your Order Summary

Add / Edit Items

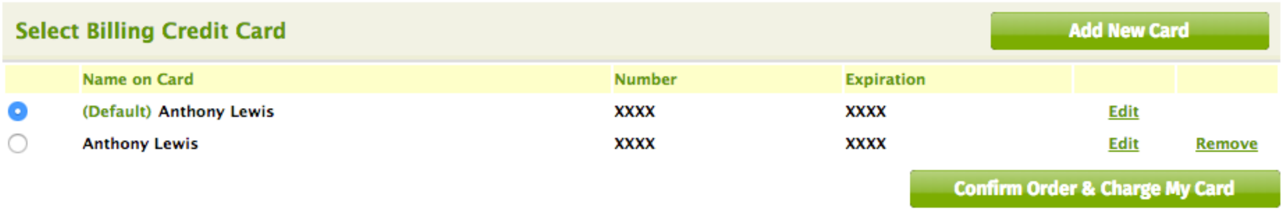
Item	Options	Cost	Quantity	Total
Unlimited two-way trunk		$24.99	2	$49.98
Inbound DID Number(s)	8436405645	$1.00	1	$1.00
Inbound DID Number(s)	8437909846	$1.00	1	$1.00
			Total:	$51.98

The section "After Your Order, Your Subscription Will Include" section at the bottom of the page shows what the new monthly subscription fees will be if you make this purchase. ***Totals do not include taxes and fees, which vary by location.*** We are required to collect various taxes and fees according to federal, state, and local regulations. Please see our wiki on Taxes and Fees for more information.

- Click on the checkbox to agree to the terms and conditions.



- Select an existing credit card to use for this order, or add a new card. Then click **Confirm Order & Charge My Card.**



- If your order completes successfully, you should get the following notification:

Special-Ordering Local DIDs and Toll-Free Numbers

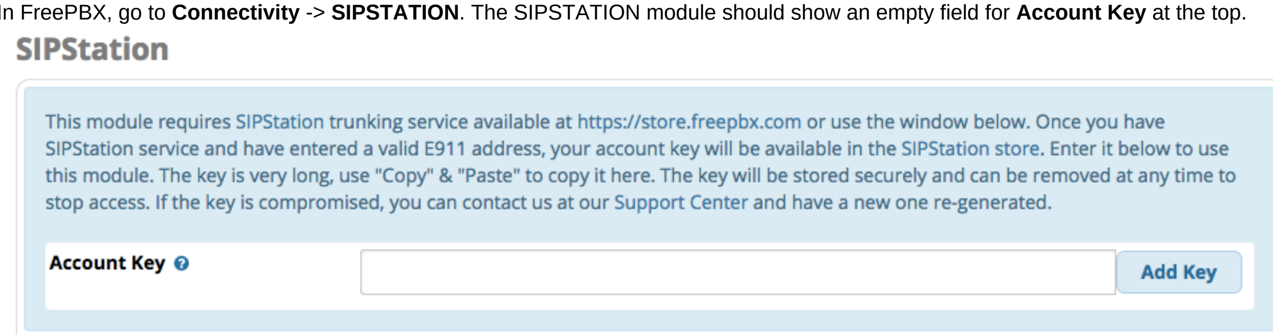
Can't find what you are looking for in the store, or need some additional help with the ordering process? We may be able to help. You can use our special order form to make your request.

For full instructions on special-ordering, please see our wiki Special Ordering Local DIDs and Toll-Free Numbers.

Manually Adding SIPStation Trunks to FreePBX

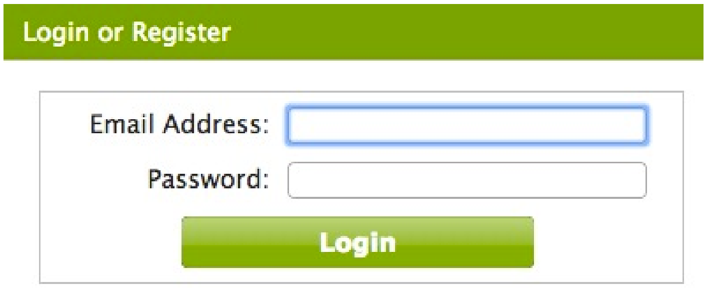
These instructions assume you have purchased a SIPStation trunk and DID, and you are using a regular SIPStation account (not a Free Trial). If you already have at least one trunk and DID in SIPStation, all you will need to do is find your FreePBX module keycode and paste it into the SIPStation module for the "account key." Here is how to do that:

- In FreePBX, go to **Connectivity** -> **SIPSTATION**. The SIPSTATION module should show an empty field for **Account Key** at the top.

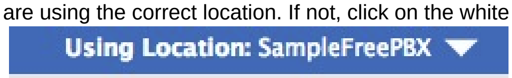


- The SIPSTATION module gives you access to the SIPStation Store without leaving the page. You will use the store to find your account key. In the SIPSTATION module, click on the **SIPSTATION Store** tab. (Alternatively, you could visit the store separately at http://www.sipstation.com.)

- Log into your SIPStation account.

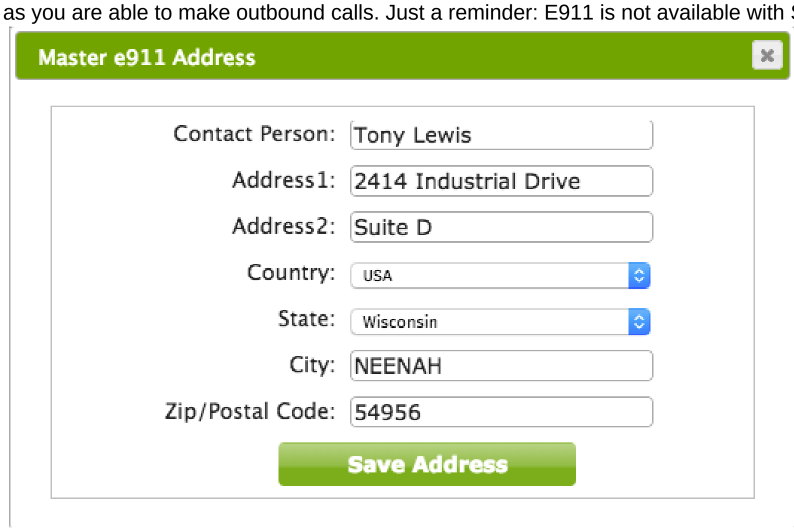


- Note: SIPStation allows you to create multiple locations within the same account. Check the blue bar at the top of the page to ensure you are using the correct location. If not, click on the white triangle to bring up a list of other locations to choose from.

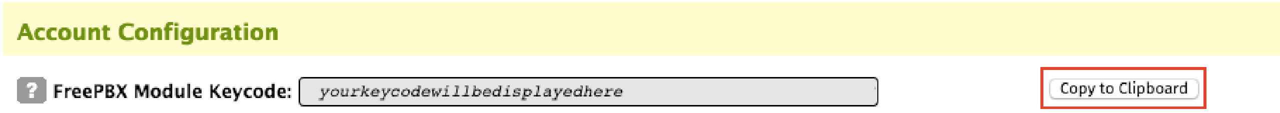


- Click on **My Account** in the green main menu near the top of the page.

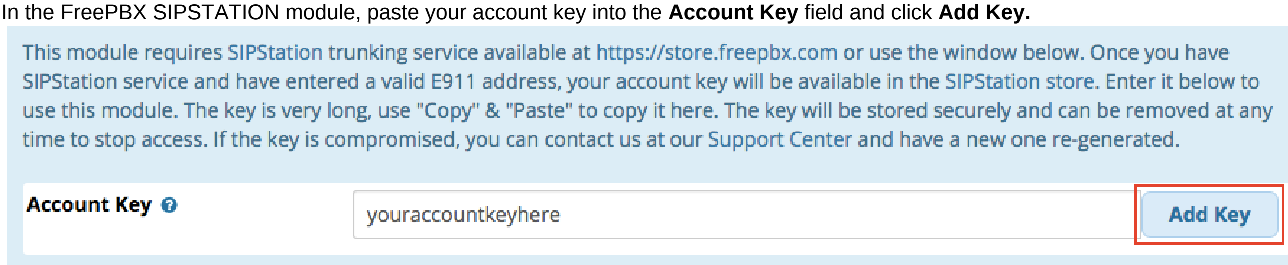
- Provide a valid E911 address if you have not already done so. You are required to set this up in order to use the SIPStation service. It is your responsibility to provide a valid U.S. or Canadian address and verify that E911 is working. You can verify this by dialing 933 as soon as you are able to make outbound calls. Just a reminder: E911 is not available with SIPStation Free Trial accounts.



- In My Account, scroll down to find the Account Configuration section.
- You should find your FreePBX Module Keycode here. Click the **Copy to Clipboard** button. This keycode will later help you easily auto-configure trunks and routes in FreePBX.



- In the FreePBX SIPSTATION module, paste your account key into the **Account Key** field and click **Add Key.**



- The page should reload. Click the **Apply Config** button at the top.
- If you scroll down the page, you should see that several settings have been set up for you, including a few basic routes. Your DID(s) will be shown at the bottom with the inbound destination set to Sipstation-DID Verification by default.

Reviewing your SIPStation Account Services

The SIPSTATION module allows you to conveniently check several settings without leaving the module. You can set a global failover number here. Your Asterisk server would send incoming calls to the global failover number in case it is unable to connect to SIPStation. You can also see and change the status of international calling, outbound fax, and SMS services.

The following instructions assume that you have set up SIPStation trunks as described earlier, and that you have navigated to the SIPSTATION module by going to **Connectivity** -> **SIPSTATION**.

To edit the global failover number from within the SIPSTATION module:

- Click on the **Local Settings** tab.

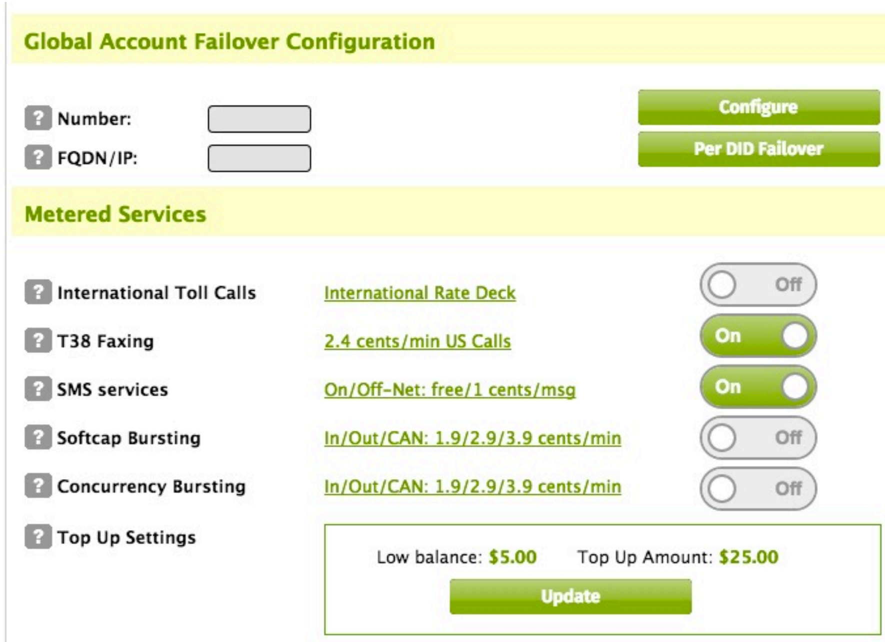
- Scroll down to the **Account Settings** section.
- Enter the desired failover number into the Global Failover Number field.

- Click the **Update Account Info** button.

To enable/disable account services such as international calling, outbound fax, and SMS Support from within the SIPSTATION module:

- Within the SIPSTATION module, click the **SIPSTATION Store** tab.

- Log in if you haven't already.
- Click on **My Account** and then scroll down to view your Metered Services.
- You can turn these services on or off by clicking the **ON/OFF** switches. Notice that above your Metered Services section, you can also adjust failover settings here.



Reviewing your E911 Configuration

In the Local Settings tab of the SIPSTATION module, you can see your E911 Master Location settings. Your E911 caller ID number is shown here.

The following instructions assume that you have set up SIPStation trunks as described earlier, and that you have navigated to the SIPSTATION module by going to **Connectivity** -> **SIPSTATION**.

Remember, a regular SIPStation account (not Free Trial) will not work unless you set up at least one DID with e911 information. You may optionally set up e911 on additional numbers for an additional monthly fee. Please confirm the accuracy of the information and ensure E911 is working by dialing 933. DO NOT DIAL 911 to test the service. Please view our "Additional e911 Addresses" wiki for more information.

- Click on the **Local Settings** tab.

- Scroll down to the E911 Master Location section.
- You can view and edit the master E911 name and address. If you need to make changes to the Master e911 caller ID, name, and/or address, enter your changes here, click the **Update Account Info** button, then click the **Apply Config** button. Please note, there may be up to a 15-minute delay before your settings are actually applied. You can test the accuracy of the information by dialing 933.

E911 Caller ID	9209449990
Name	Tony Lewis
Address 1	2414 INDUSTRIAL DR
Address 2	STE D
City	NEENAH
State	WI
Zip	54956

Update Account Info

Please see our SIPStation Trunking wiki for more information.

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Setting Up Inbound Routes

When a call comes into your system from the outside, it will usually arrive along with information about the telephone number that was dialed (also known as the "DID") and with the Caller ID of the person who called. The Inbound Routes module works together with most of the other modules in FreePBX. Calls come into your system on trunks that are configured in the Trunks module. The Inbound Routes Module then tells FreePBX/Asterisk where to send these calls. Calls can be sent to a variety of destinations, including Extensions, Ring Groups, Queues, IVRs, Time Conditions, Feature Codes, DISAs, Conferences, etc., all of which are configured in their own modules.

Setting Up an Inbound Route to an Extension with SIPStation

The next steps are for SIPStation users who would like to create an inbound route to an extension. The SIPSTATION module will take care of many settings automatically. If you are not a SIPStation user, you can skip this section and go to the instructions for setting up an inbound route to an extension manually.

If you are not already within the SIPSTATION module, go to **Connectivity** -> **SIPSTATION** and check that you are viewing the **Local Settings** tab.

- Scroll to the bottom of the page and you should see a table showing your DID(s).
- In the **Description** column, enter a description for this route. *In our example we are calling it "Main DID."*
- In the **Route To** column, select **Extensions** in the top menu, and then choose an extension in the bottom menu.

DID	Set E911	Failover Num	Description	Route To	Set CID	Emergency CID	Last Call
9203831234	Modify		Main DID	Extensions <4000> John Doe	None	Unchanged	N/A

- Click the **Update DID Configurations** button and then click the **Apply Config** button when finished.
- Your new route should now show up in the Inbound Routes module. If you go to **Connectivity** -> **Inbound Routes**, you should see your new inbound route in the list.

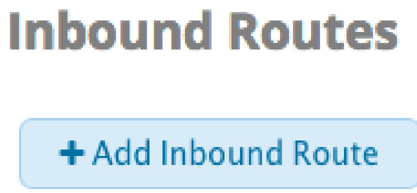
DID	CID	Description	Destination	Actions
9203831234	Any	Main DID	Core: <4000> John Doe	

Setting Up an Inbound Route to an Extension Manually

If you are not using SIPStation, you will need to set up inbound routes manually.

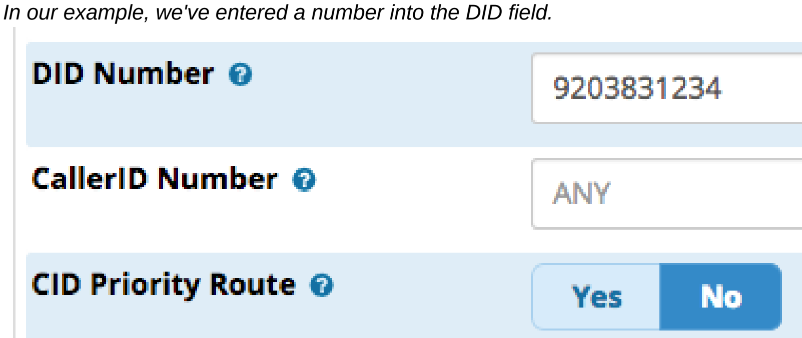
- Go to **Connectivity** -> **Inbound Routes**

- Click the **Add Inbound Route** button.

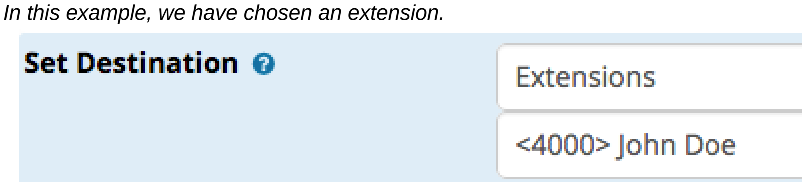


- Give the new route a description.

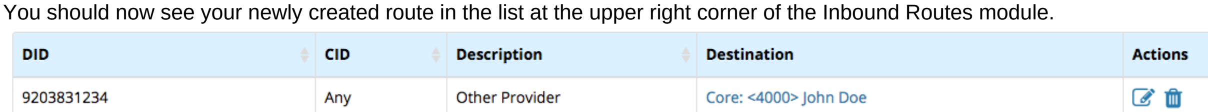
- The PBX allows two specific types of inbound routing: DID & CID Routing. These two routing methods can be used on their own or together. Enter a number in the **DID Number** and/or **CallerID Number** fields if desired. Leaving both fields blank will create a route that matches all calls. If you set the **CID Priority Route** to Yes, CID will only be considered a priority if there is no entry in the DID field.
In our example, we've entered a number into the DID field.



- You can configure several other options. Please see our Inbound Route User Guide for a description of the options, and/or view the pop-up tooltips.
- At the bottom, choose the destination for this inbound route. You must set this in order for your new inbound route to function properly.
In this example, we have chosen an extension.



- When finished, click "Submit" and then click "Apply Config."
- You should now see your newly created route in the list at the upper right corner of the Inbound Routes module.



DID	CID	Description	Destination	Actions
9203831234	Any	Other Provider	Core: <4000> John Doe	

Please see our Inbound Routes Module wiki for more information.

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Setting Up Outbound Routes

The Outbound Routes Module is used to tell your FreePBX/Asterisk system which numbers your phones are permitted to call, and which trunk to send the calls to. Without setting up outbound routes, you will not be able to make calls outside your system.

If you are a SIPStation user, a few basic outbound routes are automatically configured for you when you set up SIPStation trunks. You should see these defaults:

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

You can modify the settings as needed or create new routes. See the examples below for typical configurations.

If you are not a SIPStation user, you can set up outbound routes manually as described in the instructions below.

A typical configuration will include an emergency route for 911 calls and another route for ordinary calls. You might also wish to configure routes for interoffice calls, international calls, and other special circumstances.

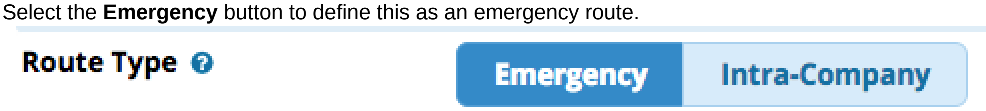
> The order of your routes is very important. An emergency route needs to be kept first (on top of the list).

Creating an Emergency Outbound Route Manually

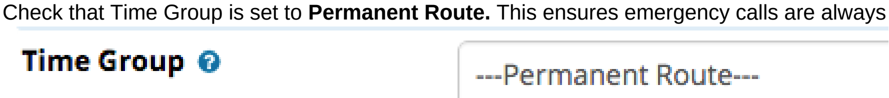
An emergency route enforces the use of a device's emergency Caller ID (CID), if set.

- Go to **Connectivity** -> **Outbound Routes**
- Click the **Add Outbound Route** button.
- You should be viewing the **Route Settings** tab.
- Give the route a name. Spaces are not allowed.
  In our example, we've named this emergency route "E911-Leave-First" to remind ourselves to keep this route at the top of our list. Note, this is also the name that SIPStation uses for an automatically created emergency route.

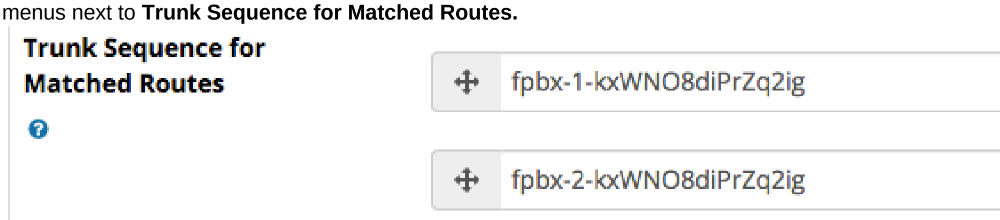
- Select the **Emergency** button to define this as an emergency route.



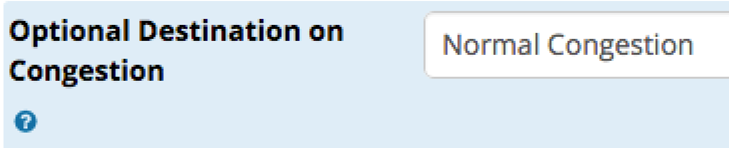
- Check that Time Group is set to **Permanent Route.** This ensures emergency calls are always allowed.



- Select which trunks this outbound route will use, and in what order. Choose the desired trunk(s) from the drop-down menus next to **Trunk Sequence for Matched Routes.**

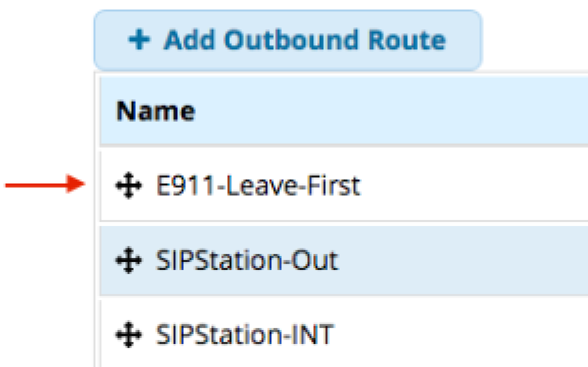


- Check that the **Optional Destination on Congestion** drop-down menu is set to **Normal Congestion.** The default behavior for this setting is to play the "all circuits busy" recording or another option configured in the Route Congestion Messages module if installed.



- Click the **Dial Patterns** tab. Enter the following dial patterns (use the **+** button to add new lines):

  Note: "933" is the e911 address verification service. Dialing 933 will allow us to test the functionality of our emergency route as well as verify our CID and e911 address. We have also added three patterns with prefixes in case someone dials these extra digits when trying to make an emergency call. The system will strip the prefix from the number that is sent to the trunk.
- Click the **Submit** button and then click the **Apply Config** button.
- IMPORTANT: Verify your emergency route is first in the list. If you create other routes, always double-check that the emergency route remains first. If it is not first, drag it to the top of the outbound routes list.



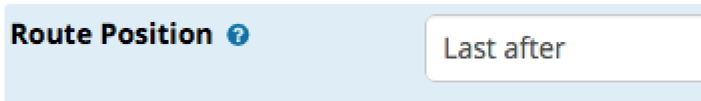
Creating a Local & Toll-Free Outbound Route Manually

In our example we are going to create a single route for local and toll-free numbers, combined.

- Go to **Connectivity** -> **Outbound Routes.**
- Click the **Add Outbound Route** button.
- You should be viewing the **Route Settings** tab.
- Give the route a name. Spaces are not allowed.

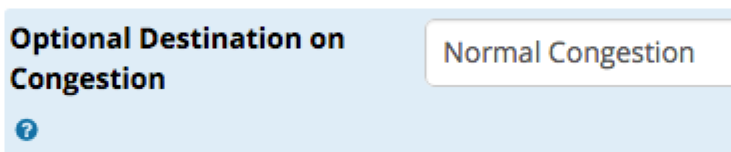
- For **Route Type**, leave both buttons de-selected (light blue in color). This is neither an emergency route nor an intra-company route. This route is used for for non-emergency local and toll-free outbound calls.

- For **Route Position**, select **Last after.** This will ensure your emergency route remains above this route. You can adjust the position of this new route later if needed, but keep your emergency route as #1 in the list.

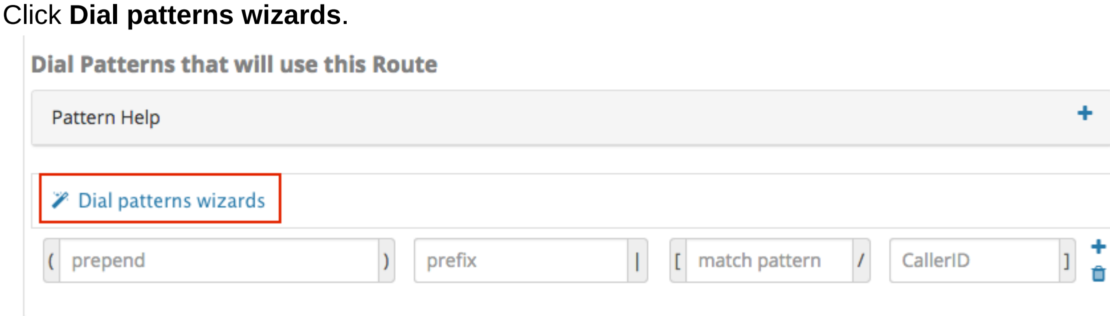


- Select which trunks this outbound route will use, and in what order. Choose the desired trunk(s) from the drop-down menus next to **Trunk Sequence for Matched Routes.**

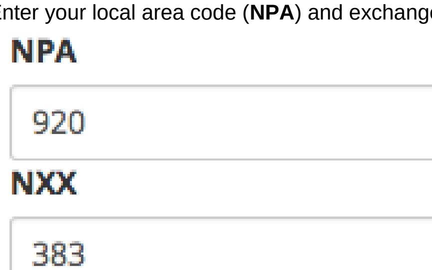
- Check that the **Optional Destination on Congestion** drop-down menu is set to **Normal Congestion.** The default behavior for this setting is to play the "all circuits busy" recording or another option configured in the Route Congestion Messages module if installed.



- Click the **Dial Patterns** tab.
- Click **Dial patterns wizards**.

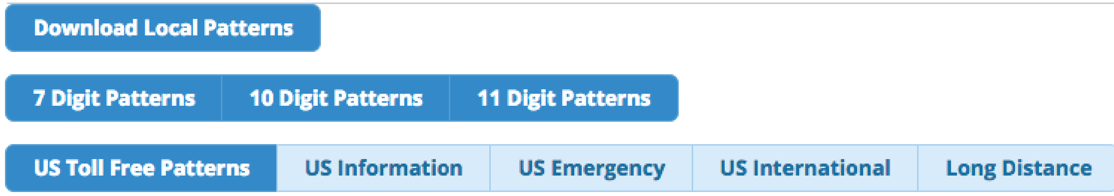


- Enter your local area code (**NPA**) and exchange (**NXX**).



- Select the following buttons: **Download Local Patterns, 7 Digit Patterns, 10 Digit Patterns, 11 Digit Patterns,** and **U**

S Toll Free Patterns. De-select the other buttons.



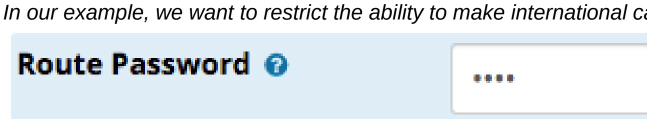
- Click the **Generate Routes** button. The dial pattern wizard window should close, and the system should create several local and toll-free dial patterns for you.

- Enter any additional dial patterns as desired. One common option is to create dial patterns with "9" as a prefix that will be stripped, to accommodate users who are accustomed to dialing 9 to reach an outside line.
- Click the **Submit** button and then click the **Apply Config** button.

Setting Up an International Outbound Route Manually

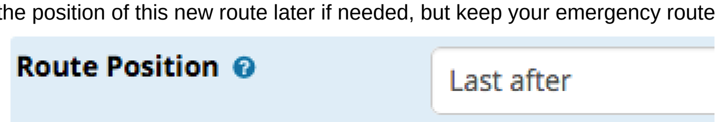
- Go to **Connectivity** -> **Outbound Routes**
- Click the **Add Outbound Route** button.
- Give the route a name. Spaces are not allowed.

- Optional: If you would like to restrict access to this route, enter a Route Password.
  In our example, we want to restrict the ability to make international calls, so we have entered a password.

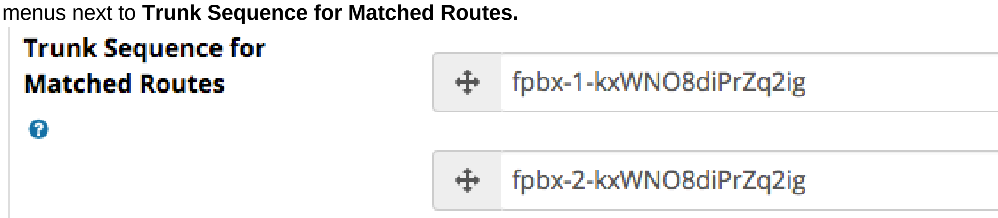


- For **Route Type**, leave both buttons de-selected (light blue in color). This is neither an emergency route nor an intra-company route.

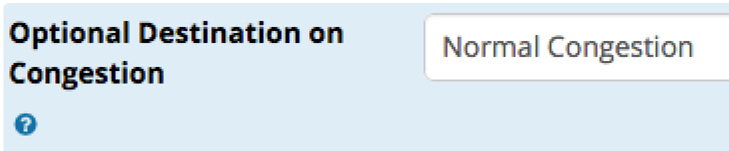
- For **Route Position**, select **Last after.** This will ensure your emergency route remains above this route. You can adjust the position of this new route later if needed, but keep your emergency route as #1 in the list.



- Select which trunks this outbound route will use, and in what order. Choose the desired trunk(s) from the drop-down menus next to **Trunk Sequence for Matched Routes.**



- Check that the **Optional Destination on Congestion** drop-down menu is set to **Normal Congestion.** The default behavior for this setting is to play the "all circuits busy" recording or another option configured in the Route Congestion Messages module if installed.



- Click the **Dial Patterns** tab.
- Enter the following dial pattern in the **match pattern** field: "**011.**" (Include the period.) The period (.) is a wildcard character. When a user dials anything beginning with 011, the number will be recognized as an international number and will use this route.

- Enter any additional dial patterns as desired. For example, you might add a second 011. dial pattern using a "9" as a prefix that will be stripped, in order to accommodate users who are accustomed to dialing 9 to reach an outside line.
- Click the **Submit** button and then click the **Apply Config** button.

Changing the Order of Routes

When making an outbound call, the system will for the first matching dial pattern by working down the list of routes. The first item on the list will be checked first, then the next, and so on. If the system finds a match, it does not continue to look for a "better" match. Keep this in mind when deciding the order of your routes.

> Remember, keep the emergency route above all of the others in the list.

- Go to **Connectivity** -> **Outbound Routes**. Your route list is displayed.
- To change the order, click on the arrow button (

) and drag the route to a new position.
  - Alternatively, if you are not able to use drag-and-drop, you can click on the edit button (

) for the route and then select a new position from the **Route Position** menu. Click **Submit** after using this method.
- After you have made your changes, click the **Apply Config** button.

Please see our Outbound Routes Module wiki for more information.

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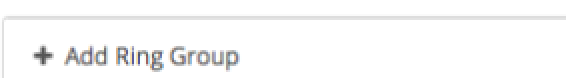
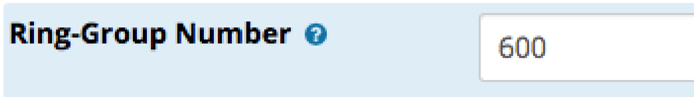
Setting Up a Ring Group

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**.

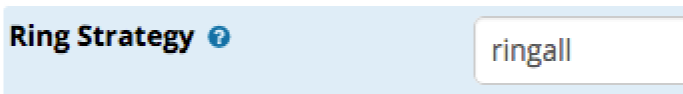


- Enter a **Ring-Group Number**. This will be the number a user may dial to ring the phones in the ring group. Be sure it does not match any existing extensions or feature codes. The default is 600.



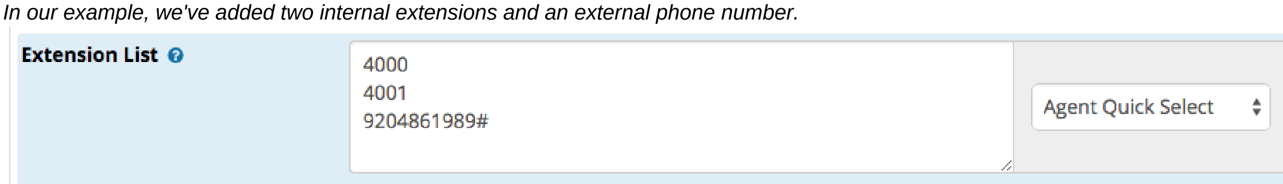
- Enter a **Ring Group Description.** This can be a "friendly name" and spaces are acceptable.

- 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.

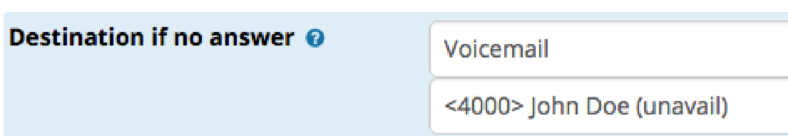


- Set the **Ring Time**, in seconds (default=20, maximum=300 seconds).

- 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.

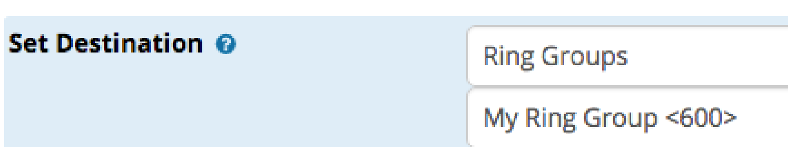


- At the bottom of the page, set a destination to which to route the inbound call in case no one answers. For example, you may wish to route it to a voicemail box.



- 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.



Please see our Ring Groups Module wiki for more information.

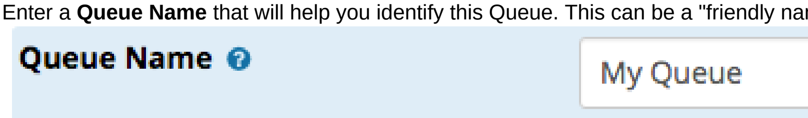
top

Setting Up a Queue

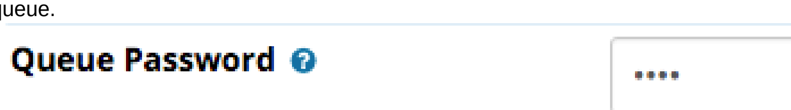
You can set up automatic call distribution with the Queues module. A queue differs from a ring group because it allows advanced call routing options and escalation rules. This wiki gives you a brief overview of some basic settings, but there are many more. To learn more about queue settings, you can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

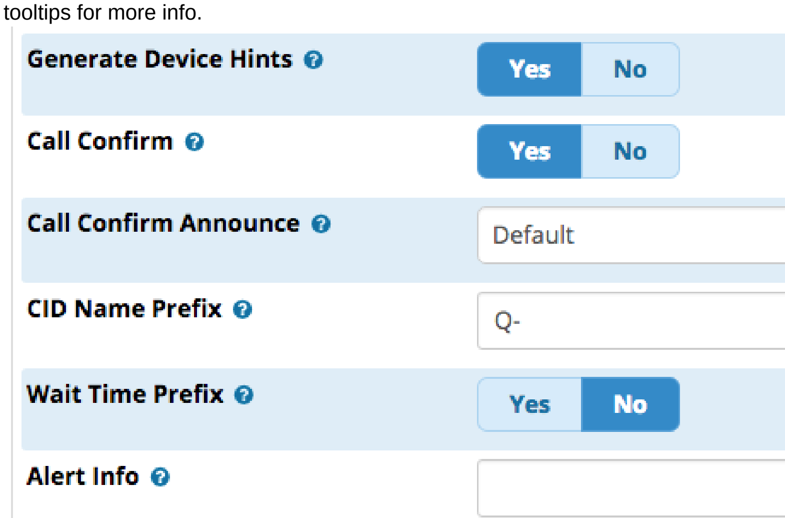
- Enter a **Queue Name** that will help you identify this Queue. This can be a "friendly name," and spaces are allowed.



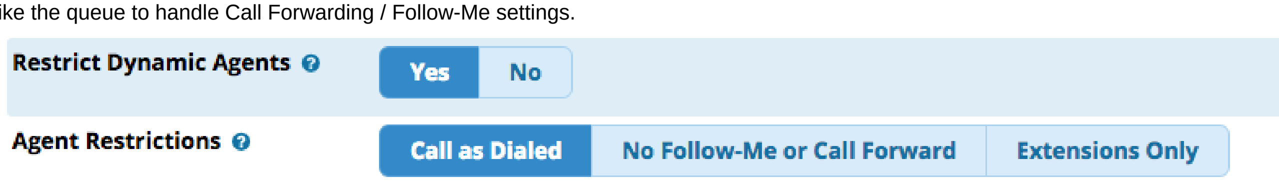
- Optionally, set a **Queue Password** (numbers only). This would require agents to enter a password before being allowed to log into the queue.



- 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.

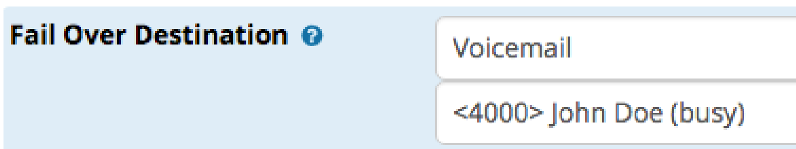


- Choose how you would like to handle agent restrictions. If you set **Restrict Dynamic Agents** to **Yes**, you can prevent unauthorized users from logging into the queue. Later, you will define a list of users who are allowed to log in. If this is set to **No**, any user who knows the queue number (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.

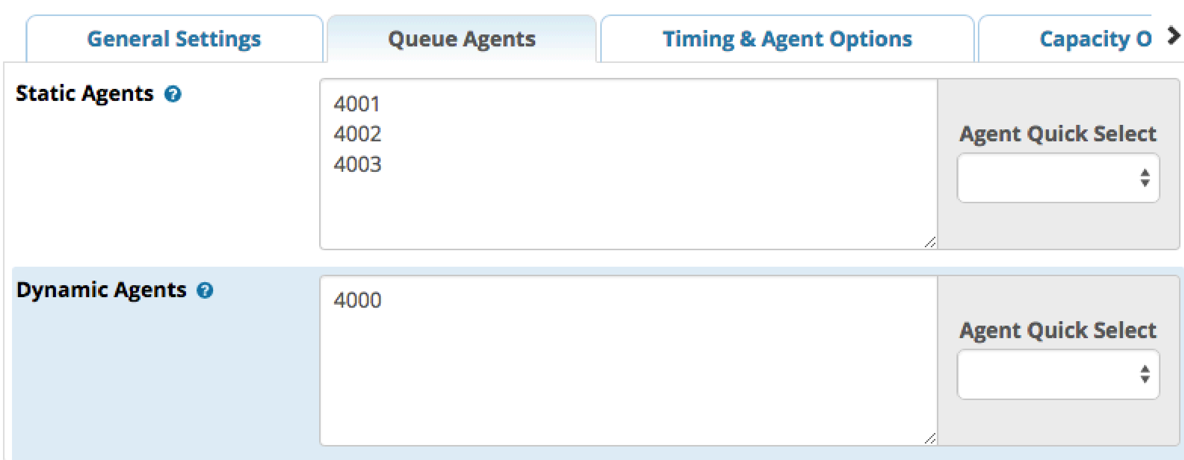


- You can configure several more options at the bottom of the General Settings section. These include autofill, skip busy agents, queue weight, music on hold class, join announcement, call recording, and whether to mark calls answered elsewhere.

- 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.
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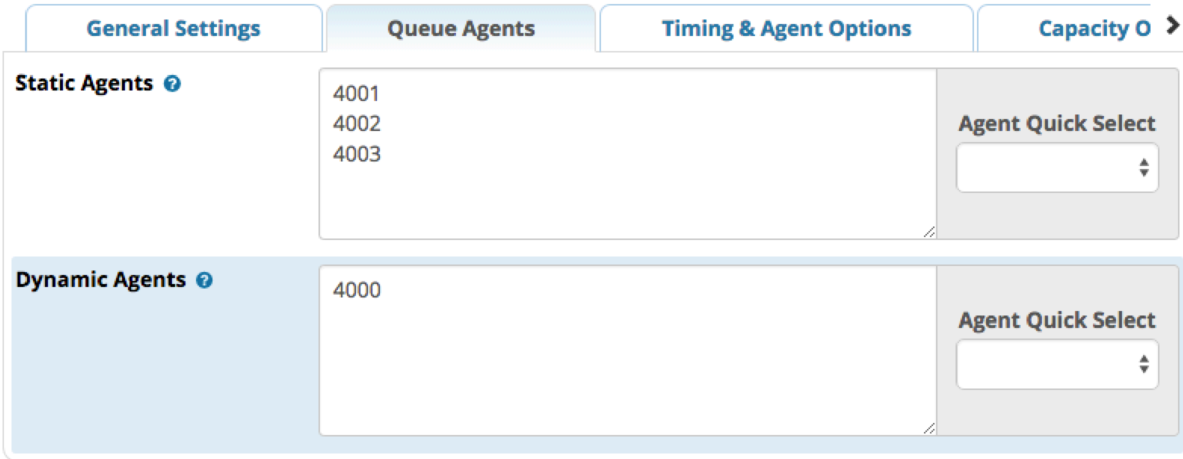
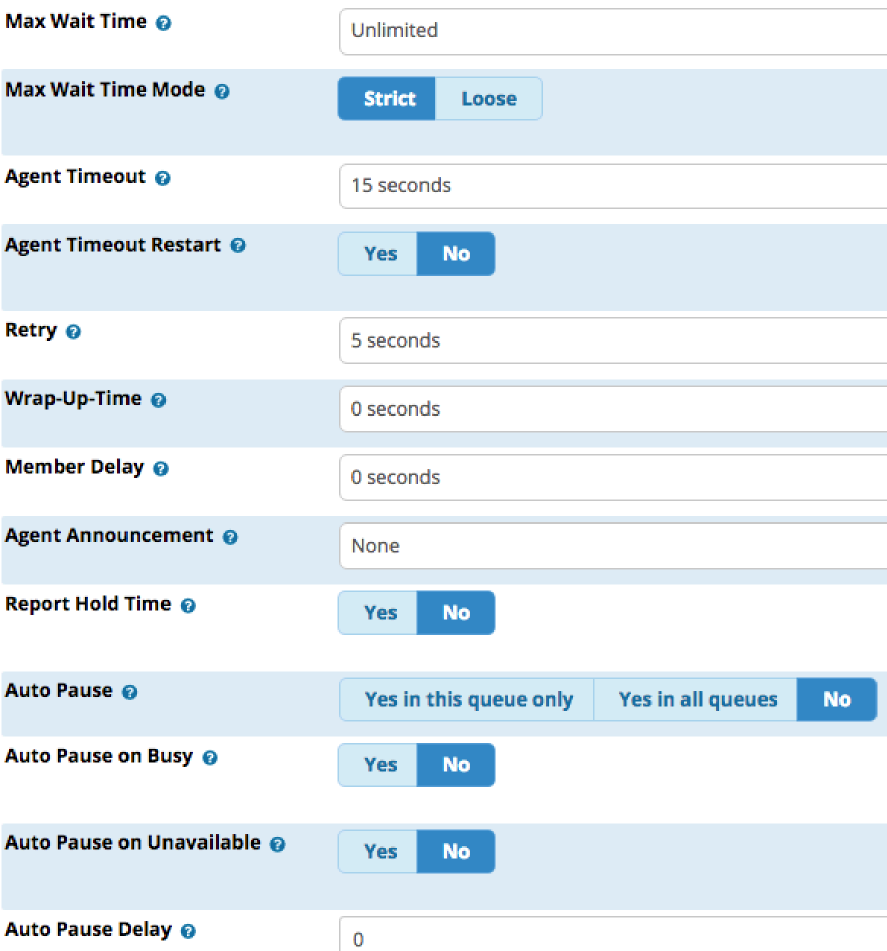
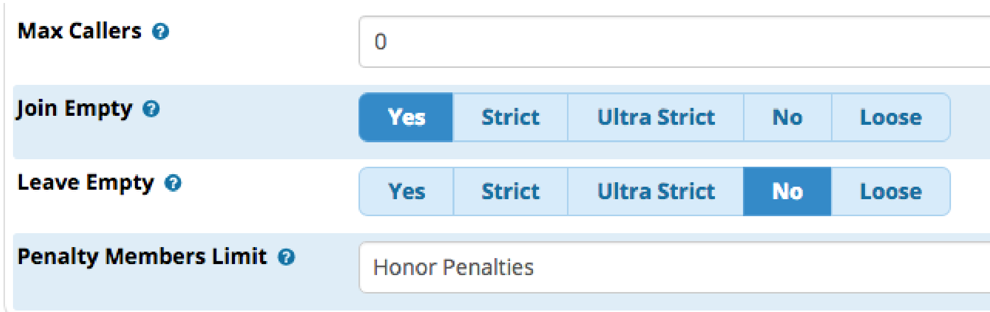


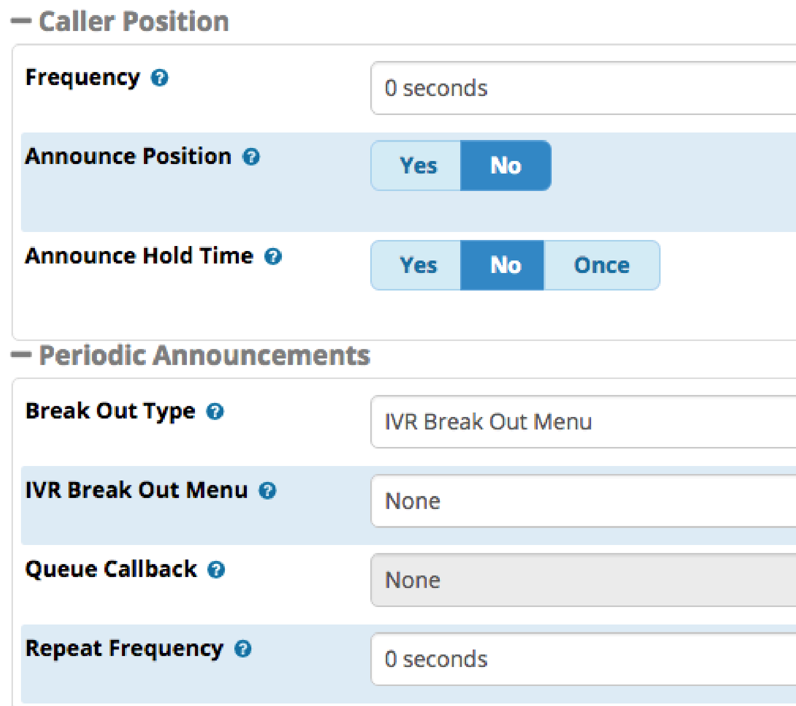
- Click on the **Timing & Agent Options** tab to configure several settings related to wait time, timeout, retry, agent pause, and more. The default values are shown below.



- 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.

Please see our Queues Module wiki for more information.

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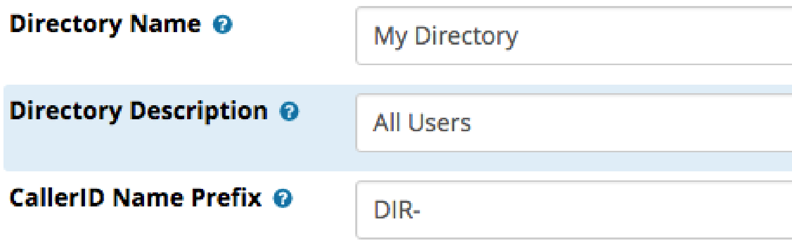
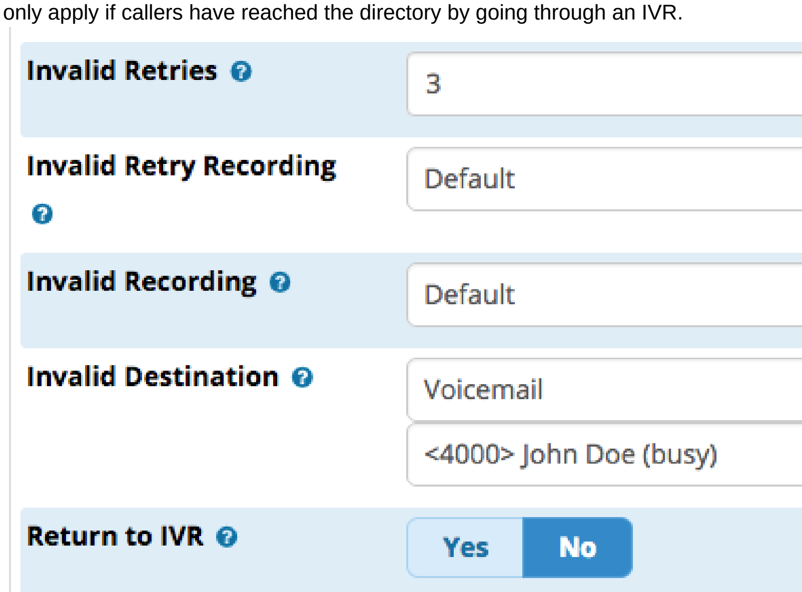
Setting Up a Company Directory

A directory gives the caller the option to dial a person's first or last name using their phone keypad in order to find and be connected to an extension. It is most commonly used as an IVR option. Your directory is not limited to internal extensions. You can add custom entries such as remote extensions, ring groups, queues, and outside numbers. You can add as many directories as you would like.

- Go to **Applications** -> **Directory**
- Click **Add Directory**
- Enter a **Directory Name**. This can be a "friendly name," and spaces are allowed. Enter a **Directory Description** to help you remember what this directory is for. If you would like a prefix to be added to the incoming caller's caller ID, you may enter it in the **CallerID Name Prefix** field.



- Several settings determine how to handle invalid entries made by callers. When a caller enters something into their keypad that cannot be found in the directory, that is considered an invalid entry. For example, the caller begins typing the last name "Thomas" but there is no one with that name in the directory.
    - **Invalid Retries:** How many invalid entries are allowed before a caller is sent to the invalid destination.
    - **Invalid Retry Recording:** Message played to a caller upon an invalid entry, asking them to try again.
    - **Invalid Recording:** Message played to a caller when they have reached the invalid retry limit, before sending them to the invalid destination.
    - **Invalid Destination:** Where to send a caller if they have reached the invalid retry limit.
    - **Return to IVR:** Whether to send a caller back to the parent IVR instead of sending them to an invalid destination. This would only apply if callers have reached the directory by going through an IVR.

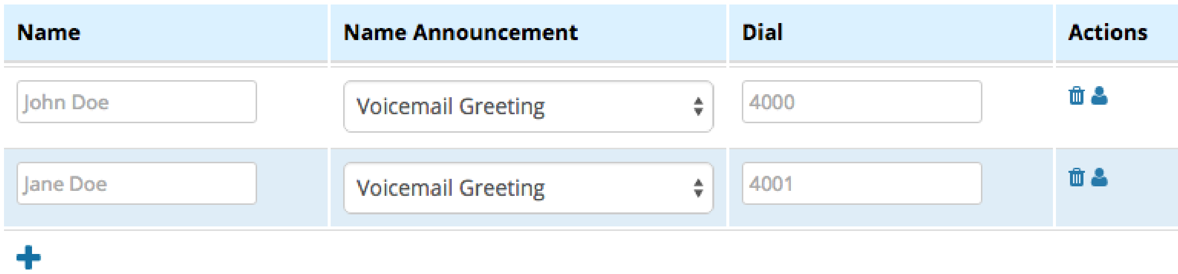


- Optional: Enable **Announce Extension** if you would like the system to tell callers the number of the extension after it is found in the directory. This is useful if you'd like your callers to know the exact extension number so they can direct-dial it in the future without spelling someone's name.

- At the bottom of the page, create your directory by choosing an option from the drop-down menu. You can select **All Users, Custom**, or individual extensions. Add new lines by clicking the plus sign icon (

).
    - **Name**: The person's name. Callers will need to begin dialing the person's name on their keypad, as shown here (i.e. first name first, if that is how it appears here).
    - **Name Announcement:** After a caller begins dialing someone's name, this announcement will be played to the caller upon a successful match. You can choose from the person's voicemail greeting (if an internal extension), Text to Speech, Spell Name, or a custom recording.
    - **Dial:** The extension or external number you would like the system to dial when a user selects the directory entry.

    Note: If you have chosen an existing internal extension, the name and dial fields are greyed-out because you cannot edit them here. You would need to edit the information in the Extensions module. The info shown here is tied to that module.



- Click the **Submit** button and then click the **Apply Config** button when finished making changes.
- In order for the directory to work, you will need to give callers the option to dial it (i.e. from an IVR), or you will need route inbound calls to a DID or extension to the directory. These options are controlled within the Inbound Routes module and elsewhere in other modules.

Please see our Directory Module wiki for more information.

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Creating System Recordings

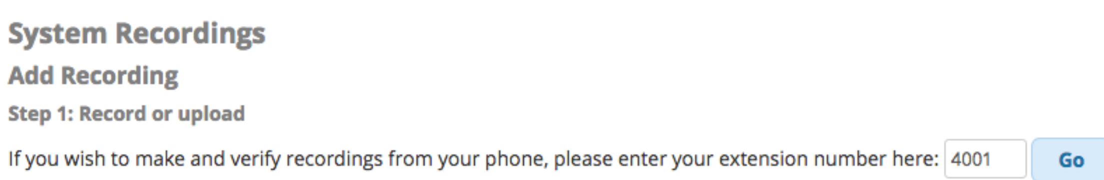
Modules such as Interactive Voice Response (IVR) are able to use custom system recordings in addition to default recordings. You can use the System Recordings module to create and save custom system recordings.

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

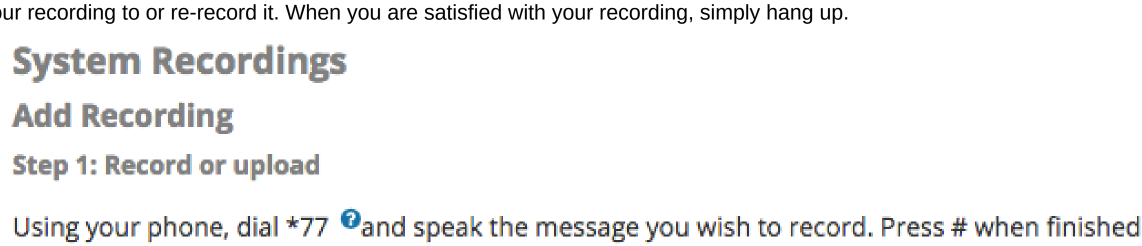
After you have created a recording, you can update it by uploading a replacement audio file or by making a new recording over the phone. You can make updates in the System Recordings module and/or enable users to dial a specific feature code that will allow them to re-record over the phone.

Creating a System Recording Over the Phone

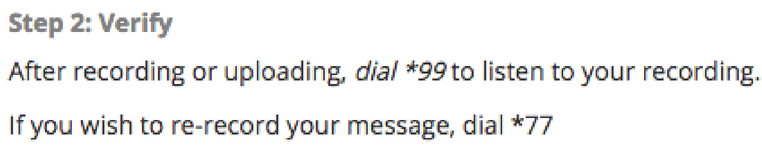
- Go to **Admin** -> **System Recordings**
- Step 1 (a): Enter the extension number of the phone that you want to use to make your recording, and click **Go**.



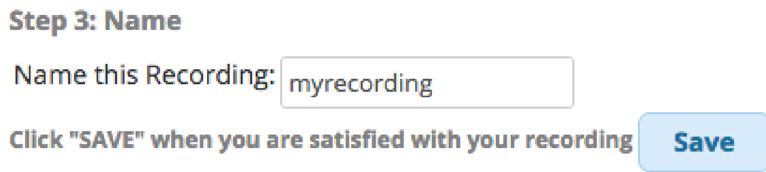
- Step 1 (b): The page will reload and display instructions that give you a code to dial. Dial this code from the same extension you entered above. You will hear a beep. Begin speaking after the beep, and press # when finished. You will be prompted to listen to your recording to or re-record it. When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

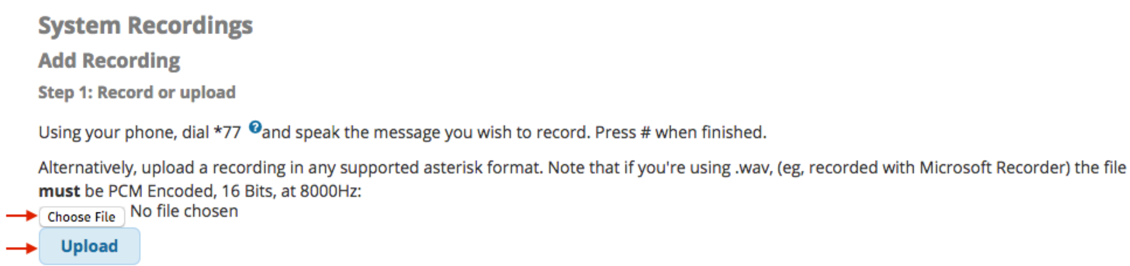


- The page will reload and you will see a message letting you know your recording has been saved.

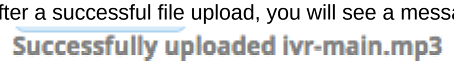
Note: There is no Apply Config button in this case. You should see your new recording show up in the list of recordings.

Uploading an Audio File from your Computer

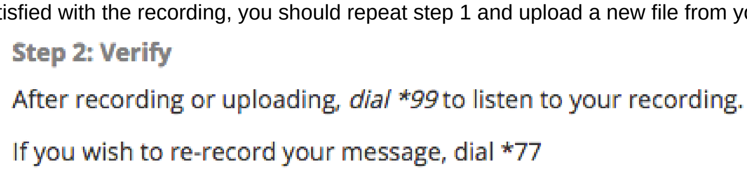
- Go to **Admin** -> **System Recordings**
- Step 1: Choose a sound file from your computer. The file must be an Asterisk-supported format. Click **Choose File**, select a file from your computer, then click **Upload**.
  - A window will pop up asking you to wait until the page reloads. Don't let this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



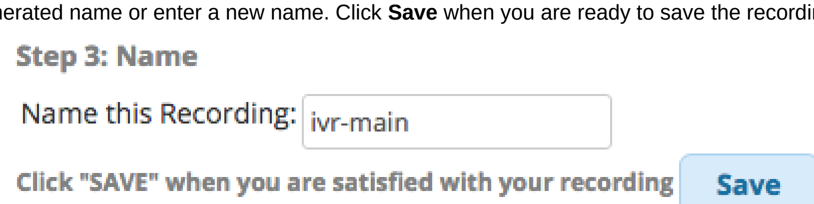
  - After a successful file upload, you will see a message confirming the upload:



- 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.

Please see our System Recordings Module wiki for more information.

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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.

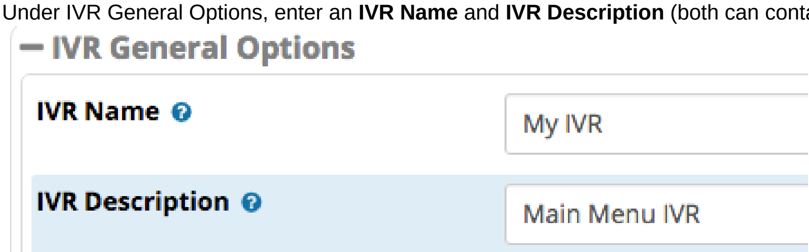
Your IVR system can be simple, with just a few options, or it can be complex with multiple options and even multiple IVRs. As a best practice, we recommend giving callers only a few options at a time, and using multiple IVRs to “nest” or “layer” various menus. For example, a single IVR might lead to 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!

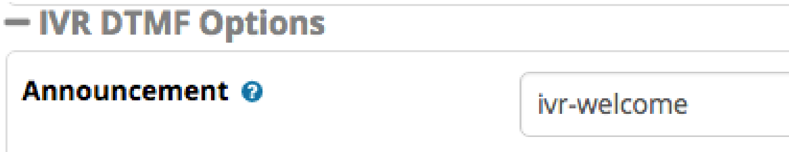
An IVR may contain "hidden" options. In other words, your recorded announcement would not tell a caller that the option exists, but a caller could dial the option. For example, you could set up a "hidden" option for VIP users to receive queue priority or for employees to dial their voicemail from an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

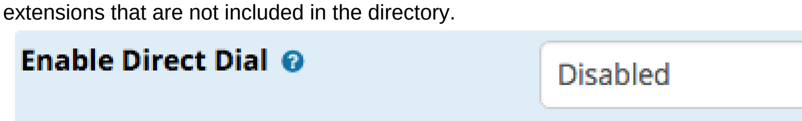
- Go to **Applications** -> **IVR**
- Click the **Add IVR** button
- Under IVR General Options, enter an **IVR Name** and **IVR Description** (both can contain spaces).



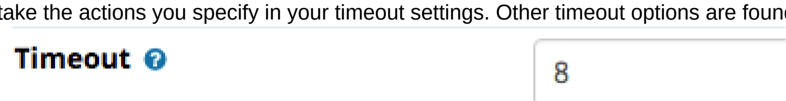
- Use the **Announcement** drop-down menu to choose the system recording you would like to play to callers when they arrive at the IVR. The default option is “None.” A setting of “None” would not be advised, since it would not explain any options to the caller. (Please see visit the System Recordings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



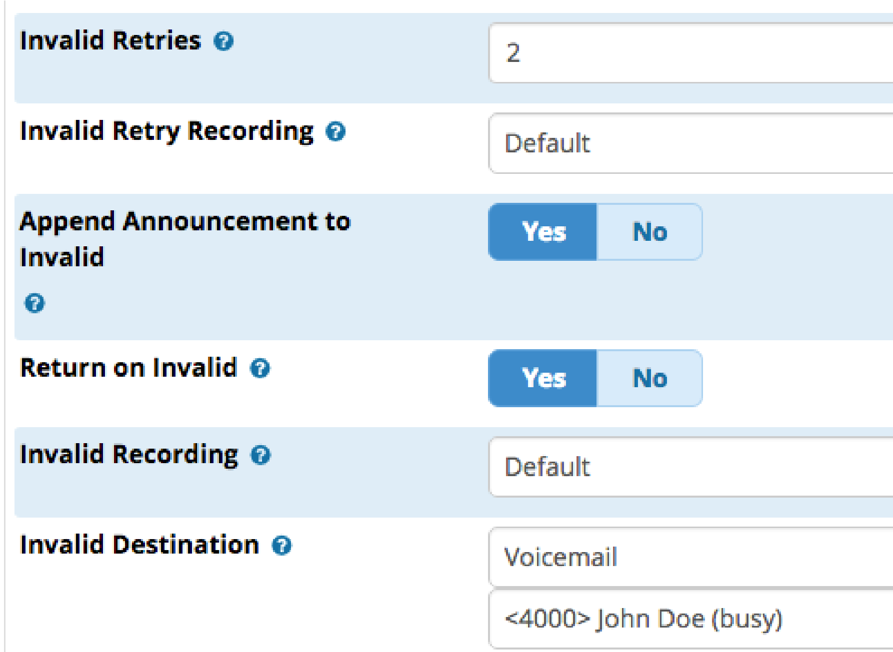
- Use the Direct Dial drop-down menu to choose whether you would like callers to be able to direct-dial an extension from your IVR. It is disabled by default. Enabling direct-dial saves your callers time if they already know a person’s extension and do not wish to navigate through an IVR. You can enable 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.



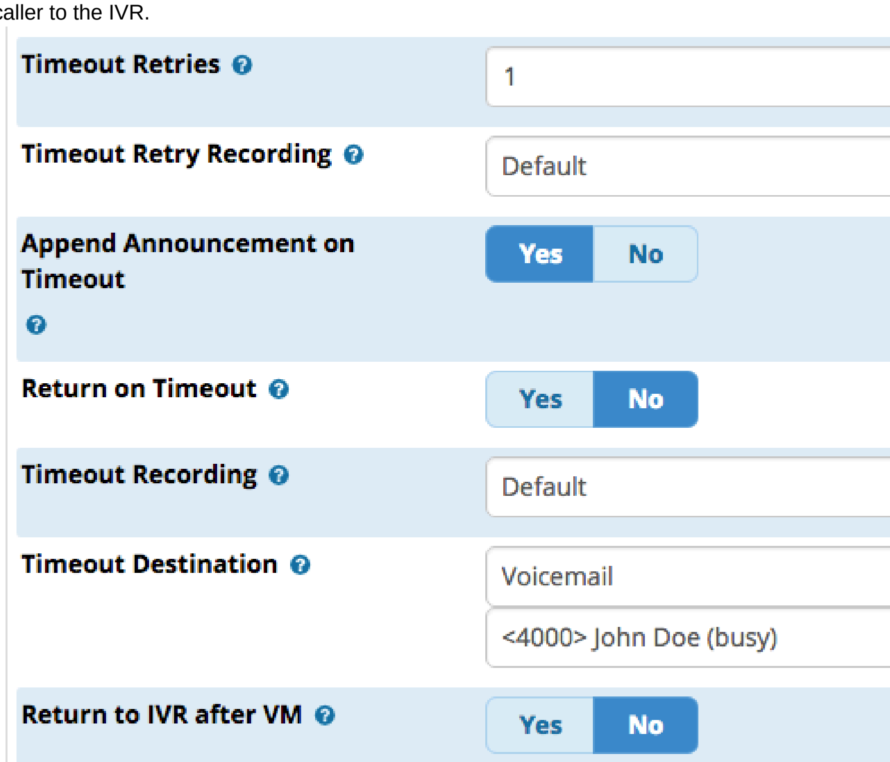
- Set the number of seconds you want to be considered a **Timeout.** If no DTMF tones are heard during this time period, the system will take the actions you specify in your timeout settings. Other timeout options are found further down the page.



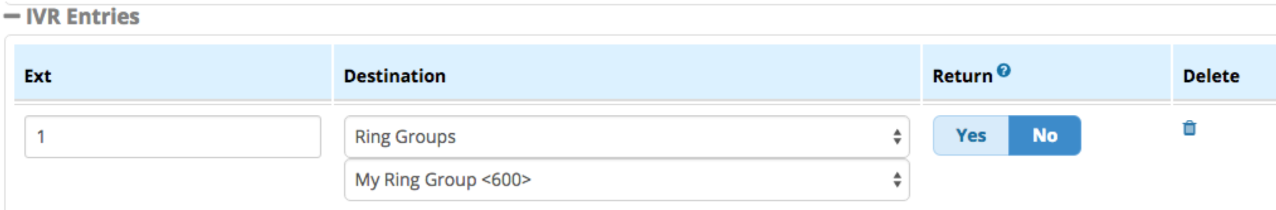
- Configure your invalid entry settings. If the caller dials something that isn’t one of the choices in your IVR, or tries to dial an extension for which direct dialing is disabled, the system considers it an invalid entry.
  - **Invalid Retries:** Number of chances you give a caller to make a correct 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.

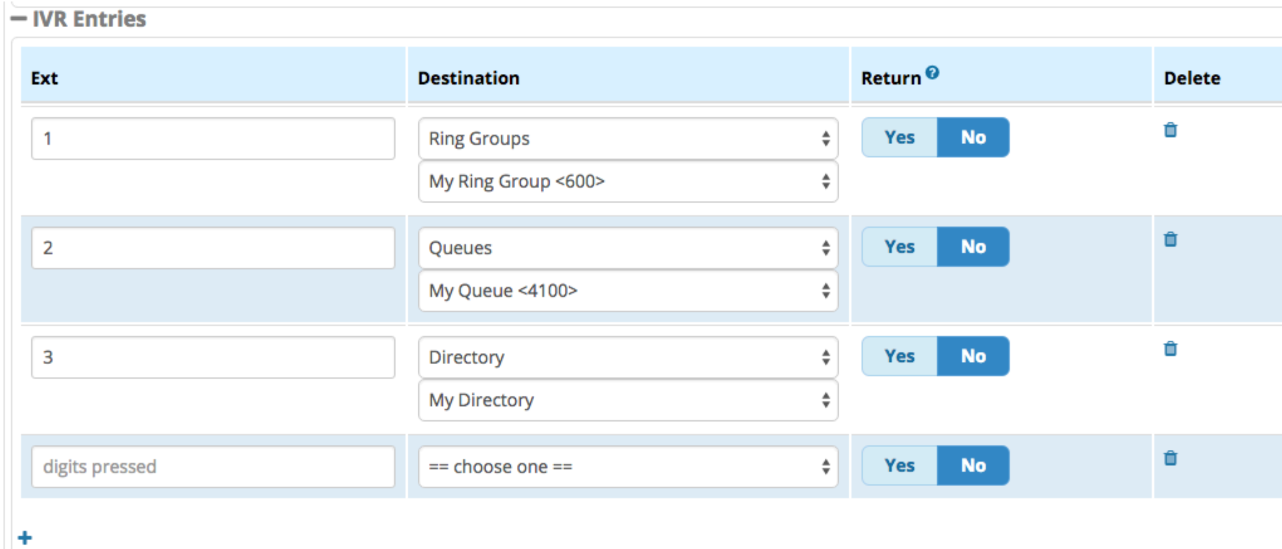


- Set up your IVR entries. These are the options that callers can dial. The **Ext** field is the number the caller needs to dial in order to reach the specified destination. For example, if your IVR announcement tells callers to "Press 1 for sales," you would enter "1" in this field. Choose the **Destination**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

Please see our IVR Module wiki for more information.

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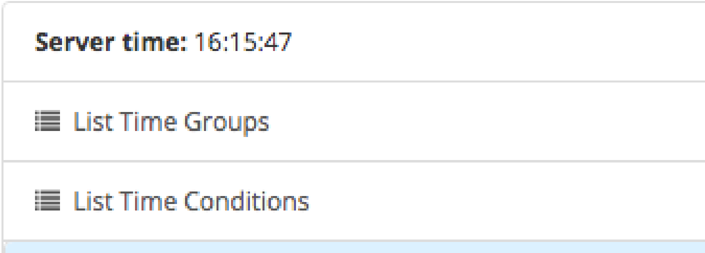
Setting Up a Time Group

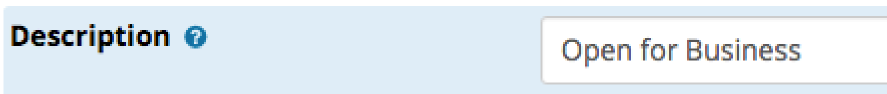
A Time Group defines periods of time that a module such as Time Conditions or Outbound Routes will use. For example, you might create a time group that defines your open business hours.

> 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.

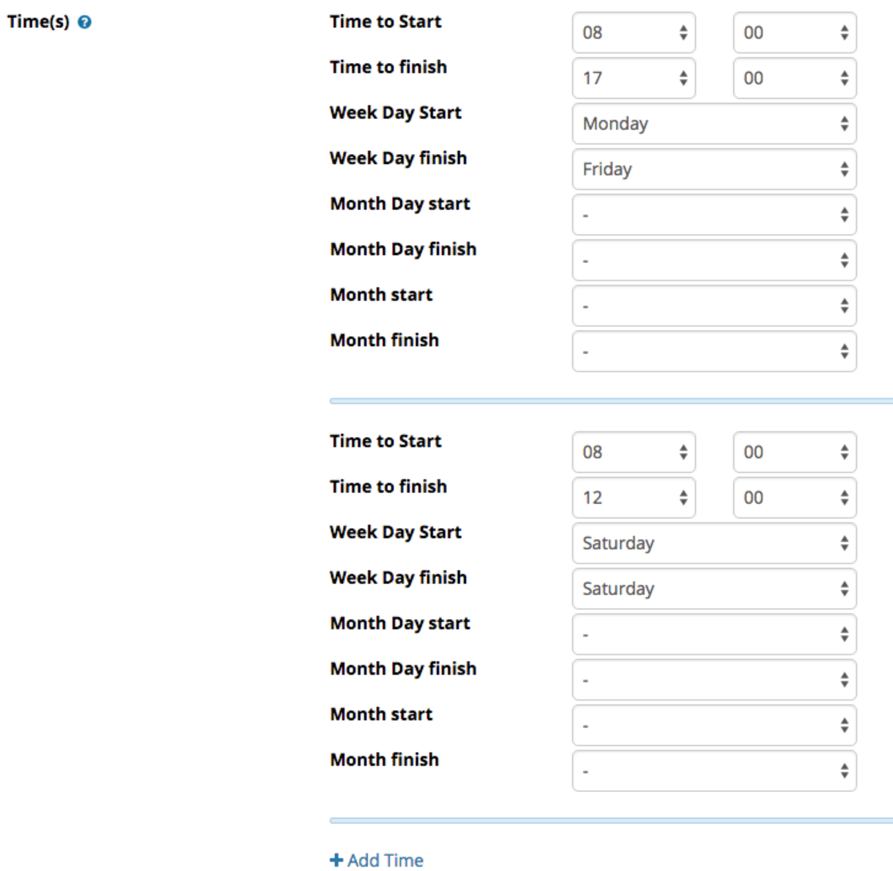


- Click **Add Time Group**.
- Enter a **Description** for your time group. In our example, we will be creating a list of times that our business is open.

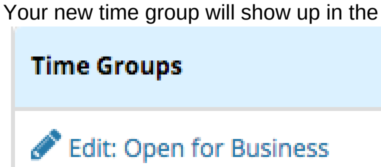


- Define your set(s) of start and finish times, days of the week, days of the month, and months as necessary. If you need to define more than one set of times, click **Add Time.**

  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.



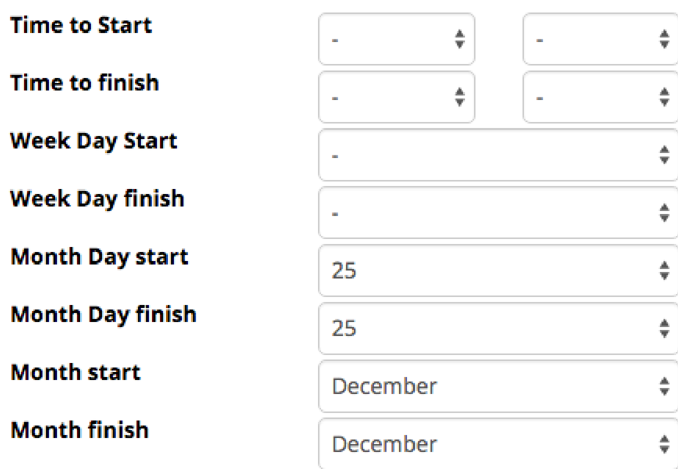
- After you have made your time selections, click the **Submit** button and then click the **Apply Config** button.
- Your new time group will show up in the time group list.



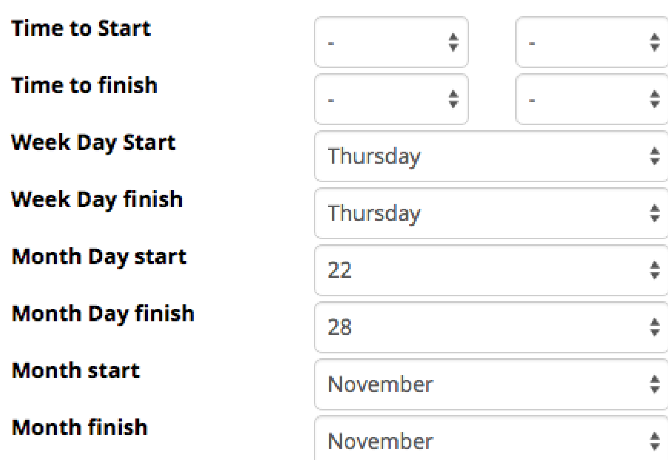
Excluding Specific Time Periods, Such as Holidays

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 may of these types of these all-day “exceptions” as you wish. We have more examples of holidays in our Time Groups and Sample Configurations wiki.

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

Please see our Time Groups Module wiki for more information.

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Setting Up a Time Condition

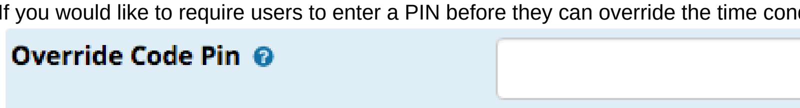
A time condition is a destination that checks the current time against a time group, and then routes calls to one of two destinations based upon the result. You need to create a time group first in order for a time condition to work. Otherwise your time condition wouldn’t have anything to check. Please 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.

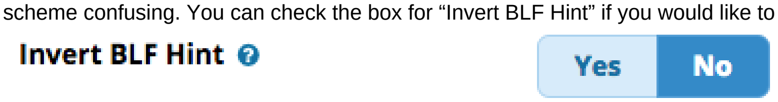
> Your time condition will only affect call routing if you define it as a destination within another module, such as inbound routes, extensions, IVR, etc.

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

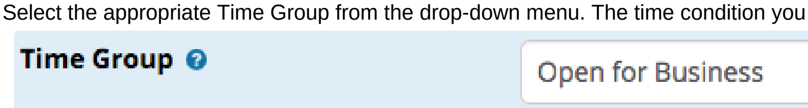
- If you would like to require users to enter a PIN before they can override the time condition, enter it in the Override Code PIN field.



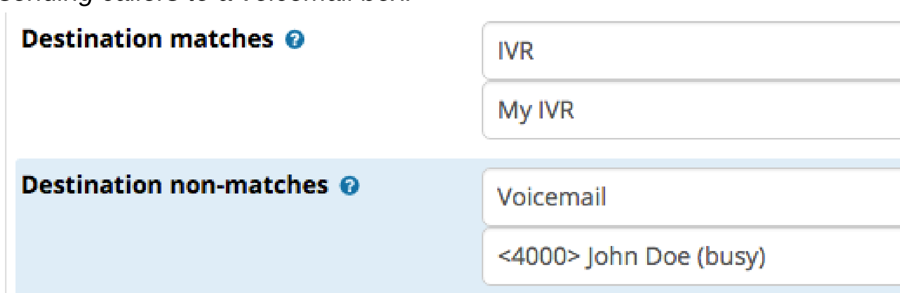
- If you set up a Busy Lamp Field (BLF) on your phone to monitor the status of your time condition, its default behavior will be to show an "INUSE" status when your time condition is **not matched**. In other words, if your time group defines your open business hours, the BLF would show the "INUSE" status outside of business hours. Depending upon the phone being used, some users may find the color scheme confusing. You can check the box for "Invert BLF Hint" if you would like to reverse the behavior of the BLF.



- 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.

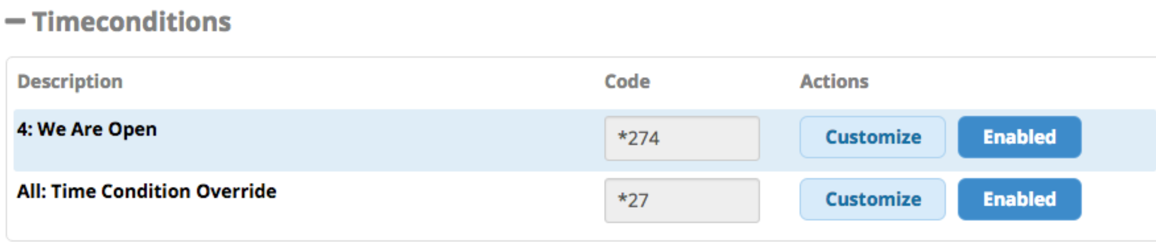


- Click the **Submit** button and then click the **Apply Config** button.

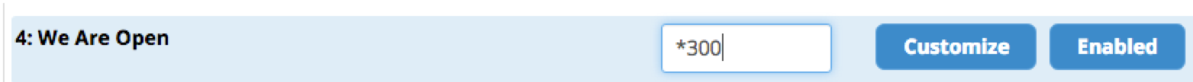
Setting Up a BLF for a Time Condition

When you create a Time Condition, the system will generate a feature code for it that will let you manually override the time condition. The feature code can also be used when setting up a BLF for the time condition.

- To view the feature code, go to **Admin** -> **Feature Codes** and scroll down to the **Timeconditions** section. You should find the name of your time condition along with a feature code. Be sure the feature code is **Enabled.**
  *In this example it is *274.*



- If you would like to change the feature code from the auto-generated code, you can click the **Customize** button to enable editing the field. Make your edits and be sure the feature code is **Enabled**. Then click the **Submit** button followed by **Apply Config**.
  *In this example we've changed it to *300.*



- To set up a line key to monitor the status of a time condition, use the feature code as the **Value** for a BLF button. Pressing this BLF button will enable a temporary override of the time condition.
  *For example, here's how we would configure a line key as a BLF that monitors our Open/Closed for business status, if our feature code is *274:*

*In our case, with a Digium D70, our line key lights up green whenever our time condition is met (open for business). If we would like it to light up red when we are open instead, we can go back to the Time Conditions module and enable **Invert BLF Hint**.*

Overriding a Time Condition

A temporary override remains in effect until you toggle the time condition again, or until the system reaches the next match in your time group, whichever comes first. A "permanent" override can be thought of as a "sticky" override that stays in place until you manually cancel it. It causes the system to ignore the time group schedule until you cancel the override. Below are instructions on creating or canceling both temporary and permanent overrides.

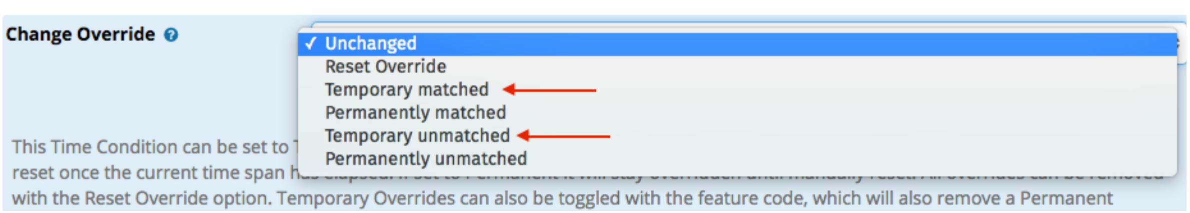
Temporary overrides can be controlled via feature codes/BLFs or the Time Conditions module. Permanent overrides can only be *create d* from the Time Conditions module, but they can be *cancelled* either through feature codes/BLFs or the Time Conditions module.

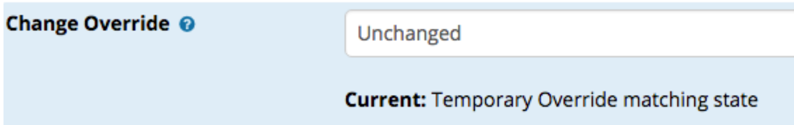
You can temporarily override your time condition in two ways. Either:

A) Dial the feature code for the time condition (or press the BLF button that you've set up for the feature code), **OR...**

B) Visit the Time Condition module and click on the edit button (

) next to the time condition. In the **Change Override** menu, select a temporary override (**temporary matched** or **temporary unmatched**). Click the **Submit** button and then click the **Apply Config** button. Your new temporary matched or unmatched state will be shown next to **Current.**



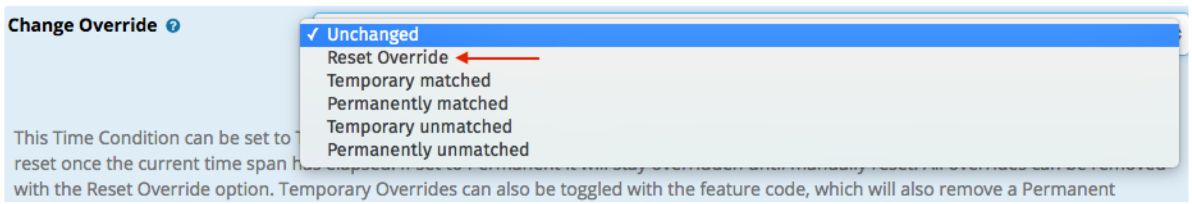


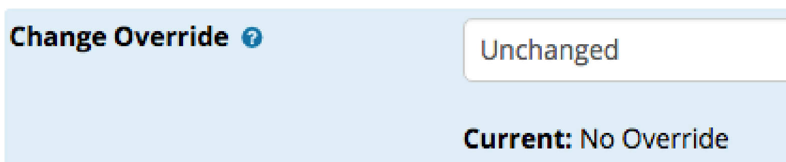
To undo your temporary override, either:

A) Dial the feature code for the time condition (or press the BLF button that you've set up for the feature code), **OR...**

B) Visit the Time Condition module and click on the edit button (

) next to the time condition. In the **Change Override** menu, select **Reset Override**. Click the **Submit** button and then click the **Apply Config** button. "No Override" will be shown next to **Current.**



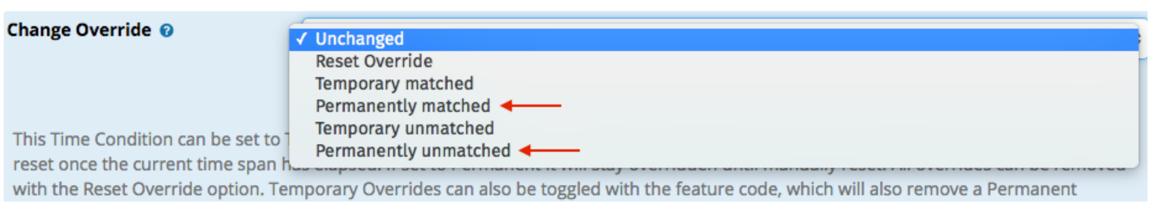


To create a permanent override:

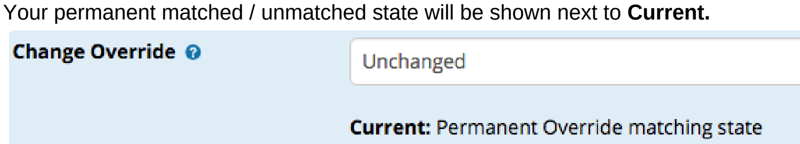
- Visit the Time Condition module and click on the edit button (

) next to the time condition.
- In the **Change Override** menu, select either **Permanently matched** or **Permanently unmatched**.



- Click the **Submit** button and then click the **Apply Config** button.
- Your permanent matched / unmatched state will be shown next to **Current.**

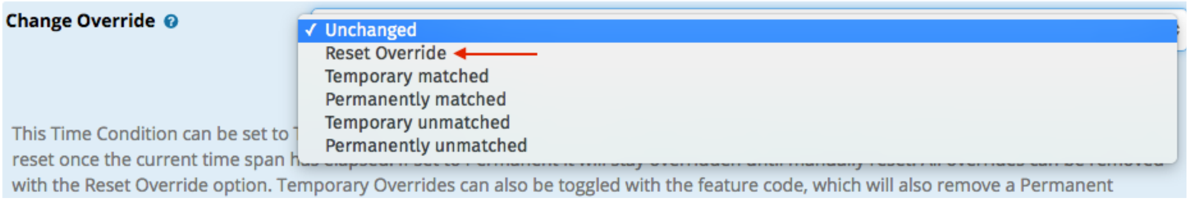


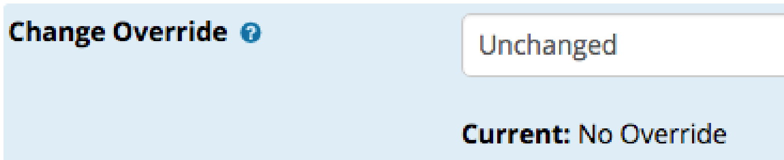
To cancel a permanent override, either:

A) Dial the feature code for the time condition (or press the BLF button that you've set up for the feature code), **OR...**

B) Visit the Time Condition module and click on the edit button (

) next to the time condition. In the **Change Override** menu, select **Reset Override**. Click the **Subm it** button and then click the **Apply Config** button. "No Override" will be shown next to **Current.**



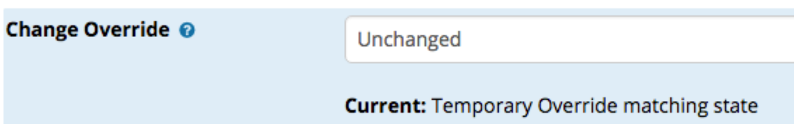


To view the override status in the Time Conditions module:

Visit the Time Condition module and click on the edit button (

) next to the time condition. The current override status will be shown next to **Current.**

In our example, we've temporarily overridden the time condition.



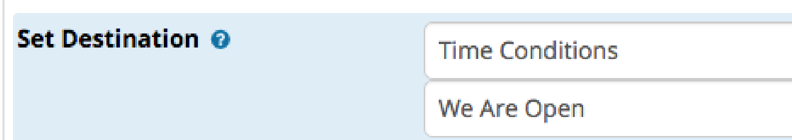
Setting a Time Condition as an Inbound Route Destination

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 **ConnectivityInbound 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.

Please see our Time Conditions Module wiki for more information.

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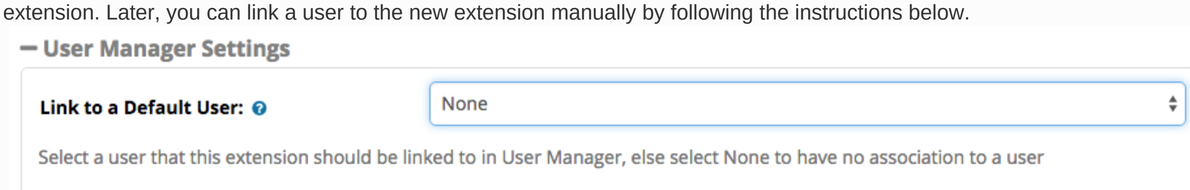
Creating and Managing UCP Users (in User Management)

The User Management module allows you to control users' access to FreePBX Administration, Bria Cloud Solutions, Fax, Rest API, and the User Control Panel (UCP). In this wiki, we will focus only on UCP.

UCP is an end user interface where users can manage the PBX and its features. Users can access it from a desktop computer, tablet, or mobile device.

As an administrator, you can decide what UCP permissions to give to a user. Depending on what you've enabled, UCP users can manage voicemail, make web-based calls, chat via SMS (SIPStation customers only) or XMPP, view call histories (CDR), listen to recorded calls, participate in conference rooms, send and receive faxes, set up follow-me, adjust the button layout of their phones, and more. You can also give a single user permission to view and manage other accounts.

Note
Unless you use the Extension Quick Create function, the Extensions module will automatically a new UCP user for each new extension by default. If you are not using Quick Create, and do not want a new UCP user to be created along with a new extension, select **None** n ext to **Link to a Default User**. This option is found under the **User Manager Settings** section in the **General** tab when creating a new extension. Later, you can link a user to the new extension manually by following the instructions below.



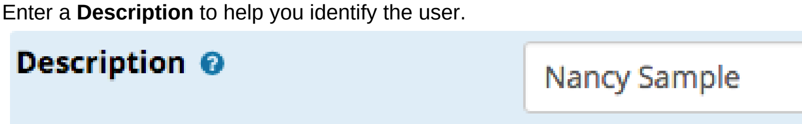
Adding a New User and Setting Up UCP Privileges

These instructions explain how to manually create a new user who has access to the User Control Panel (UCP). If your new user was automatically created when creating a new extension, some of this information will already be generated for you.

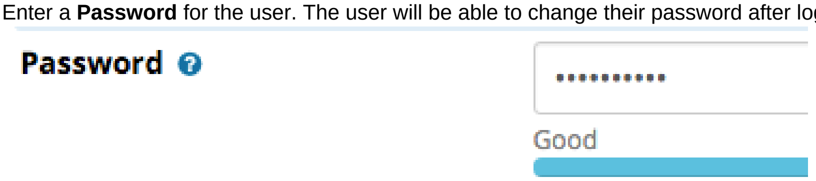
- Go to **Admin User Management**
- Click the **Add** button. You should be viewing the **Login Details** tab.

- Enter a **Login Name** for the user. You can use the extension number as the username, or set it to whatever you wish. This will be the username the user will use to log into UCP.

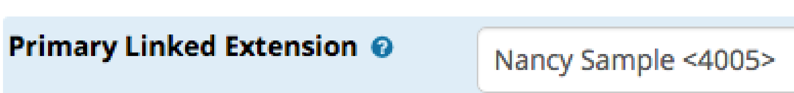
- Enter a **Description** to help you identify the user.



- Enter a **Password** for the user. The user will be able to change their password after logging in.

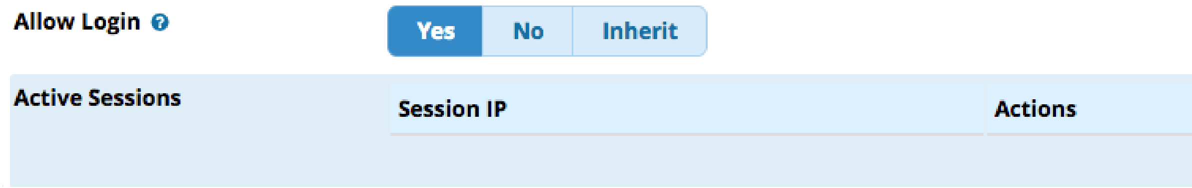


- Select the **Primary Linked Extension.** A single UCP user can only be linked to one extension, and an extension can only be linked to a single UCP user.

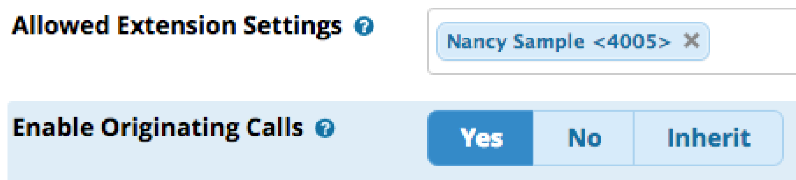


- Click the **User Details** tab. Enter as much information for the user as you would like, such as name, title, department, company, e-mail address, and phone numbers.
- Click the **UCP** tab. You should now be viewing the **General** tab within the UCP section.

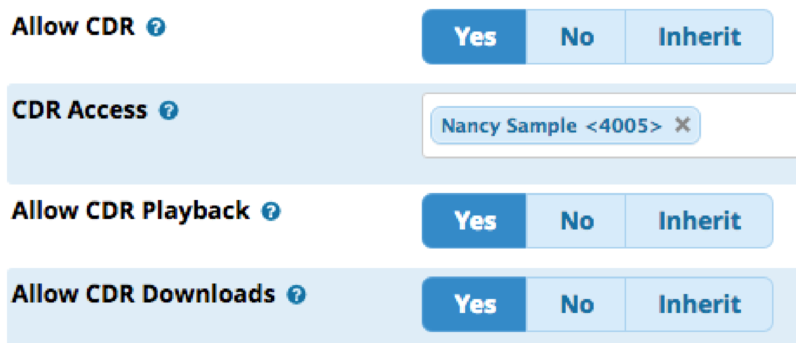
- Set **Allow Login** to **Yes.** Below, you will see any active sessions. If this is a new user, there should not be any sessions shown here.



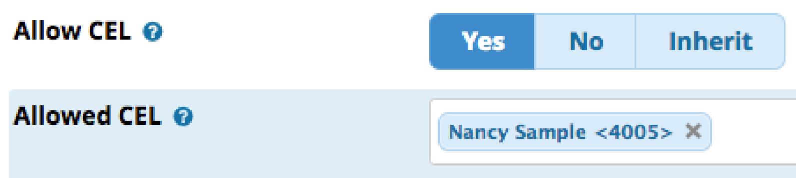
- Click the **Miscellaneous** tab. If you would like this user to be able to change settings for an extension (including their own), select the extension(s) here. Click inside the field next to **Allowed Extension Settings**. A menu will appear showing your existing extensions. Choose one, and repeat the process if you would like to select more than one. If you would like the user to be able to make calls from UCP, set **Enable Originating Calls** to **Yes**.



- Click the **Call History** tab. Here, you can control Call Detail Record (CDR) settings. You can choose whether the user can see their call history, play call recordings, and download CDR reports. You can also grant the user access to another user's call history.

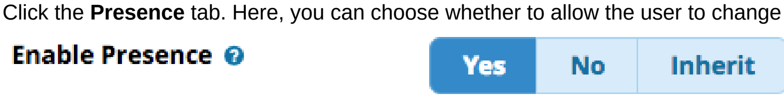


- Click the **Call Event Logging** tab. Here, you can choose whether to enable Call Event Logging (CEL) and select the extension(s).



- Click the **Conferences** tab. Here, you can grant the user access to one or more conference rooms that you have set up. The user will be able to see the conference rooms in UCP.

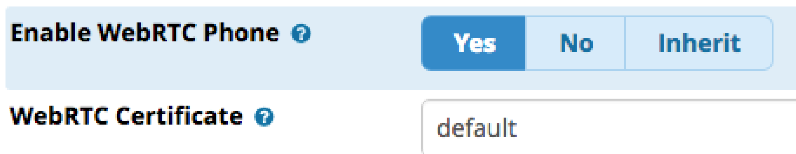
- Click the **Presence** tab. Here, you can choose whether to allow the user to change their presence state via UCP.



- Click the **SIPStation SMS** tab. If you would like a user to be able to send and receive SMS (text) messages via UCP, select one or more **SMS Assigned DIDs** from the list. (Note: for SIPStation customers only.)

- Click the **Voicemail** tab. If desired, you can set **Enable Voicemail Access** to **Yes** in order to allow the UCP user to listen to voicemail messages in UCP. Choose which extension(s) you would like to be **Allowed Voicemail** extensions for the user.

- Click the **WebRTC** tab. Here, you can **Enable Web RTC Phone** and choose a **WebRTC Certificate.**



- If you would like this user to have access to XMPP chat, click the **XMPP** tab located near the top of your screen (above the row of UCP-related tabs). Then set **Enabled** to **Yes.**

- When finished setting up the user, there are two different options for how to submit the information. You can either click the **Sub mit & Send E-mail to User** button or the **Submit** button, depending on whether you would like the user to receive an e-mail with login instructions. After clicking one of the submit options, click the **Apply Config** button.

Please see our User Control Panel (UCP) wiki for a detailed guide on the user interface.

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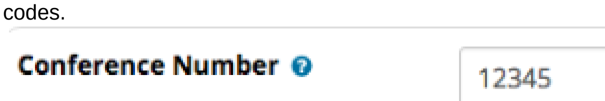
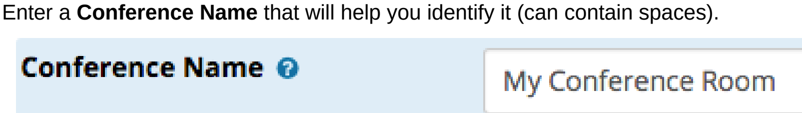
Setting Up Conferences

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.

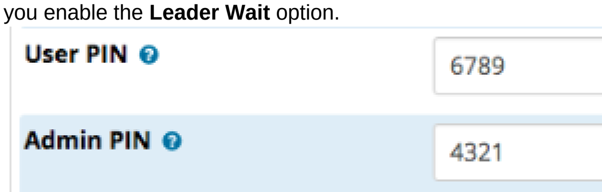
- Go to **Applications Conferences**
- Click the **Add** button.
- Enter a **Conference Number** that callers will dial to reach the conference. This cannot match any existing extensions or feature codes.



- 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.

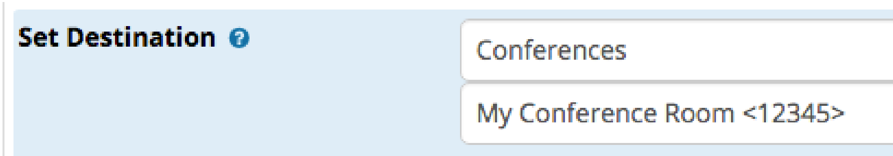
Direct-Dialing Into a Conference

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.



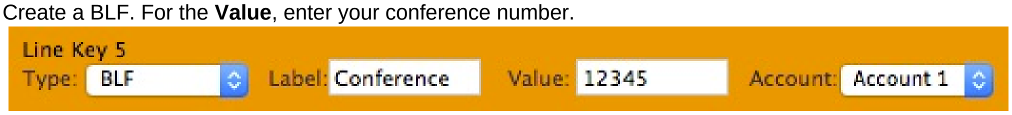
Internal Callers:

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

Setting up a BLF for a Conference

You can set up a Busy Lamp Field (BLF) to monitor the status of a conference room. Pressing the BLF button will allow you to dial into the conference.

- Go to **Settings End Point Manager**
- Click on your phone’s brand.
- Under “Manage,” click on the name of your phone template.
- Click on your phone model under “Available Phones.”
- Create a BLF. For the **Value**, enter your conference number.



- 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, please see our instructions for setting up an IVR earlier in this page.

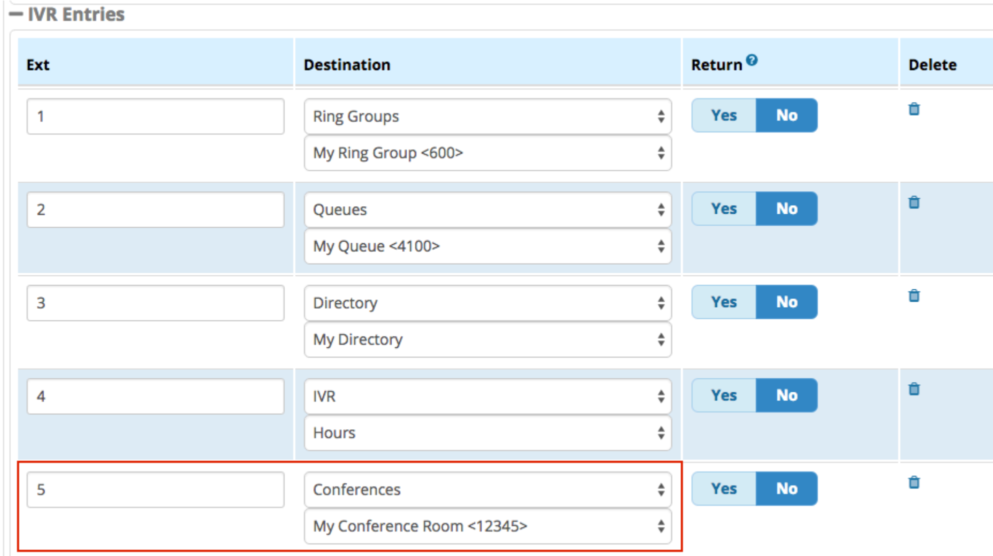
If you have an IVR set up with the the **Enable Direct Dial: Enabled** option set up, outside callers would be able to dial the conference number from within the IVR.

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.

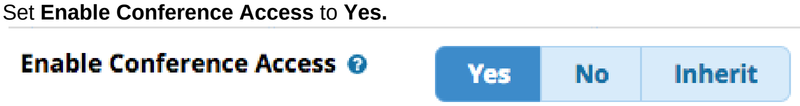
Giving a User Access to Conferencing in UCP

Conferences are not enabled by default in the User Management module. You can visit this module to give a user access to a conference room in UCP.

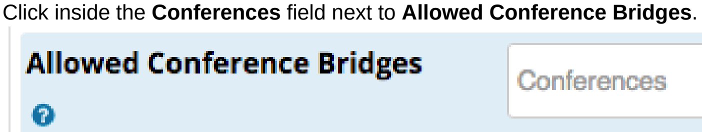
- Go to **Admin User Management**
- Click the edit button (

) next to the user.
- Click the **UCP** tab. (You may need to use the arrow button

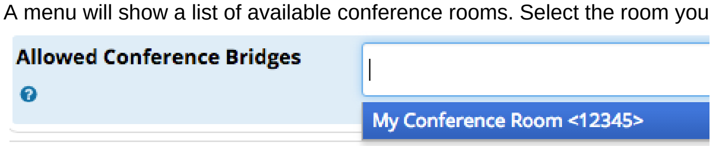
to navigate to this tab.)
- In the bottom row of tabs, click the **Conferences** tab.
- Set **Enable Conference Access** to **Yes.**



- Click inside the **Conferences** field next to **Allowed Conference Bridges**.

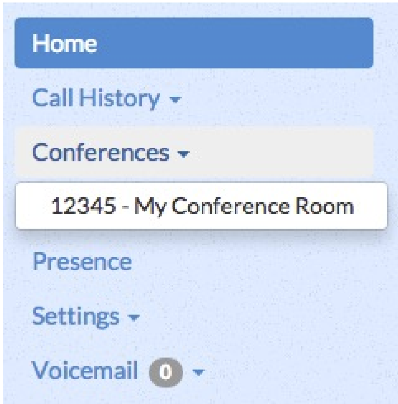


- A menu will show a list of available conference rooms. Select the room you would like to give the user access to.

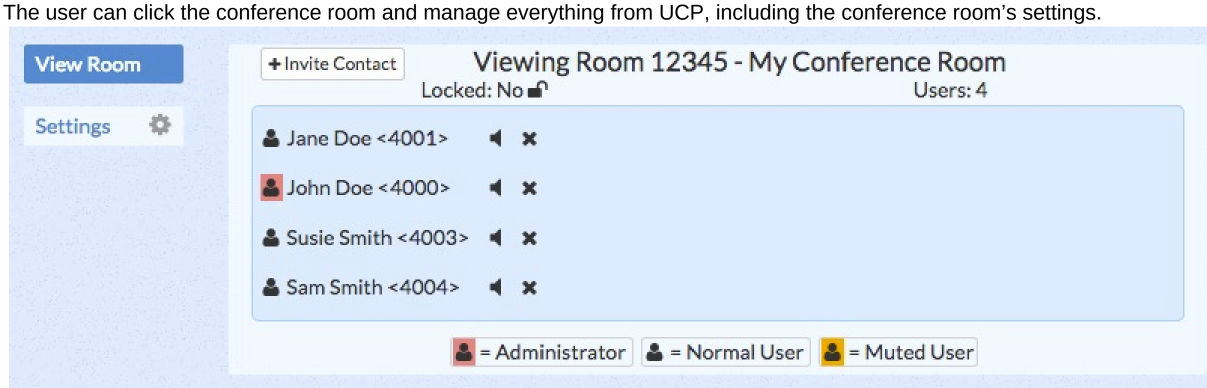


- Your selected conference room will appear in the **Allowed Conference Bridges** field. You can repeat the process to select additional rooms if needed.

- Click the **Submit** button and then click the **Apply Config** button.
- Now, when the user logs into UCP, a “Conferences” option will show up in the menu, and the user will see the available conference(s).



- The user can click the conference room and manage everything from UCP, including the conference room’s settings.



Please see our Conferences Module wiki for more information.

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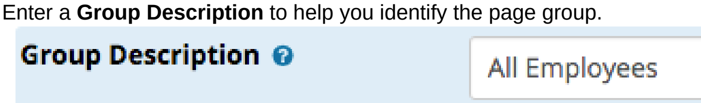
Setting Up Paging

Paging allows you to dial a group of phones that will auto-answer in speakerphone mode. You can use the Paging module to create a list of phones and configure several settings. You can choose what to do when phones being paged are on another call. Settings are available for two-way communications (duplex mode), outbound notifications, recorded messages, and valet-style paging, and scheduled paging. Some of these features require the Paging Pro commercial module.

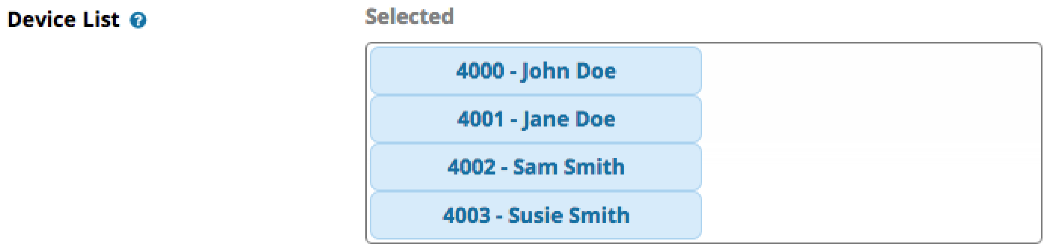
> Only phones that support auto answer SIP signaling will be able to auto-answer. Most soft phones do not support this. Phones that do not support auto-answer will ring instead of auto-answering.

- Go to **Applications Paging and Intercom**
- Click **Add Page Group**.
- Enter a **Paging Extension**. This will be the number users can dial to reach the page group. It cannot match any existing extensions or feature codes.

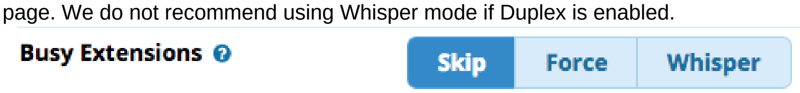
- Enter a **Group Description** to help you identify the page group.



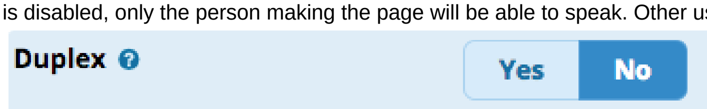
- Drag the desired devices from the **Not Selected** bin to the **Selected** bin.



- Choose how to handle busy phones:
  - **Skip**: Busy extensions will not be paged.
  - **Force:** Busy extensions will be paged even when the phone is in use. This is useful for “emergency” paging.
  - **Whisper:** Busy extensions will receive the page in the earpiece instead of the speakerphone. The remote party will not hear the page. We do not recommend using Whisper mode if Duplex is enabled.



- Choose whether you want to enable duplex mode. This essentially creates an instant conference call where all of the users are able to communicate with each other. Duplex mode does not make much sense if you have “Busy Extensions” set to “Whisper.” If duplex mode is disabled, only the person making the page will be able to speak. Other users would be able to un-mute themselves by pressing *1.



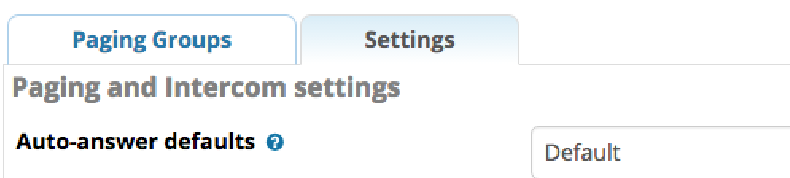
- When finished setting up your page group, click the **Submit** button and then click the **Apply Config** button.

Changing the Auto-Answer Sound

By default, a beep is played when a phone auto-answers a page. You can change this to a pre-recorded announcement or turn off this feature.

(If you need to create a new system recording, go to **Admin System Recordings** to record audio or upload an audio file.)

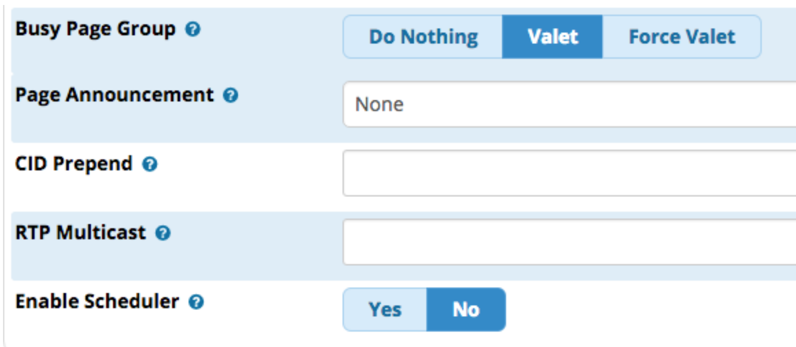
- Go to **Applications Paging and Intercom**
- Click the **Settings** tab.
- Change the **Auto-answer defaults** by choosing a different recording from the drop-down menu. *Default is shown here.*



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

Paging Pro Features

Paging Pro features, if enabled, are shown below the “Add Paging Group” or “Modify Paging Group” options. Paging Pro lets you choose how to page a group that is already in use, play an announcement, prepend text to the Caller ID, enable RTP multicast, and schedule automatic pages. Scheduled pages could be a replacement for school bell systems or lunch break buzzers, for example.



If you modify these settings, don't forget to click the **Submit** button followed by the **Apply Config** button.

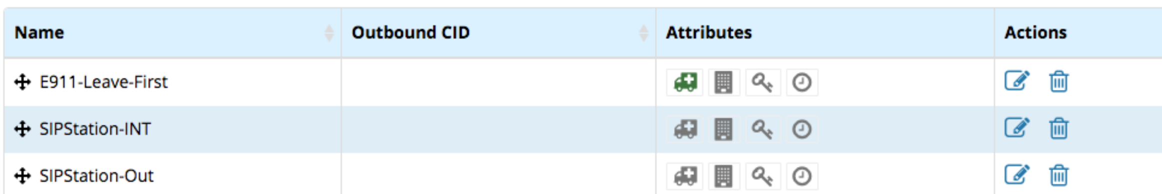
Setting up Outbound Route Notification

Outbound notifications can be set up in order to notify a phone or group of phones when a user dials a specific number. For example, you could set up an outbound notification for 911. If a user dials 911, the linked page group would be notified of the number that was dialed and which user made the call. Everyone in the group would be connected to the call with the ability to listen. Any user in the page group can dial *1 to enable duplex mode (two-way communication), which would allow the user to speak.

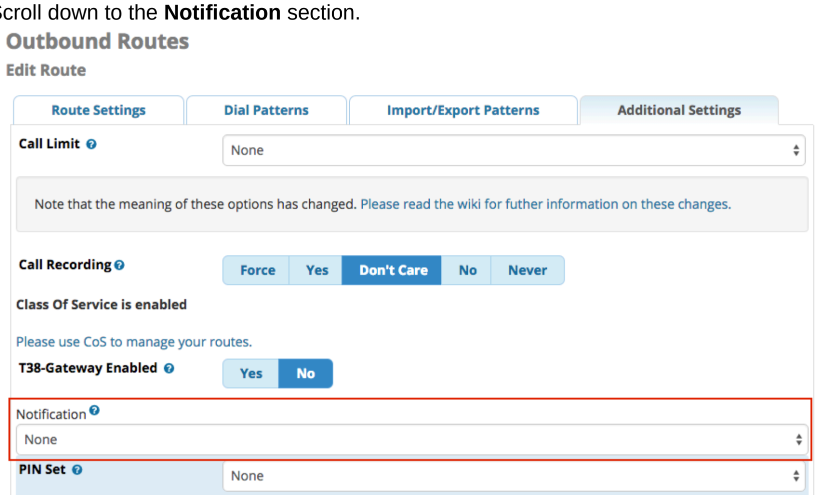
- To set up an outbound notification, first go to **Connectivity Outbound Routes**.
- Click on the edit button (

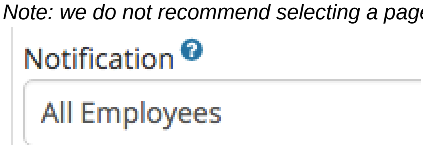
) next to the outbound route that you are creating the notification for.
  In our example we will be creating a notification for our E911 route.



- Click on the **Additional Settings** tab.
- Scroll down to the **Notification** section.



- Select the desired page group from the drop-down menu.
  Note: we do not recommend selecting a page group that uses Force Valet Paging.



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

> If you've chosen to set this up for your emergency route, obviously you wouldn't want to dial 911 in order to test your outbound route notification… Instead, you can dial 933 (the e911 address verification service) to test the functionality of the notification. Be sure 933 is included as a dial pattern in your emergency route.

Please see our Paging & Intercom Module wiki for more information.

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Setting Up Parking

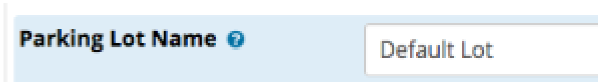
The Parking module creates and configures parking lots, sometimes referred to as parking orbits, to which calls can be transferred in order to allow another extension to retrieve the calls. The standard Parking module allows one parking lot to be available to all phones on the system. The Parking Pro module allows multiple parking lots, plus the ability to announce parked calls when combined with the Paging Pro module.

- Go to **Applications Parking**
- We will focus on editing the default lot for now. Later, you can create new lots if desired (if using Parking Pro). Click on the edit button ( ) next to the default lot.

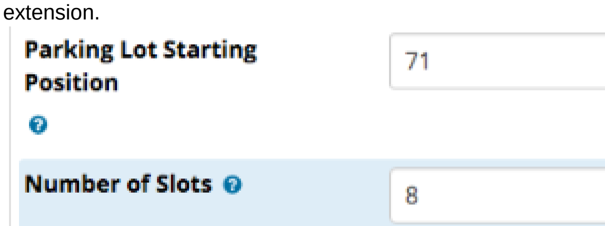
Extension	Slots	Name	Actions
70	71-79	Default Lot	

- The first item displayed is the **Parking Lot Extension**. This is the extension to which you will transfer calls in order to park them in the first available slot. Take note of this extension. The default is 70 for the first lot. You can change it, but we recommend that you leave it at its default.

- The **Parking Lot Name** is "Default Lot," but you can change it if you would like.

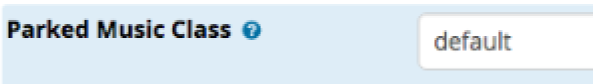


- You can decide how many parking "slots" you would like to set up for your lot. When a call is parked, it is placed into the first available slot. If another call comes in and is parked while the first call is still parked, the second call is placed into the next available slot, and so on, until all slots are filled. If all slots are filled, a parked call will automatically go to the "alternate destination" that we will talk about below. Each slot has a number. The **Parking Lot Starting Position** will be the first slot number in your sequence. You can define the **Nu mber of Slots** to make available. The system will number them in sequence. For example, if your starting position is 71, and you want 8 slots, the slots will be numbered 71 through 78. To avoid confusion, the starting position number should be different from the parking lot extension. You can park a call directly into a specific slot by transferring the call to the slot number instead of the general parking lot extension.

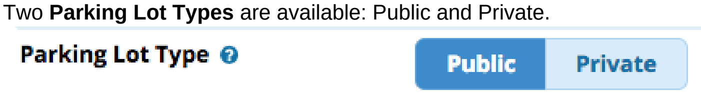


- Set the **Parking Timeout** in seconds. This is the amount of time a call will stay in a parking lot until it times out. The default is 45 seconds. Later, you'll set the timeout behavior and destination at the bottom of the page.

- You can change the **Parked Music Class** if desired. This is the music played to parked callers. Note that this setting will be ignored if you set a different music class at a point earlier in the call flow, such as within a queue or ring group.

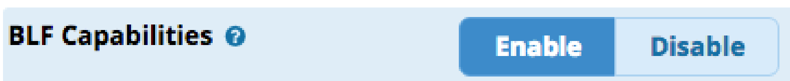


- Two **Parking Lot Types** are available: Public and Private.

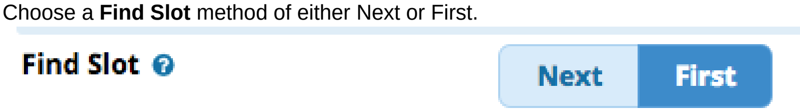


  - **Public:** Anyone can transfer calls to a public lot.
  - **Private:** Only certain extensions can transfer calls to a private lot. To give an extension access to the lot, go to **Applications Extensions,** choose an extension to edit, and select a lot.

- Choose whether to enable **BLF Capabilities** for the parking lot.

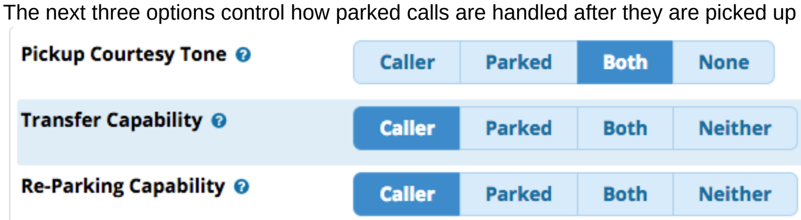


- Choose a **Find Slot** method of either Next or First.



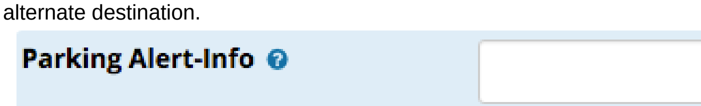
  - **Next:** Park calls in the next sequential slot relative to the last parked call.
  - **First:** Park calls in the first available slot.

- The next three options control how parked calls are handled after they are picked up or not picked up.

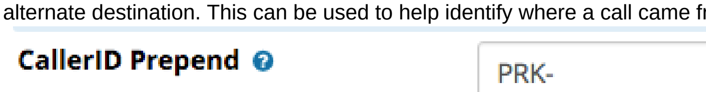


  - **Pickup Courtesy Tone:** Who (if anyone) will hear a beep when a parked call is picked up.
  - **Transfer Capability:** Who (if anyone) can transfer a parked call that has been picked up.
  - **Re-Parking Capability:** Who (if anyone) can re-park a parked call that has been picked up.

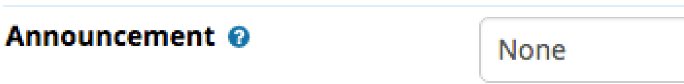
- You can optionally set **Parking Alert-Info** that will be added to a timed-out parked call prior to sending it back to the originator or alternate destination.



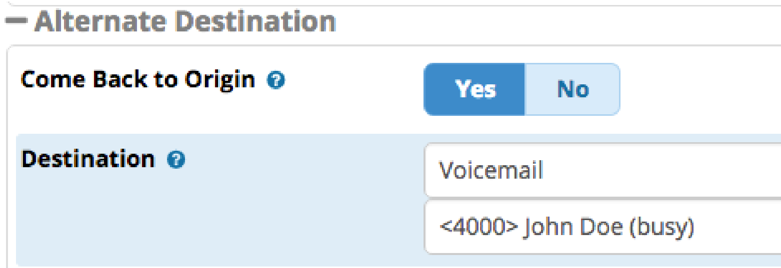
- You can optionally set a **CallerID Prepend** that will be added to the current caller ID prior to sending the call back to the originator or alternate destination. This can be used to help identify where a call came from.



- Choose an **Announcement** to play to parked callers before they are returned to the originator or sent to the alternate destination, or select "None."



- At the bottom, set whether you would like a timed-out parked call to **Come Back to Origin.** If set to **Yes,** the call would be sent back to the device that parked it. If that device is busy, the call would then be sent to the alternate destination. If set to **No**, a timed-out parked call would be sent directly to the alternate destination. Select a **Destination** from the drop-down menus.

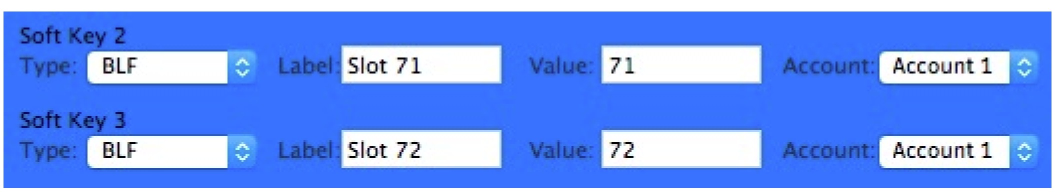


- Click the **Submit** button and then click the **Apply Config** button.

Setting Up a BLF for a Parking Slot

You can set up Busy Lamp Field (BLF) keys to view and retrieve parked calls. This makes it easy for phone users to see when calls are parked, and to pick them up from their extensions by pressing a button.

- Go to **Settings EndPoint Manager**
- Select your phone brand.
- Select the template you would like to edit.
- Select the phone model.
- Set up a BLF key by choosing **Type: BLF** and using the slot number as the **Value**.
  In our example, we've set up soft keys on our phones to monitor two of our parking lot slots, 71 and 72. We've labeled them "Slot 71" and "Slot 72" but you can use whatever labels you wish. Note the slot numbers themselves are used as the Values.



- Click **Save Model.** You should be taken back to the main page for the phone template.
- Click **and Rebuild Configs** at the bottom of your template page, and then reboot your phone.

Please see our Parking Module wiki for more information.

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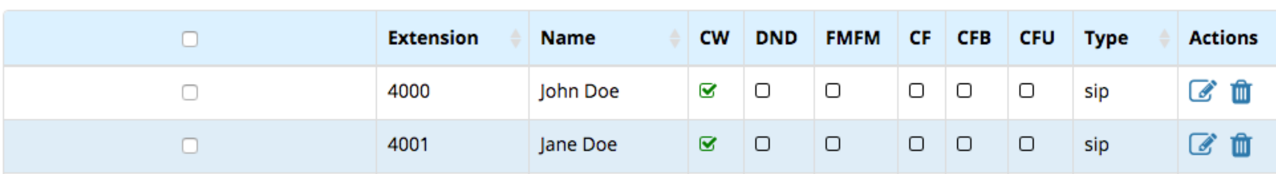
Setting Up Follow Me

The Follow Me module allows you to redirect an inbound call to another location. This is designed to help an inbound caller reach you when you are not available at your main phone. For example, you could set Follow Me to ring your extension for a certain period of time, then to ring another extension and/or an outside number, such as your mobile phone.

Users may dial the feature code of ***21** to toggle Follow Me (enable/disable). You can also set up a BLF button for this purpose. A user can also access this feature in the User Control Panel (UCP).

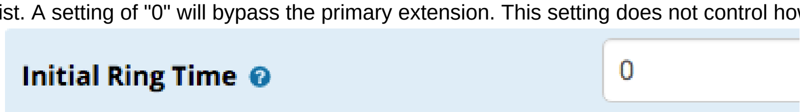
- Go to **Applications Follow Me**
- At a glance, you can see if Follow Me is enabled for an extension by looking for a green check mark in the **FMFM** column. (FMFM stands for "Find Me Follow Me.") The boxes only display the status; you cannot click them to change the status. Click the edit button (
  
) for the extension you wish to edit.



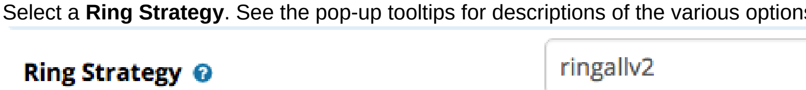
	Extension	Name	CW	DND	FMFM	CF	CFB	CFU	Type	Actions
☐	4000	John Doe	☑	☐	☐	☐	☐	☐	sip	
☐	4001	Jane Doe	☑	☐	☐	☐	☐	☐	sip	

- Click the **Find Me/Follow Me** tab.
- At the top, you will see Yes/No buttons showing you whether Find Me/Follow Me is enabled. Click **Yes** to enable Follow Me. This will allow you to edit the settings for Follow Me. Regardless of whether you want Follow Me to be active or not, you will need to temporarily enable it in order to edit and submit the settings. Later, after you apply config, you can come back and disable it if desired.

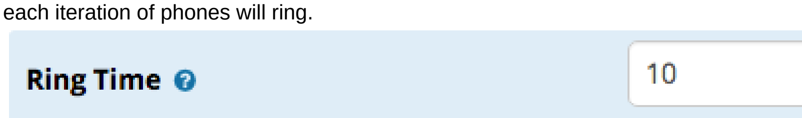
- Set your **Initial Ring Time**. This is how long (in seconds) the primary extension will ring before the system proceeds to call the follow-me list. A setting of "0" will bypass the primary extension. This setting does not control how long your follow-me list phones will ring.



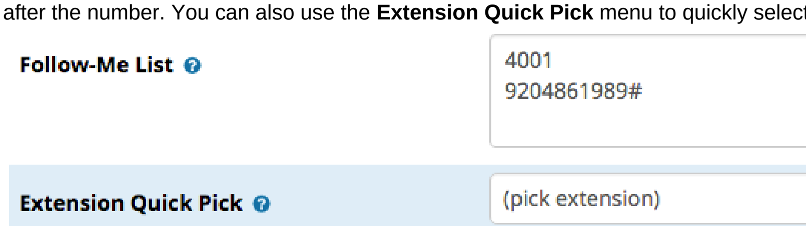
- Select a **Ring Strategy**. See the pop-up tooltips for descriptions of the various options.



- Set the **Ring Time** in seconds. This is the time that your follow-me list phones will ring. For hunt-style ring strategies, this is the time that each iteration of phones will ring.



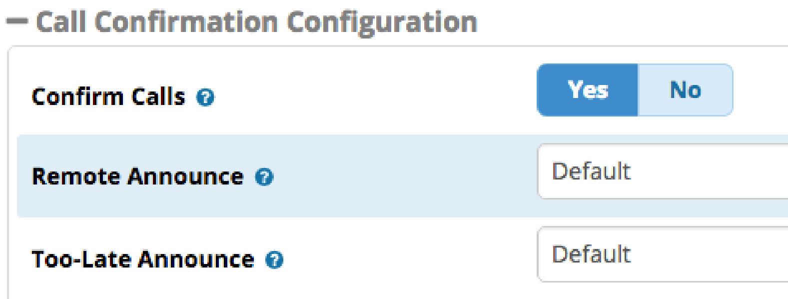
- Add at least one number to your **Follow-Me List**, one per line. (Do not include the primary extension. If you wish to ring the primary extension, set an **Initial Ring Time** above). You can add extensions and/or external numbers. If you add an external number, place a # after the number. You can also use the **Extension Quick Pick** menu to quickly select existing extensions.



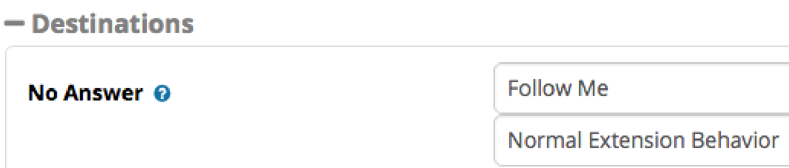
- Choose an **Announcement** to play to callers, if desired. Select whether callers will hear ringing, music, or nothing for the **Play Music On Hold** option.

- Optionally, set a **CID Name Prefix** to prepend to the caller ID. This can help you identify when a call is coming from Follow Me. You can also set **Alert Info,** which can create distinctive rings on certain phones.

- The Confirm Calls feature is designed to prevent a caller from reaching an external voicemail box in case no one answers the call. For example, you might want to prevent an unanswered work-related call from reaching a mobile phone's voicemail. If you set **Confirm Calls** to **Yes**, the user would be required to press "1" to accept an incoming follow-me call on an external phone. **Remote Announce** is the message played to the person answering the call. **Too-Late Announce** is the message played if someone answers the call after someone else has already pressed "1" to accept the call.



- Set the **No Answer** destination. This defaults to Follow Me: Normal Extension Behavior.



- Important: Do not go back to the top of the page and disable Follow Me at this point. You will need to submit your information and apply config first, or you will lose the settings you've entered. You can disable Follow Me *after* you submit and apply your settings, if desired.
- Click the **Submit** button and then click the **Apply Config** button. The settings will be applied. If you go back to your extension list, you will see the box under **FMFM** checked with a green check mark.

> **Note**
> Follow Me is now enabled for the extension, and calls will immediately begin flowing to the extension's follow-me list unless you disable Follow Me. You can stop here if you want Follow Me to remain active.

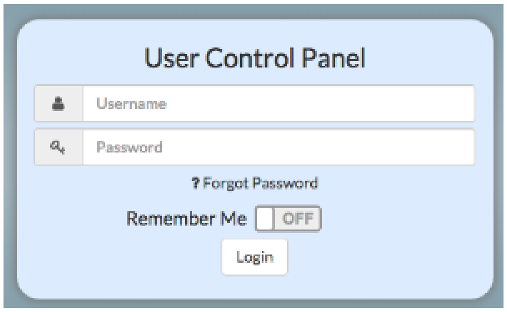
- If you want to turn off (disable) Follow Me at this time, go back to the **Find Me/Follow Me** tab for the extension. Next to **Enabled**, click **No**. Click the **Submit** button and then click the **Apply Config** button. The user's follow-me list and settings will be saved, but calls will now flow only to the primary extension. The user can enable Follow Me by dialing *21, using a BLF set up for that purpose, or turning it on in User Control Panel (UCP) as described below.

Configuring Follow Me in UCP

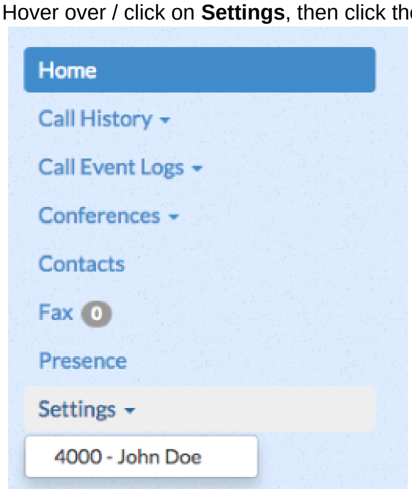
Users can adjust their Follow Me settings directly from the User Control Panel (UCP) if they have access to UCP and are allowed to manage the extension settings within UCP. For more information on controlling a user's UCP permissions, please see the section **Adding a New User and Setting Up UCP Privileges** section earlier in this wiki.

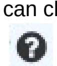
If a user has access to UCP and is allowed to manage extension settings, they can manage Follow Me by following these steps:

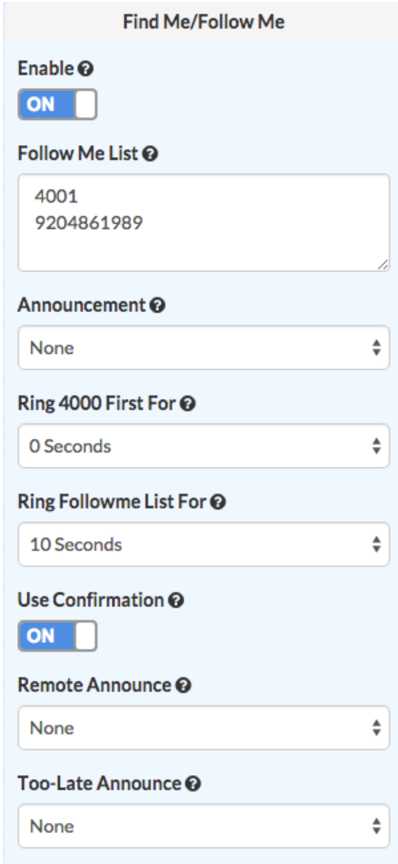
- Log into UCP.



- Hover over / click on **Settings**, then click the extension.

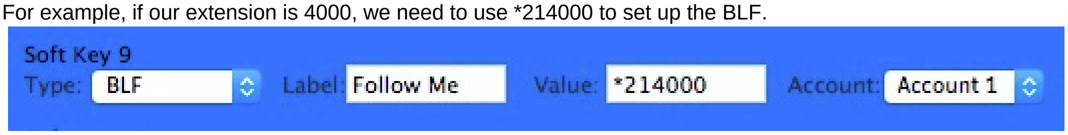


- The current Follow Me status is shown (ON / OFF). If you toggle the ON/OFF switch, settings will be applied immediately. You can edit the follow-me list, announcement, initial ring time, and follow-me list ring time. You can also choose whether to use call confirmation, and choose what announcements to play to the person answering the call. These options are a "lighter" version of what is available in the FreePBX Administration GUI. Some settings, such as ring strategy, cannot be adjusted in UCP. Users can click on the pop-up tooltips (

) for more information. Changes take effect immediately.



Setting Up a BLF for Follow Me

- Go to **Settings EndPoint Manager**
- Select your phone brand from the list at the upper right.
- Click on your phone's template under "Manage."
- Click on the button for your phone.
- The default feature code to toggle Follow Me is *21. Use *21 followed by your extension number as the "value" for the BLF key. For example, if our extension is 4000, we need to use *214000 to set up the BLF.



Please see our Follow Me Module wiki for more information.

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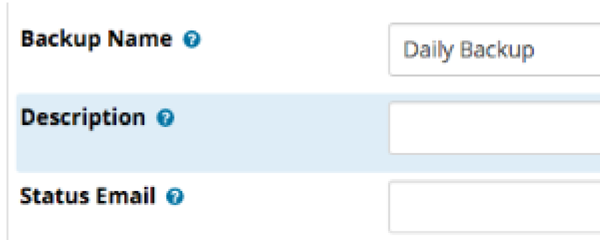
Setting Up Backups

The Backup and Restore module automatically includes a default backup that will run monthly. You can set up custom backups on a schedule of your choice.

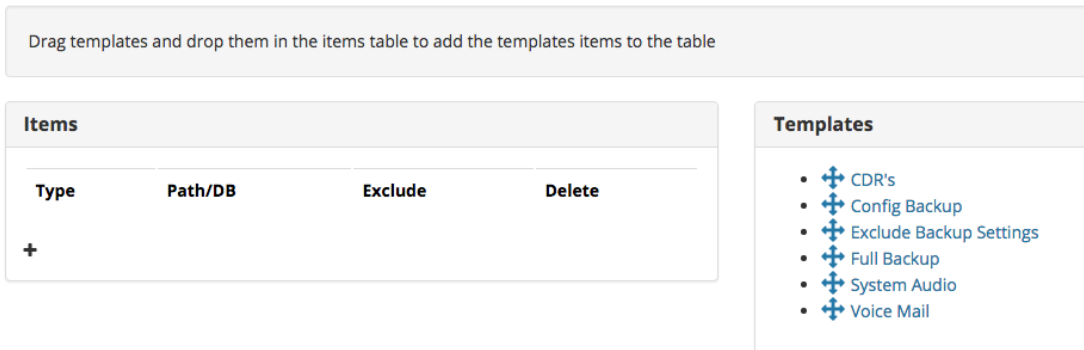
Creating a Backup

- Go to **Admin Backup and Restore.**
- Click **New Backup.**
- Give the backup a unique name. Optionally, give the backup a description and enter an e-mail address where you'd like to be

notified when backups are completed.

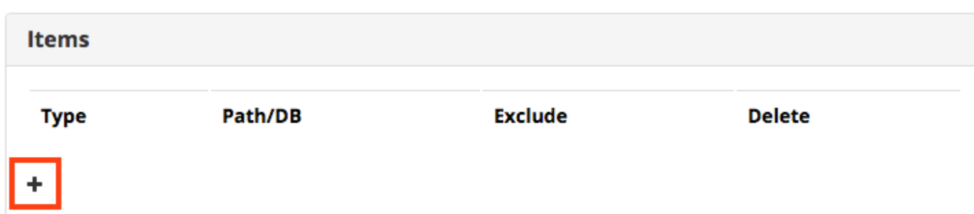


- **There are two ways to add items to your backup list:**
    - **A) Drag and Drop:** Drag the desired **Templates** to the **Items** section. You will see a “drop here” box. After you drop the template, items will appear in your list.

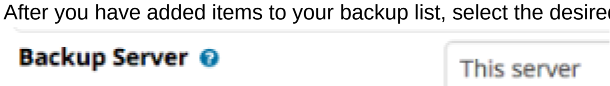


In this example, we have chosen the "Full Backup," "System Audio," and "Voice Mail" templates.

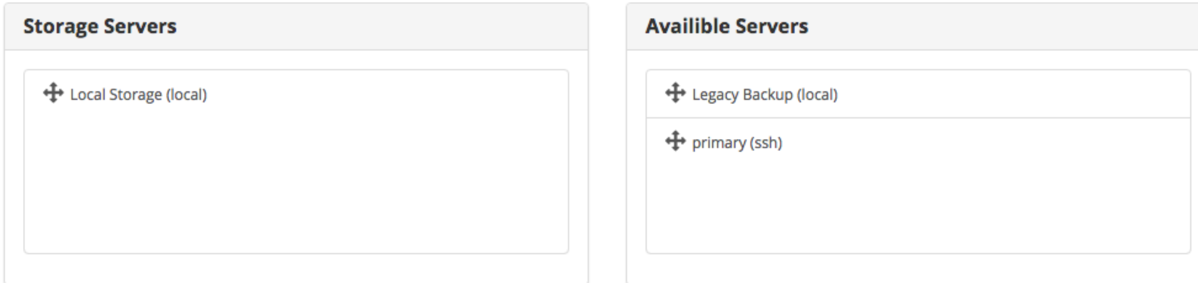
- **B) Plus Sign / Drop-Down Menu:** Click the plus sign (
  ✚
) under **Items** and make a selection from the drop-down menu. You will then be able to manually enter a Path/DB to back up, and add patterns to exclude.
  In this example, we haven't entered any items yet.



- After you have added items to your backup list, select the desired backup server.

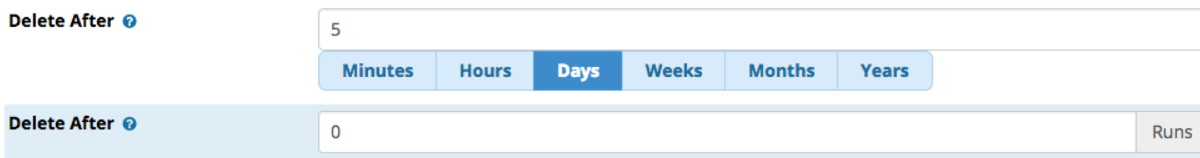


- Choose your storage server(s) by dragging the available server(s) to the Storage Servers column.



- Choose an automatic backup schedule, if desired. If set to “Never,” you will need to manually run the backup.

- Choose how long you’d like to save the backups, if at all. A setting of “0” disables deletion.

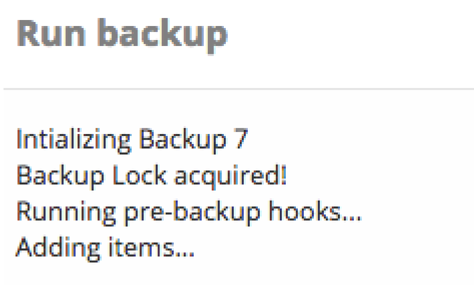


- When finished, click **Save**. You can then click “and Run” to run the backup immediately.

Running a Backup Manually

- Go to **Admin Backup and Restore.**
- If you are not already viewing your list of backups, click the **Backups** menu item.
- Click the run button (blue button with triangle)

- A window will pop up and display the progress of the backup. When done, you can click the **X** button to close it.



See our Backup and Restore wiki for more information.

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Other Modules

There are many other modules, and most are self-explanatory. You can find out more about each of them here: Standard Modules.

Consider the Paid Modules

The FreePBX Distro includes all of the modules you need to set-up a first class PBX. There are, however, some additional modules available that you may wish to purchase.   You can find out more about the Paid Modules here: http://www.freepbx.org/commercial-modules