XO Hosted PBX

MySite Portal User Guide

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About this Document

Welcome to XO Hosted PBX! With XO Hosted PBX, you get an advanced communications system that uses the latest cloud technology to deliver you crisp, high-quality voice over the nationwide XO MPLS network.

This document guides you through the key features of the **MySite web portal**. It is divided into the high-level areas, such as User Management, and Auto-Attendants and Hunt Groups. Within each area are subsections that explain how to use the features in more detail.

What is MySite?

MySite is an easy-to-use administration web interface for configuring service at your site and customizing its settings to meet your users' needs.

Don't confuse the MySite portal with MyPhone. With MyPhone, you can configure settings for your own line. By contrast, MySite lets you configure services for all users within your site. For more information on MyPhone, refer to the *MyPhone User Guide*.

Accessing MySite

Connecting to MySite is easy. You need a recent web browser with Flash support and an Internet connection. Point your browser to

https://portal.xo.bizcommservices.com/

and log in with your username and password. Your login credentials were emailed to you when your XO implementation engineer configured your service.

Tip: Don't know your username or password? Ask your XO contact to review them. They can look up your username and reset your password with just a few clicks.

The MySite User Interface

To administer a site, you must first select the *My Site* tab on the top right of the screen. Figure 1 shows the element. Click on the words *My Site*.

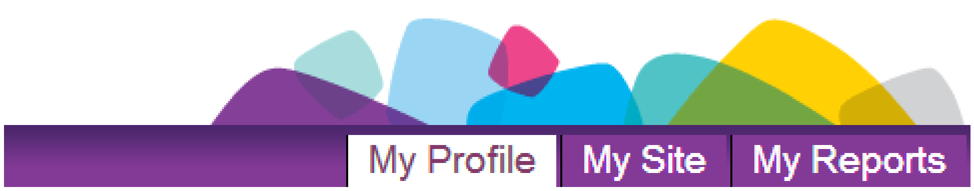


Figure 1: Select the My Site tab to administer a site

Next, you must select the individual site to administer. To do so, type the first few letters of the site's name into the entry box and select the site from the drop down list, as illustrated in Figure 2. Alternatively, double-click on the box to list all the sites and select the one that you want.

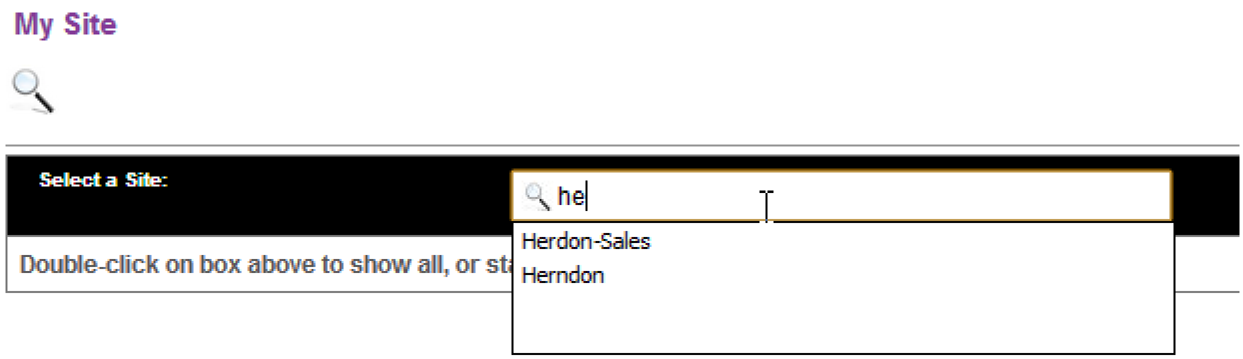


Figure 2: Choosing a site to administer

Having selected a site, you will now see its Home screen. This outlines the things that you can do in the Portal and provides a simple starting point. Immediately above the message you'll see the interface bar, like so:

Each one of the six items on the bar is clickable.

User interface elements

The MySite user interface is quick and simple to use. Here are some of the key elements that you will see as you navigate the screen.

- *Radio buttons*: you may choose one, and only one, of these options.
- *Check boxes*: you may choose, zero, one or more of these options.
- *Help button*: a button identified by a "?". Click this to see a help message.
- *Magnifying glass button*: a button labeled with a looking-glass icon. Click on this to perform a search or filter.
- *Save* or *Apply button*: a button labeled with the word Save (Apply). You must click on this to save any changes you make.

Office management: opening schedule and holidays

Defining your site's opening schedule

Your site's opening schedule describes the days and times that your site is open for business, and, by extension, the times when it is closed. The Hosted PBX system uses this information to decide how and when to route certain types of calls. For example, many enterprises have an Auto-Attendant that answers calls and provides the caller a menu of options. With an opening schedule defined, they can offer one menu for business-hours calls and one for after-hours.

To define the schedule, select the *Site Services* tab along the top of the interface and then the *Schedule* element from the list on the left.

- Select the schedule items to be edited by clicking on it, or add a new item by clicking on the *Add Schedule* button.
- Enter the schedule detail for that day, defining the period when your site will be open for business. Schedule times have a granularity of 30 minutes, e.g. you can choose values such as "09:00am" or "09:30am" but not "09:17am".

- Select *Save*.

Figure 3 illustrates the interface.

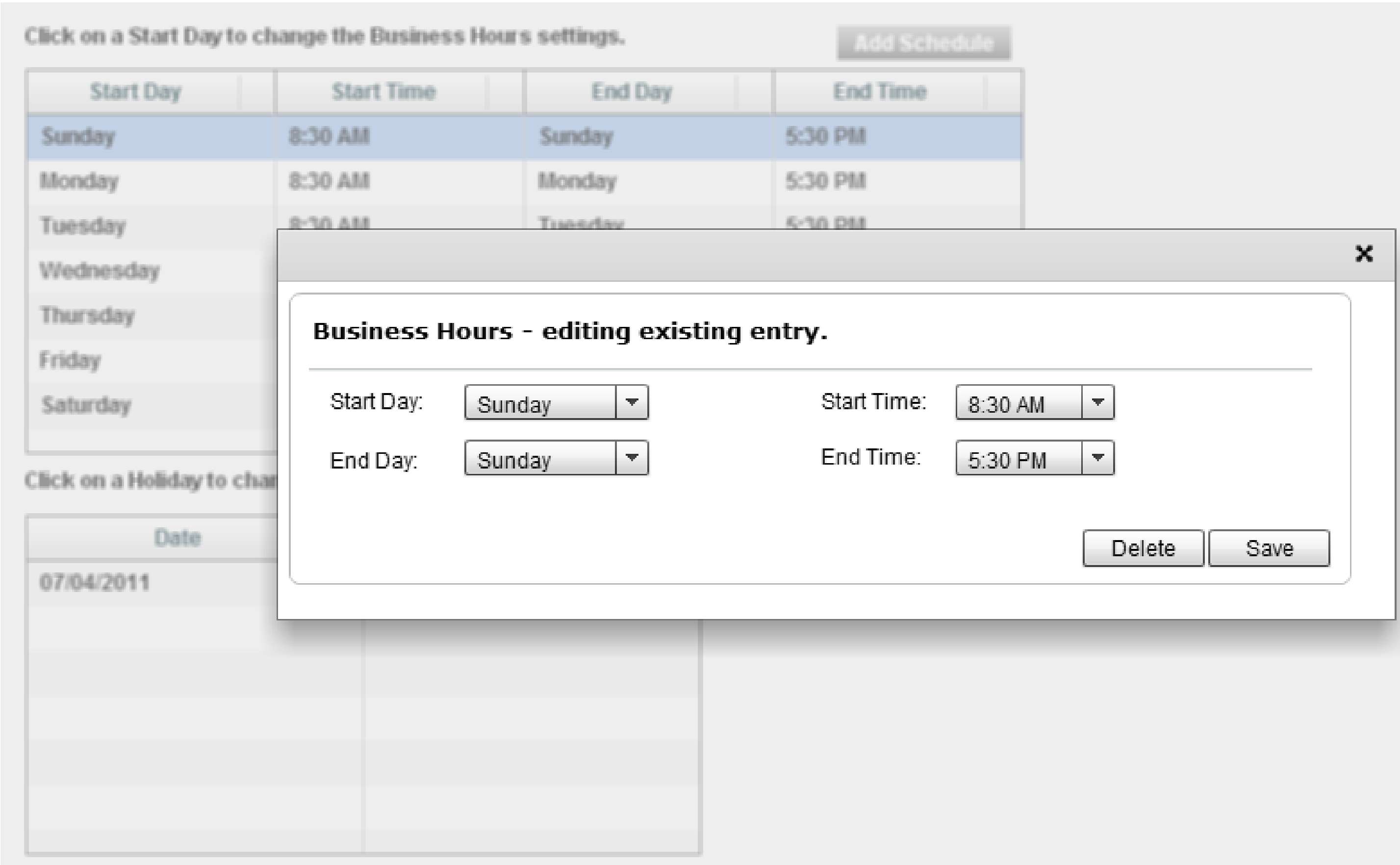


Figure 3: Example of editing the open periods for a day on the schedule

Holidays

Holidays are days when your site is closed over and above those that are defined in the schedule. For example, your schedule may define that your site is open Monday through Saturday, and therefore that it is closed on Sundays, but you may also want to indicate that your site is closed on certain public holidays.

To add a holiday, navigate to the *Schedule* interface on the *Site Services* tab, and click on the *Add a Holiday* button. Fill in the form and click *Save*. Figure 4 provides an illustration.

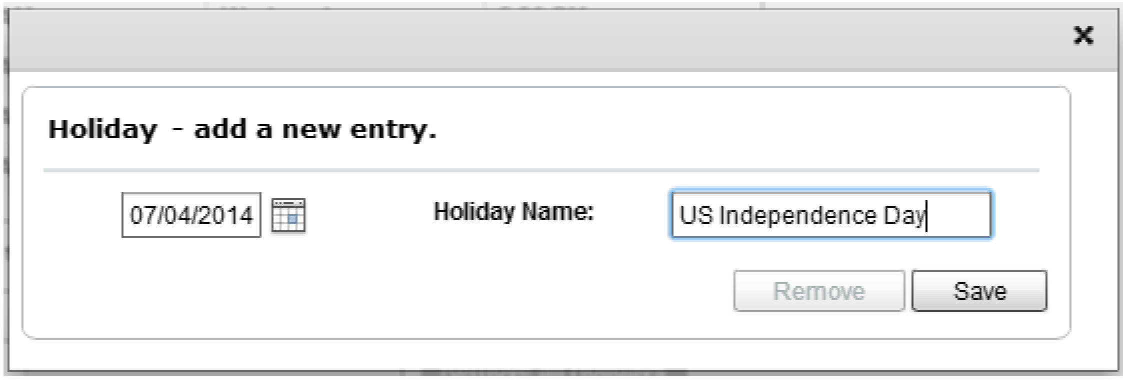


Figure 4: Adding a holiday

User Management: PINs, Passwords, Devices and Seats

Resetting a user's PIN or password

Your users will use their PIN code to access their voicemail and features such as XO Anywhere, and their password to log on to the MyPhone user portal. They can change both using the MyPhone portal at any time. But if they forget either, you can reset them using the MySite portal. (Users can also reset their password themselves at the MyPhone login screen.)

From the Home screen, click on *Phone Assignment*, and double-click on the user whose PIN/password you wish to reset. A new window will appear. Click on the words *User Info*. Figure 5 shows the interface.

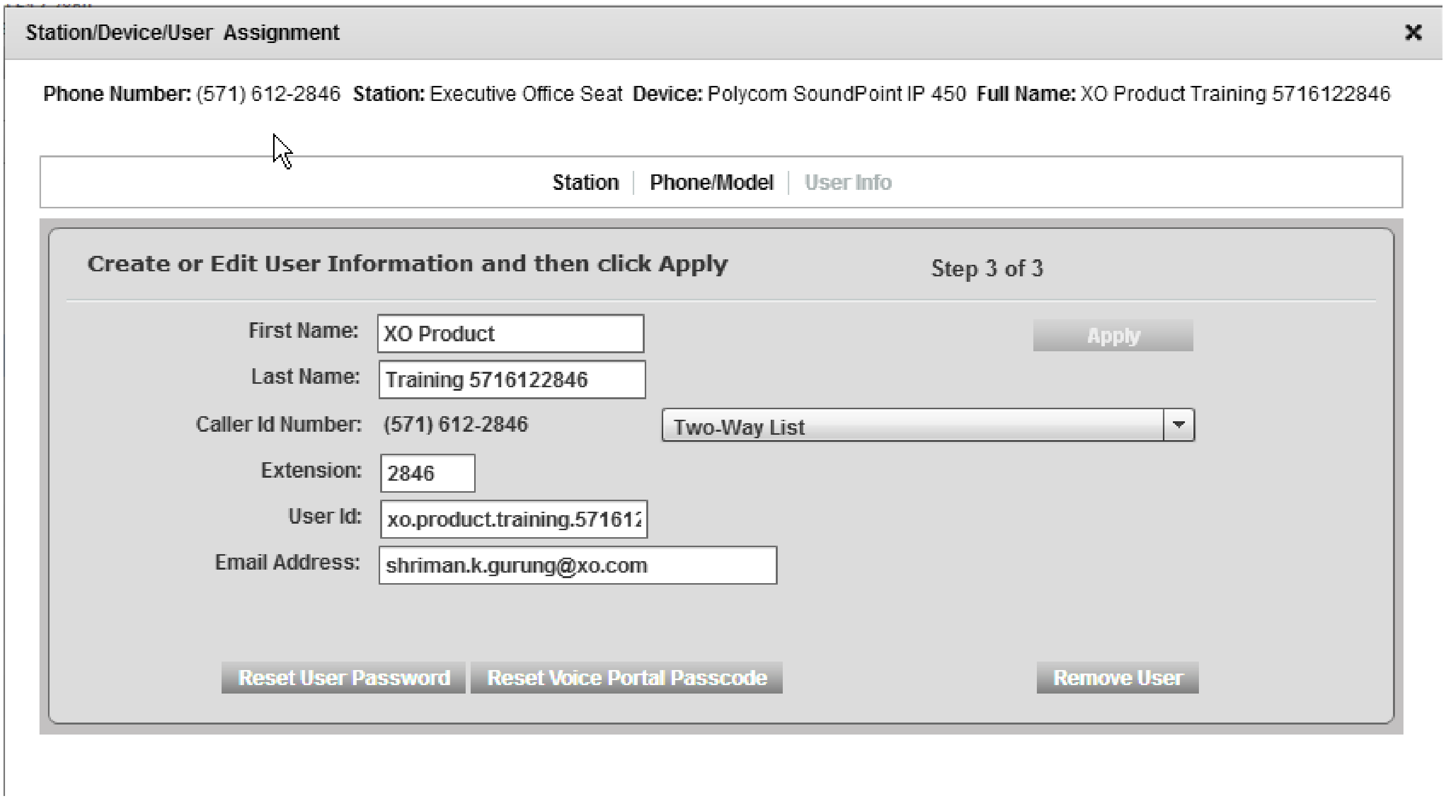


Figure 5: User Info interface screen

- Use the *Reset User Password* button to reset the user's portal password.
- Use the *Reset Voice Portal Passcode* to reset the user's PIN.

In both cases, you do not see the new credential. Instead, the user will receive an email at the email address displayed with an auto-generated credential. They should use this value to login and reset to a value of their choosing.

Tip: It is important that users pick strong passwords and PINs. For example, PINs should not contain "easy to guess" sequences such as repeated digits or fragments of the user's phone number. Another best practice is to use PINs longer than the default length of 4 digits. For example, a six- or eight-digit is considerably harder for an attacker to guess than a four-digit one.

Your enterprise may also have security policies that should be followed when selecting passwords and PINs. Consult with your security administrator if you are unsure.

Managing Devices

Every user at your site is associated with a phone number and a device. This might be a physical phone, or an adapter (ATA) that is used with an analog device such as a fax machine. The Device Management interface in MySite shows you all the end-user phones and devices that participate in the Hosted PBX deployment at your site.

To use the Device Management interface, log in to MySite and select the *Device Management* tab. The interface looks similar to Figure 6.

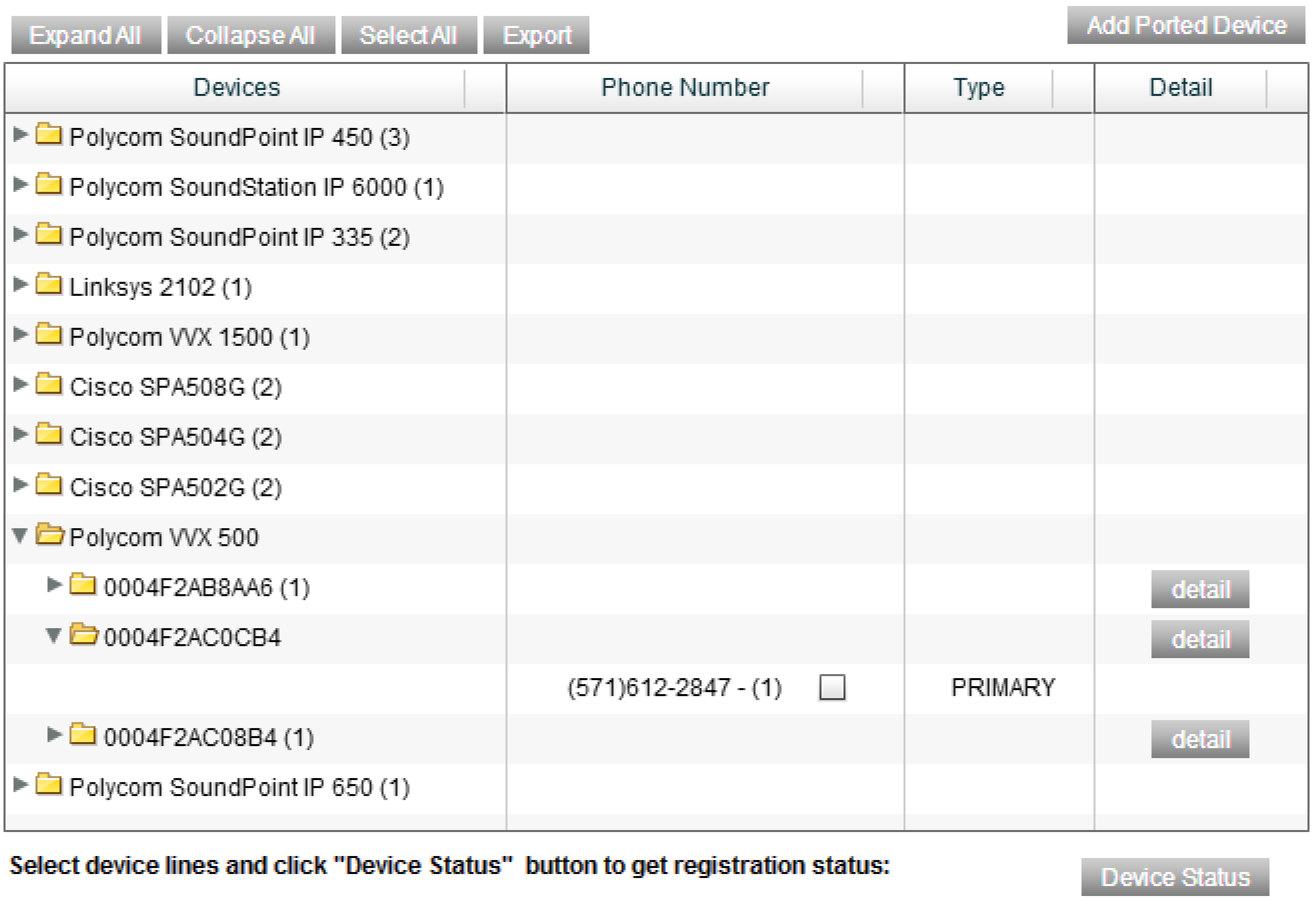


Figure 6: Device Management user interface

In the interface panel, devices are categorized by their manufacturer and model, and then by their MAC address, which is a unique twelve-character identifier. (The MAC address of a device is usually printed on a sticker on the unit.).

To add a device, click the *Add Ported Device* button, enter the MAC address and select the model from the drop down list. Then, click *Save*.

Viewing device status

Selecting a device and clicking the *detail* button will show the lines active on that device. For more information on the device's current status, return to the main *Device Management* panel, locate the device of interest and check the box next to the number in the *Phone Number* column. Then, click the *Device Status* button. After a few seconds, you will see a window like the following (Figure 7).

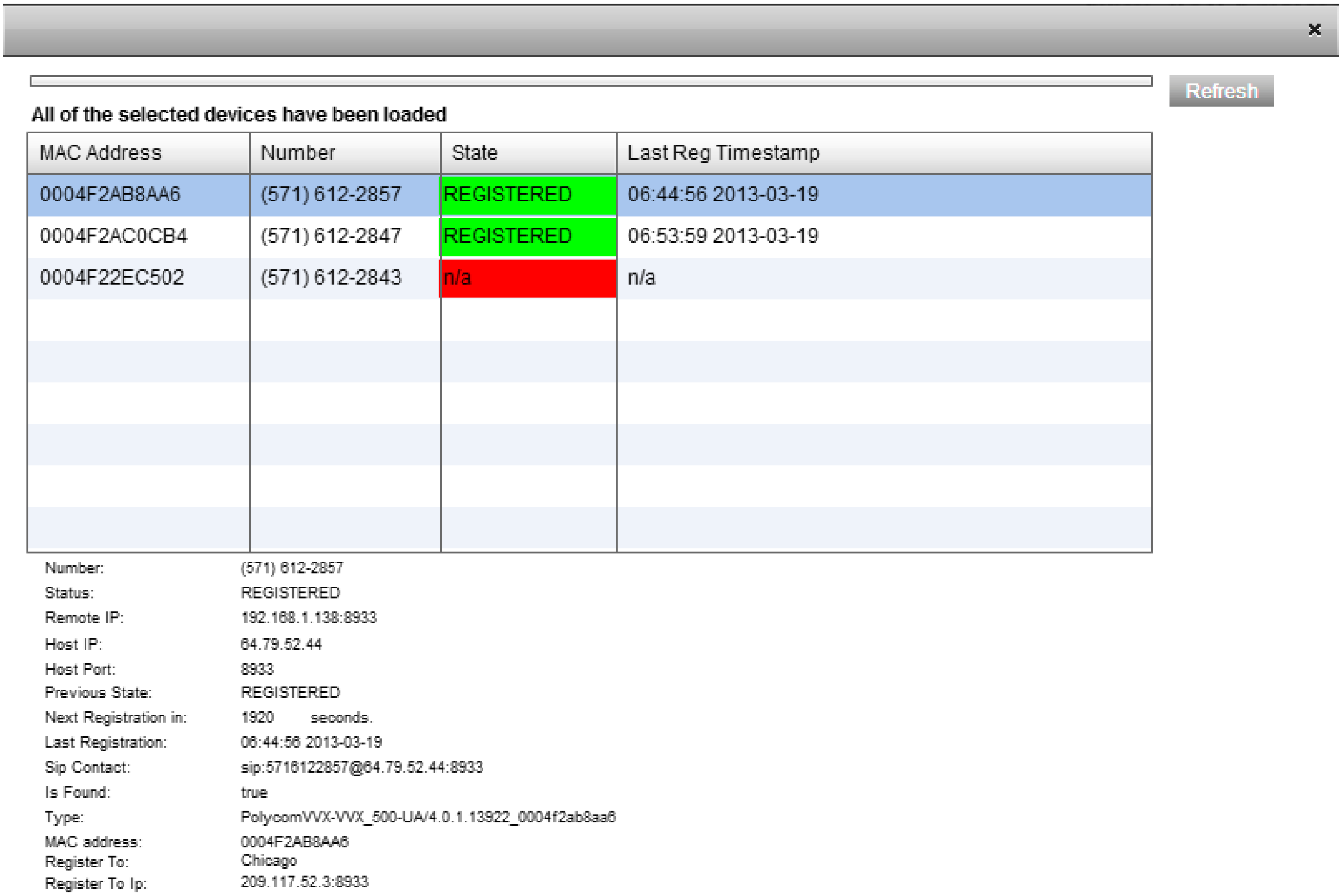


Figure 7: Showing device registration status in Device Management

You can select any line to see additional information, such as the last registration date and time.

Tip: Devices must register before they can place and receive calls; therefore, if a phone cannot do so, checking the registration is a good place to start.

Adding a user

A user in XO Hosted PBX is someone who has a phone number and a seat package assigned to them. This assignment is handled in the Phone Assignment panel in the MySite portal.

After you log in to MySite and select the *Phone Assignment* panel, you will see a screen similar to Figure 8.

Export

Phone Number	Station	Extension	Model	MAC	Port	First Name	Last Name
(469) 453-3508	Messaging Station	3508				XO ECC Central Demc	User 1
(469) 453-3509	Executive Office Seat	3509	Polycom SoundPoint IP 33	0004F2373191	1	Melissa	Johnston
(469) 453-3511	Executive Receptionist Se	3511	Polycom SoundPoint IP 65	0004F2350FAA	1	John	Haupt
(469) 453-3512	Analog Seat	3512	Polycom SoundPoint IP 65	0004F2350FAA	1	Steve	Jacklitch
(469) 453-3513	Alternate Number	3513				.	4694533513
(469) 453-3514	Alternate Number	3514				.	4694533514
(469) 453-3515	Conference Room Seat	8765	Polycom SoundPoint IP 45	0004F23927EF	1	Cynthia's Conference	4694533515
(469) 453-3516	Alternate Number	3516				.	4694533516
(469) 453-3517		3517				.	4694533517
(469) 453-3518	Alternate Number	3518				.	4694533518
(469) 453-3519		3519				.	4694533519
(469) 453-3520	Executive Office Seat	3520	Polycom SoundPoint IP 45	0004F23927EF	1	Brian	Levine
(469) 453-3521	Executive Receptionist Se	3521	Polycom SoundPoint IP 65	0004F23BCCCF	1	Scotty	Webb
(469) 453-3522	Analog Seat	3522	Polycom SoundPoint IP 65	0004F23BCCCF	1	.	4694533522
(469) 453-3523		3523				.	4694533523
(469) 453-3525		3525				.	4694533525
(469) 453-3526	Standard Office Seat	3526				.	4694533526
(469) 453-3527	Instant Group Call	3527				.	4694533527
(469) 453-3528		3528				.	4694533528
(469) 453-3529		3529				.	4694533529

Number of records in grid: 20 << < 1-20 of 35 > >>

Bulk Assignments Upload: Load .csv File | Station Inventory | Download .csv Template | Show Upload Results | Instructions

Figure 8: Phone Assignment panel interface

To add a user, do the following.

- Select the (unused) phone number that to allocate to them. A new window will open, like so (Figure 9):

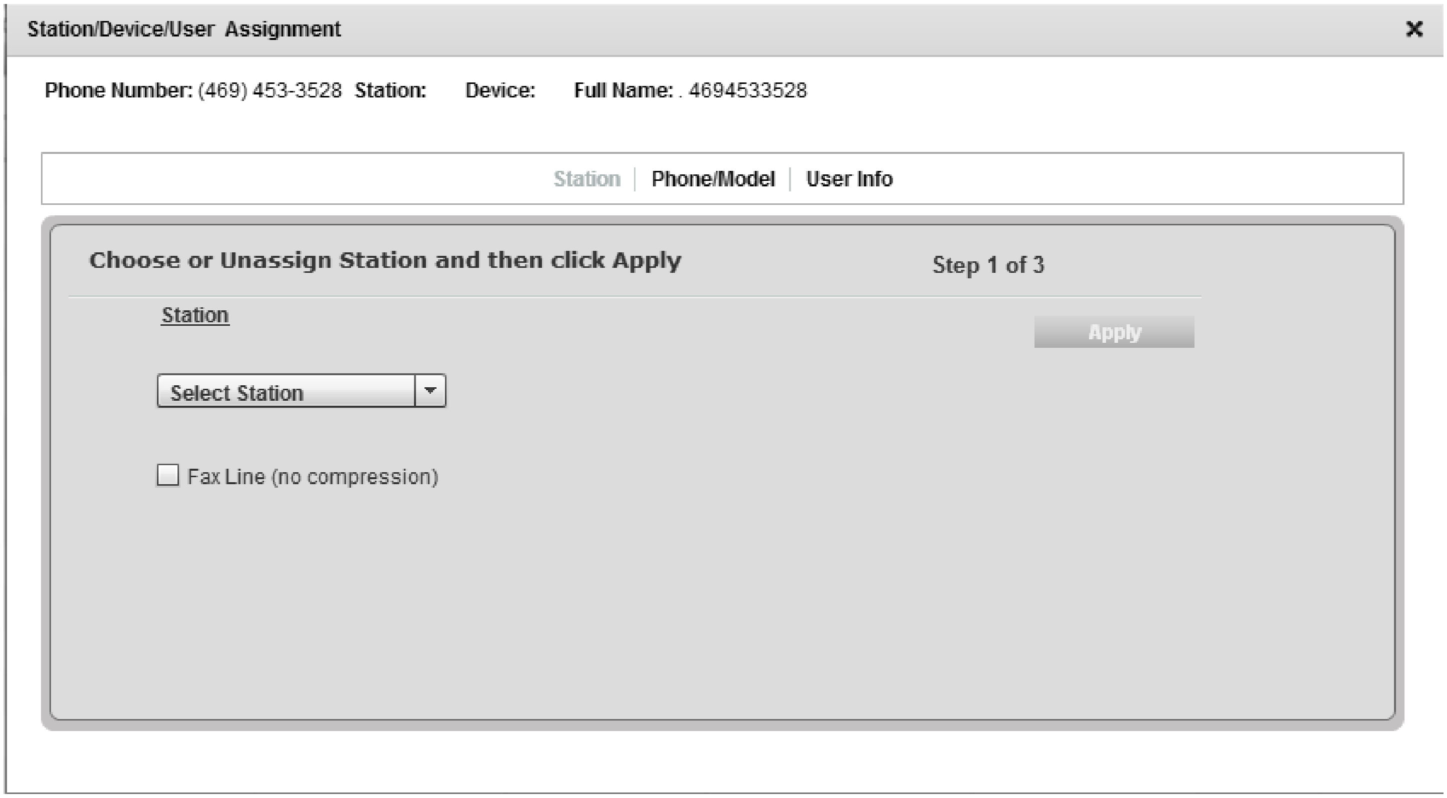


Figure 9: Interface panel after selecting an unused phone number

- Select the station (seat) type to assign to this user. (If you are adding an analog station for use by a fax machine, check the fax line box.) Click *Apply*.

You can only assign seat types that you have licensed from XO. To order more seats, contact your XO representative.

- Click on *Phone/Model*, and select the model, MAC address and port from the available choices. Click *Apply*.
- Click on *User Info*, and fill out the information.

  - First and Last Name
  - Caller ID number: defaults to the phone number, but you can change it to be an alternate, such as the front desk number, by clicking on the *Two-way list* dropdown and selecting the value. This is a common choice for sites where you want users to be able to make outbound calls without exposing their direct line number.
  - Extension: this defaults to the last four digits of the user's phone number.

    You can select an alternative, if you choose, or if the suggested value is unsuitable. The extension number can be any value of your choosing, subject to the following rules:

    - The extension must not already be in use
    - Extensions can be between 2 and 6 digits long (the default is four – the last four digits of the phone number)
    - Extensions with these values will not be allowed:

    - 0311, 1311
    - 0911, 1911, 911
    - 00, 011
    - N11, N11X, N11XX, N11XXX, where N is 2-9 and X is 0-9. For example, 311 and 411 would be barred.

  - User ID: this must be a unique value. The user will use this name to sign into the MyPhone web portal.
  - Email address: this address is where automatic emails from the system will be sent, e.g. for password resets.

- Click *Apply*. The user will receive an email at the address you specified inviting them to log in, set up a new password, and begin using the service.

Removing a user

When you remove a user, their phone number and seat are freed up for re-use, and their voicemails and other data are erased from the system.

To remove a user, do the following.

- Select their phone number in the *Phone Assignment* panel in the MySite interface.
- Click *User Info*, and click *Remove User*.
- Acknowledge the warning prompt. The user is removed.

Changing a user's seat type

In some circumstances you may wish to change a user's seat type. For example, you might have two employees who are switching roles, with one taking on Receptionist duties and one giving them up.

To change a user's seat type, you must

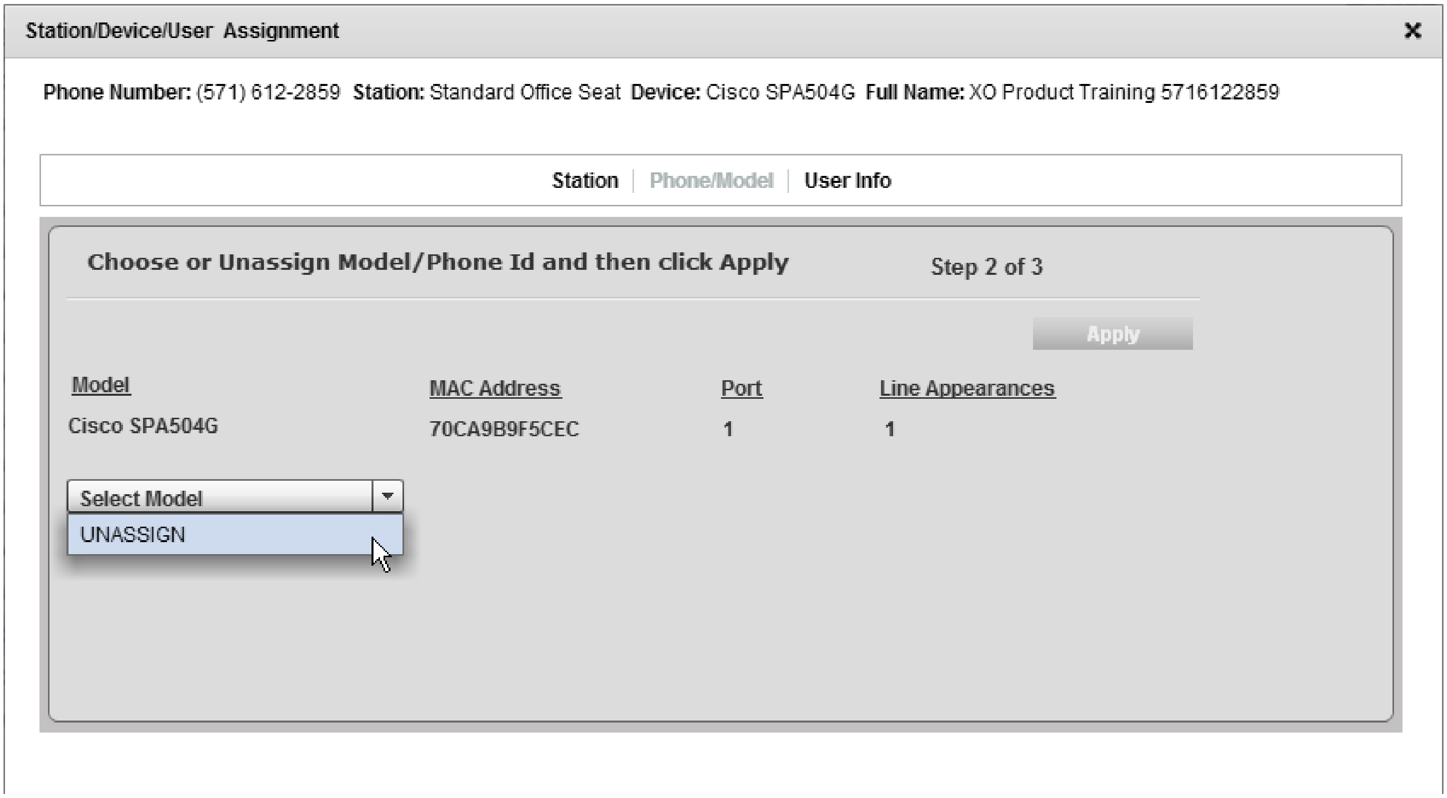
- Unassign their device
- Unassign their seat type
- Re-assign their device, and then select their new seat type

You can only assign seats from the pool of unallocated seats in the block that you obtained from your XO representative. However, new seats can be ordered at any time – contact your XO representative for details.

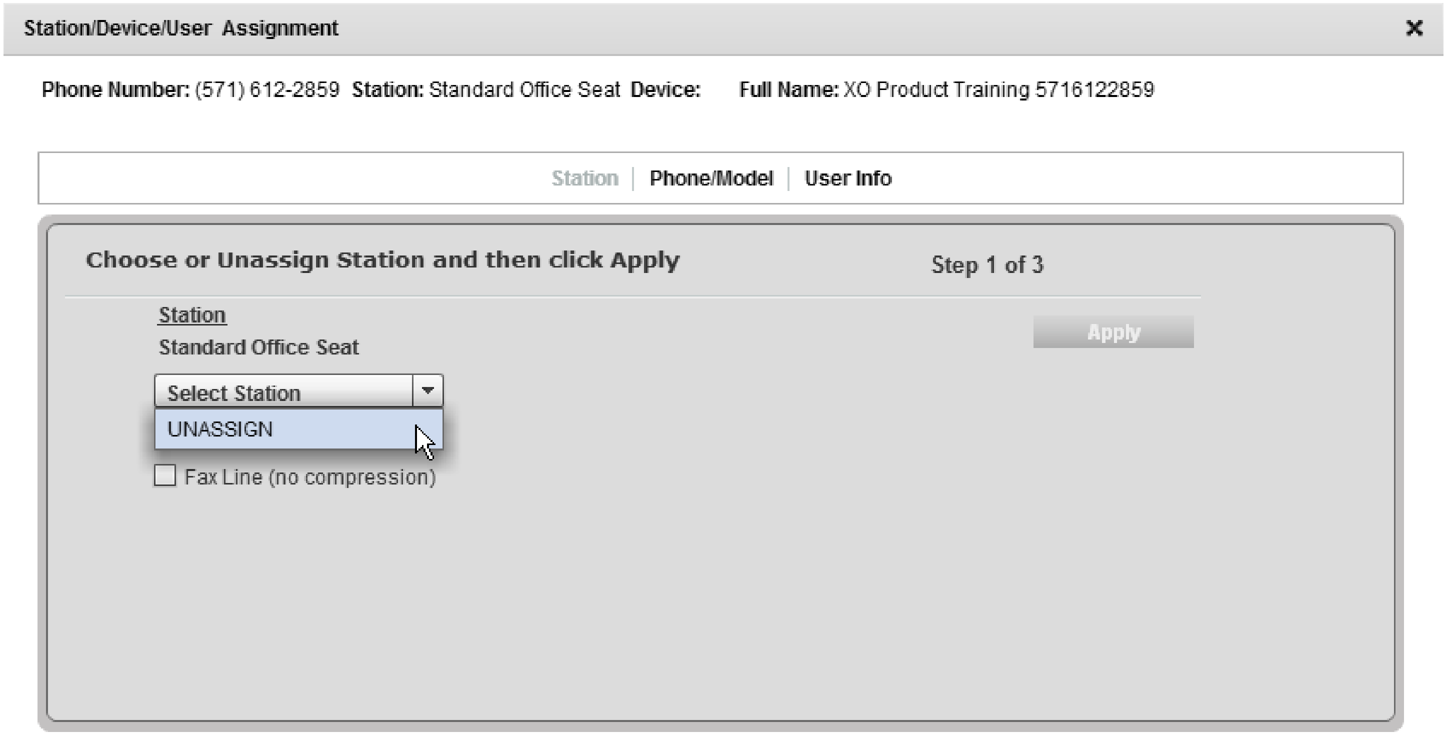
Tip: Before changing a seat type, remove any features that tie the line to other users, such as line sharing, monitoring or membership in hunt groups.

The step-by-step procedure to change a user's seat type is as follows.

- Select their phone number in the *Phone Assignment* panel in the MySite interface.
- Select *Phone/Model* in the window, and choose *UNASSIGN* from the *Select model* drop-down list:



- Click *Apply*.
- Click on *Station*, and choose *UNASSIGN* from the *Select Station* drop down list.



- Click *Apply*.
- Select *Station* again and choose the new seat type from the drop down. Click *Apply*.
  - You may be prompted to decide whether to erase any existing voicemail for this user, or keep them. In any case, the voicemail PIN will be reset. The user will receive an email from the system with the new PIN.
- Select *Phone/Model* and select the device model the user should use, along with its MAC address and other settings. If the user is simply changing roles and not phones, this will be the same device as they used before.
- Click *Apply* and close the window.

Auto-Attendants and Hunt Groups

Building an Auto-Attendant

An auto-attendant is a system to automatically answer incoming calls and offer choices to route the call to its final destination. For example, callers might hear a greeting such as “Welcome to MyCorp. For Sales, please press 1. For Support, please press 2. If you know your party’s extension, please dial it now.”

XO Hosted PBX includes one free auto-attendant. You can order more via your XO representative, if needed. Auto-attendants can also be nested, so that a key press in one attendant leads into another.

To configure an auto-attendant, navigate to the *Site Services* tab and select *Auto Attendant* from the list on the left. You will see a list of auto-attendant numbers in the main panel. These numbers are the pilot or lead numbers of the attendant(s); that is, the numbers that callers dial that take them to the start of the menu tree.

To configure the attendant, click on the lead number. A new window will open, as in Figure 10.

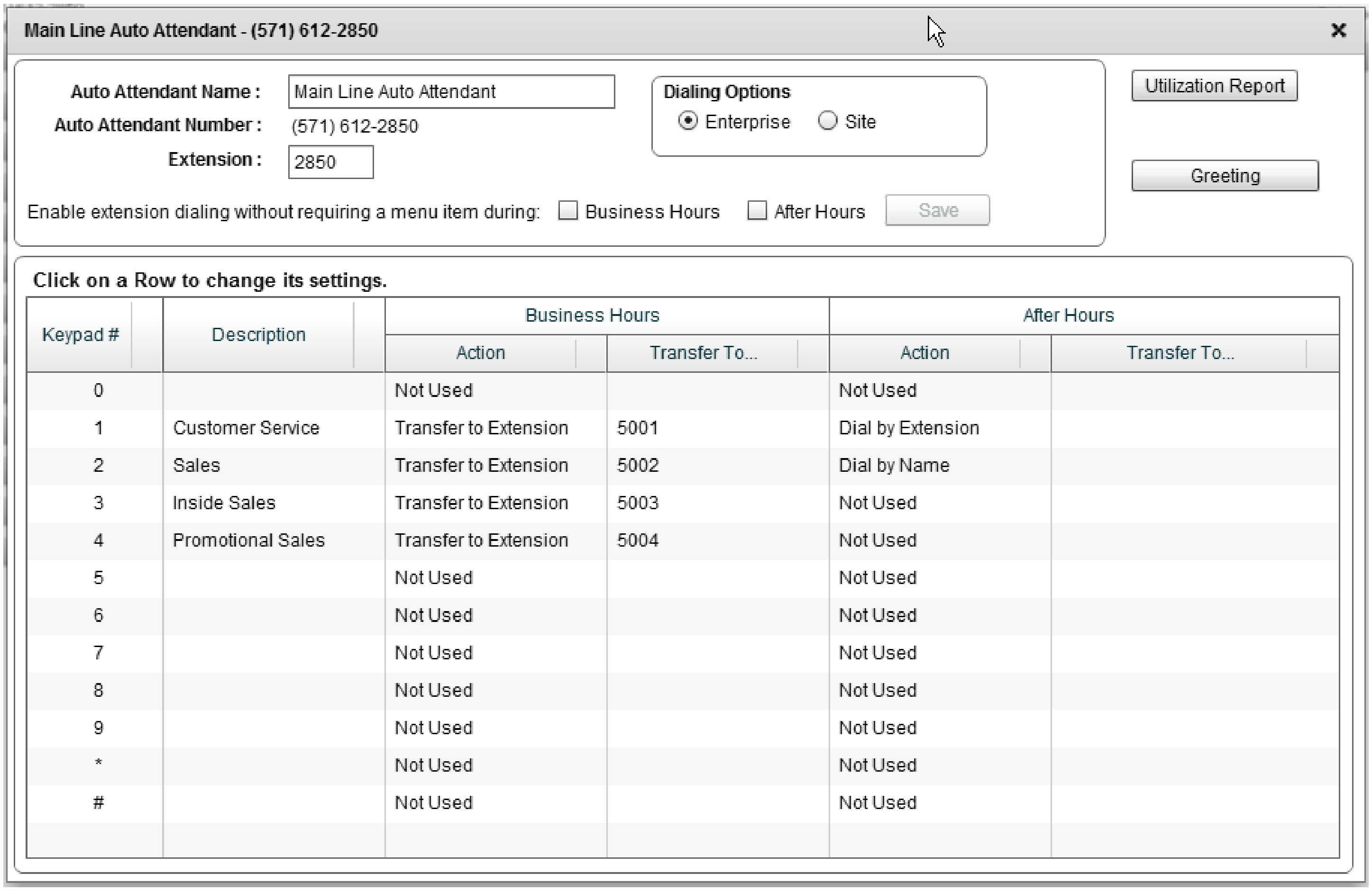


Figure 10: Auto-attendant configuration screen

When a user reaches an auto-attendant, they will hear your greeting and be able to press a key – any one of the ten digits zero through nine, plus the star (*) and pound/hash (#) keys. Click on the line corresponding to a key to configure the action to be taken. Figure 11 shows the allowable actions.

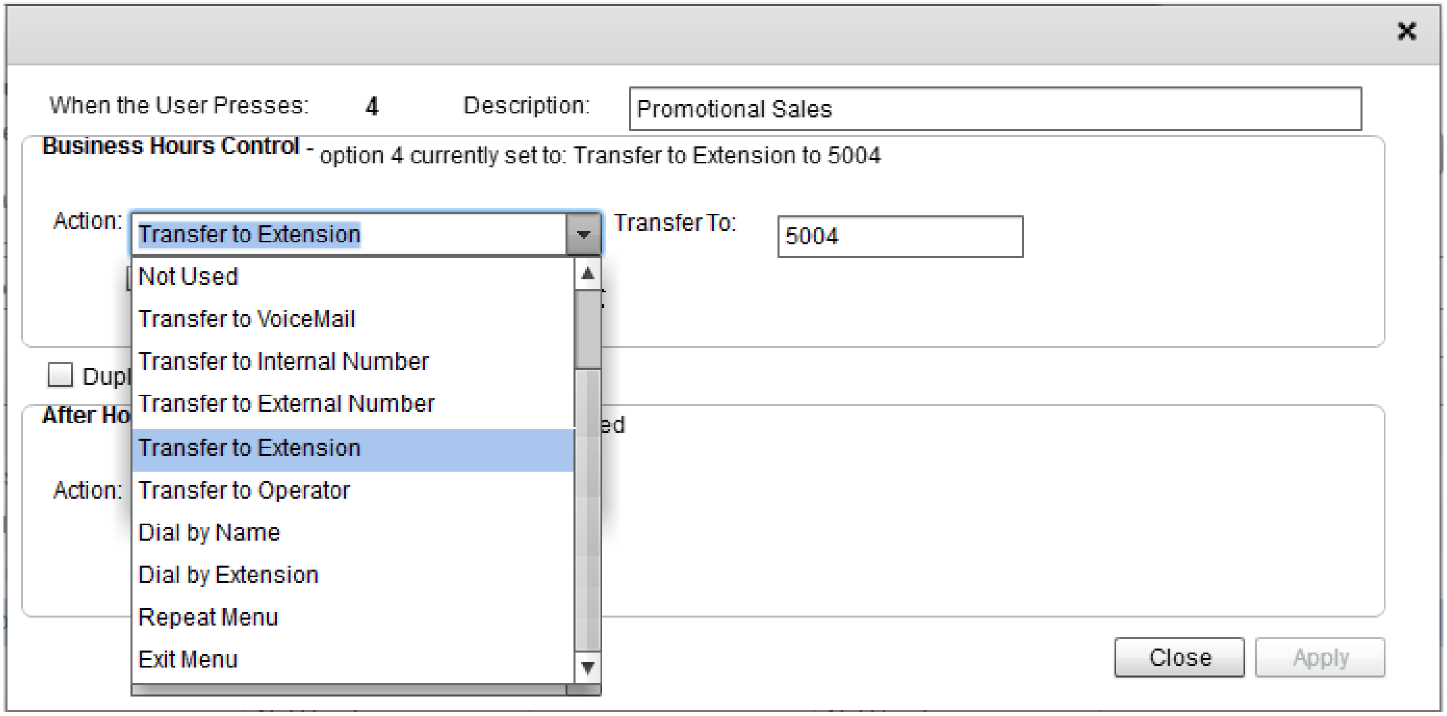


Figure 11: Allowable actions following a key press in an auto-attendant

The allowable actions are as follows.

- Transfer to voicemail, an internal number, extension or an external number.
- Transfer to an operator (i.e. an internal user who can manually route the call, such as a receptionist or front desk).
- Dial by name (where the caller uses their phone keypad to spell all or part of a user's name).
- Dial by extension (where the caller dials the recipient's extension).
- Repeat and exit the menu.

After selecting the option, you must decide whether the same action should apply when the caller reaches the auto-attendant out of hours (specifically, outside of the opening hours schedule you defined previously).

- If the answer is yes, check the box labeled "*Duplicate for After Hours*".
- Otherwise, leave the box unchecked and define an alternate target in the *After Hours* section of the window.

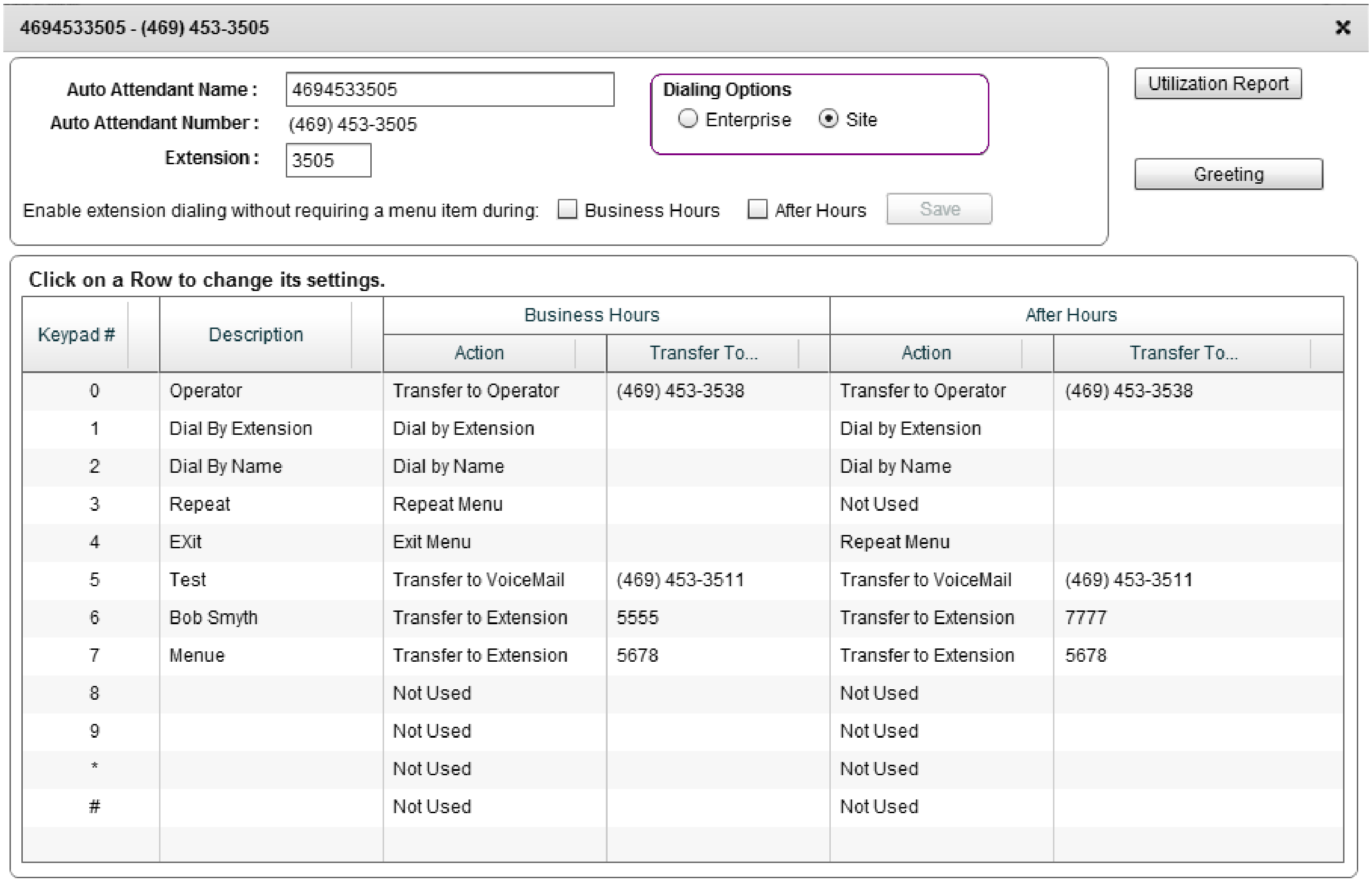
To save your changes, click *Apply*. You can now repeat the sequence for any other key presses.

Tip: Improve your customers' experience by making standard choices across menus. For example, many users expect the 0 (zero) key to transfer them to a human, so set key press zero to "transfer to operator".

Dial by Extension and Dial by Name

Your auto-attendant can offer callers the choice of dialing a user's extension directly, or using the keypad to spell out the first or last name of the recipient and having the system connect them. In the latter case, users dial the numeric equivalent of names using the letters displayed on each key (ABC = 2, DEF = 3, etc.) For example, to dial "Jean Koslowski", a user could dial 5-3-2-6 (J-E-A-N) or 5676 (the first few letters of Koslowski: K-O-S-L).

As the administrator, you must decide whether these inputs are matched against all the names and extensions in your enterprise, or restricted to only searching in your site. To configure your choice, select the auto-attendant to configure as described above and select the Enterprise or Site radio button in the dialing Options section, as illustrated in Figure 12.



Keypad #	Description	Business Hours		After Hours	
		Action	Transfer To...	Action	Transfer To...
0	Operator	Transfer to Operator	(469) 453-3538	Transfer to Operator	(469) 453-3538
1	Dial By Extension	Dial by Extension		Dial by Extension	
2	Dial By Name	Dial by Name		Dial by Name	
3	Repeat	Repeat Menu		Not Used	
4	EXit	Exit Menu		Repeat Menu	
5	Test	Transfer to VoiceMail	(469) 453-3511	Transfer to VoiceMail	(469) 453-3511
6	Bob Smyth	Transfer to Extension	5555	Transfer to Extension	7777
7	Menue	Transfer to Extension	5678	Transfer to Extension	5678
8		Not Used		Not Used	
9		Not Used		Not Used	
*		Not Used		Not Used	
#		Not Used		Not Used	

Figure 12: Dialing Options radio button

Setting the Auto-Attendant Greeting

You must record a greeting that plays callers the allowable choices. For example, “Welcome to SampleCo, For Sales, please press 1. For Support, press 2. If you know your party’s extension, press 3. To repeat this menu, press 9. To speak with an operator, press 0.”

You can record your greeting over the phone if you wish, but it is generally more convenient to pre-record the greeting elsewhere and use the MySite portal to upload it.

To configure an auto-attendant, navigate to the *Site Services* tab and select *Auto Attendant* from the list on the left. Select the lead number of the auto-attendant and click the *Greeting* button. The interface looks like Figure 13.

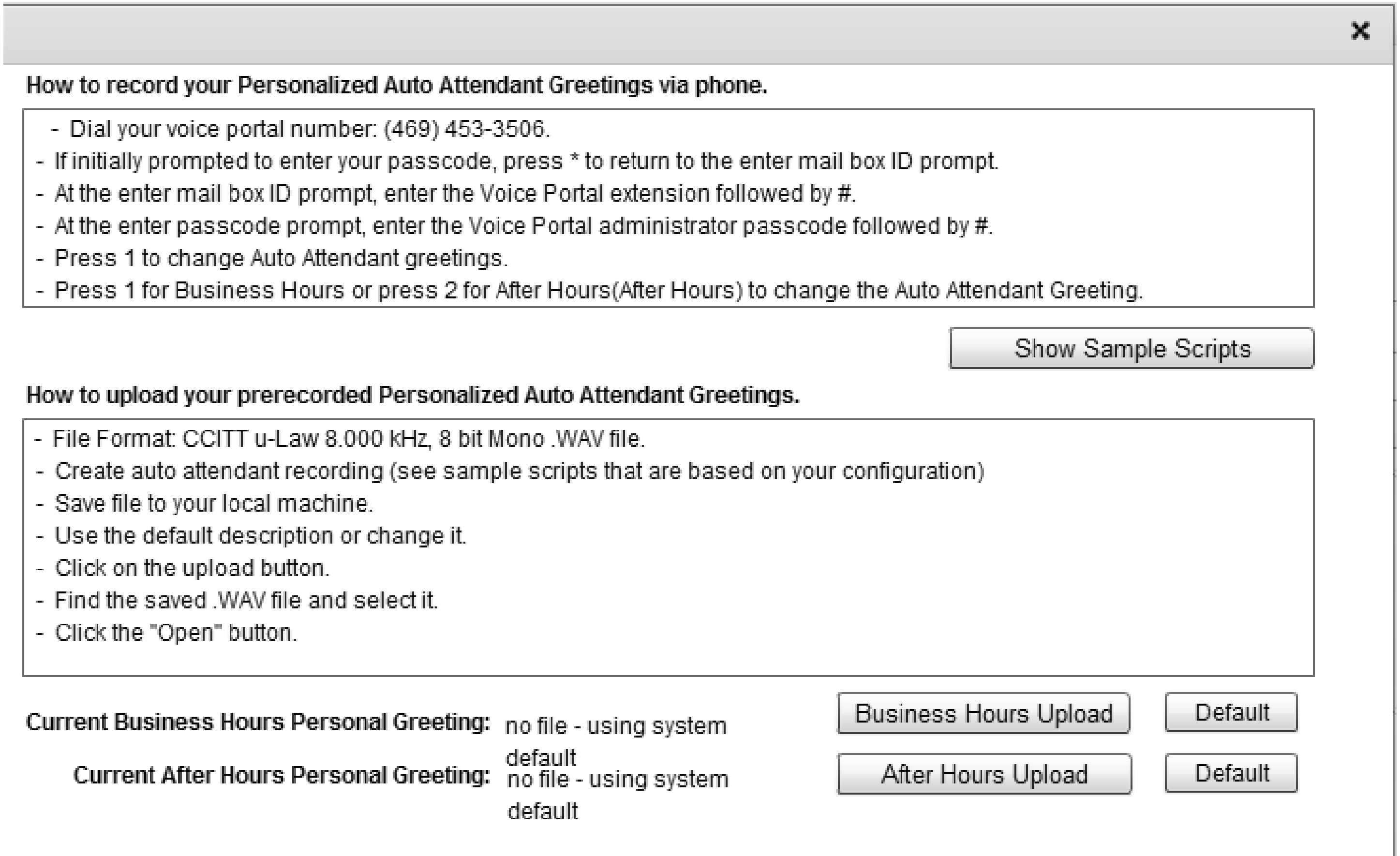


Figure 13: Auto-attendant greetings interface

Click on one of the two *Upload* buttons, select the file containing the audio, and choose *Open*. The file is uploaded and is automatically made available for use.

The audio must be in a specific technical format: CCITT u-law, 8bit mono, 8KHz WAV format. Remember: you are legally responsible for any audio that you upload to the system. For example, you cannot use audio produced by another entity without prior authorization by the copyright owner. Consult an attorney if you are unsure.

Tip: It is good practice to keep your greeting short – no more than 30 seconds – to avoid callers becoming bored and hanging up.

Configuring a Hunt Group

A Hunt Group is an option that sends an incoming call to one or more users in your site. The XO Hosted PBX platform receives the call and searches, or hunts, for a user who can take the call. For example, if you had three employees on the Support desk in your site, calls to a support hunt group number could ring the first employee, and if they didn't answer, ring the second, and then the third.

Tip: Hunt Groups are a simple form of automatic call distribution, or ACD. More advanced forms include **Call Queuing**, available within XO Hosted PBX, and **Contact Center on Demand**, a premium offering intended for dedicated call centers. Ask your XO representative for more information on these options.

Your XO implementation team will have built your first hunt group when they turned up your service. (If you need additional hunt groups, contact your XO representative.)

To configure your hunt group, select the *Hunt Group* item in the *Site Services* tab. You will see one or more hunt groups listed. Select the one you wish to modify. The resulting interface is illustrated in Figure 14.

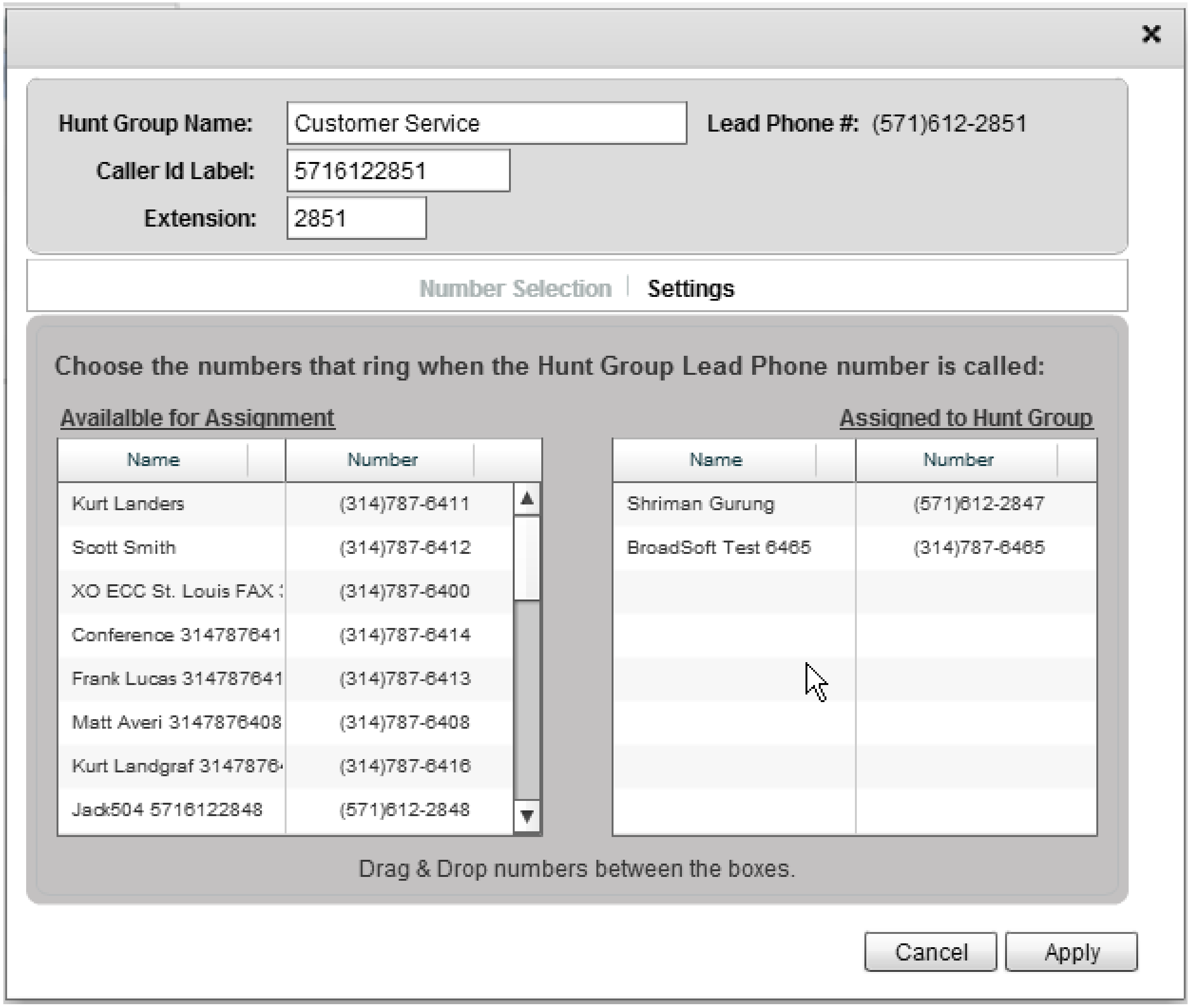


Figure 14: Hunt Group interface panel

- To make a user a member of the hunt group, select their line in the left-hand "*Available for Assignment*" panel and drag and drop it into the right-hand "*Assigned to Hunt Group*" list. Click on *Apply* to save the changes.
- To change the order in which members' phones ring, select the *Settings* option. You can choose how the phones should ring, so that either

- all phones ring simultaneously
- each phone rings in turn, in the order the numbers were assigned in the “Assigned to Hunt Group” list
- each phone rings in turn, in a circular sequence (e.g. “1, 2, 3, 1, 2, …”)
- calls are evenly distributed amongst members
- calls are distributed according to a weighting.

Figure 15 shows an example.

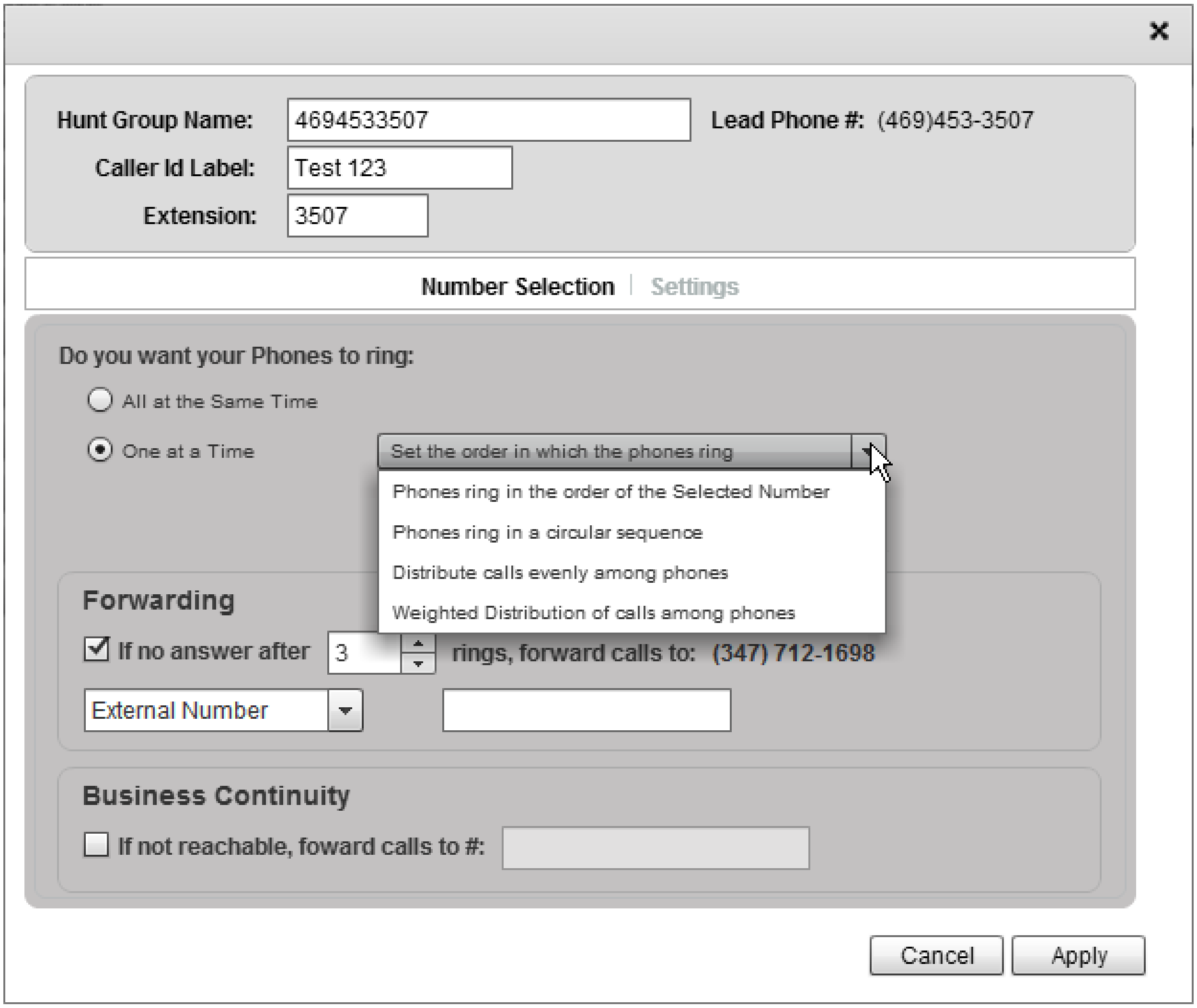


Figure 15: Selecting the order that Hunt Group phones ring

As a last resort, you can configure a target number to call if no one in the hunt group answers, or, worse still, if the site is unavailable (e.g. due to a power failure). Simply complete the *Forwarding* and *Business Continuity* options as desired and click *Apply* to save.

If you chose to deliver calls using a weighted distribution, clicking Apply will open up a new window, as illustrated in Figure 16.

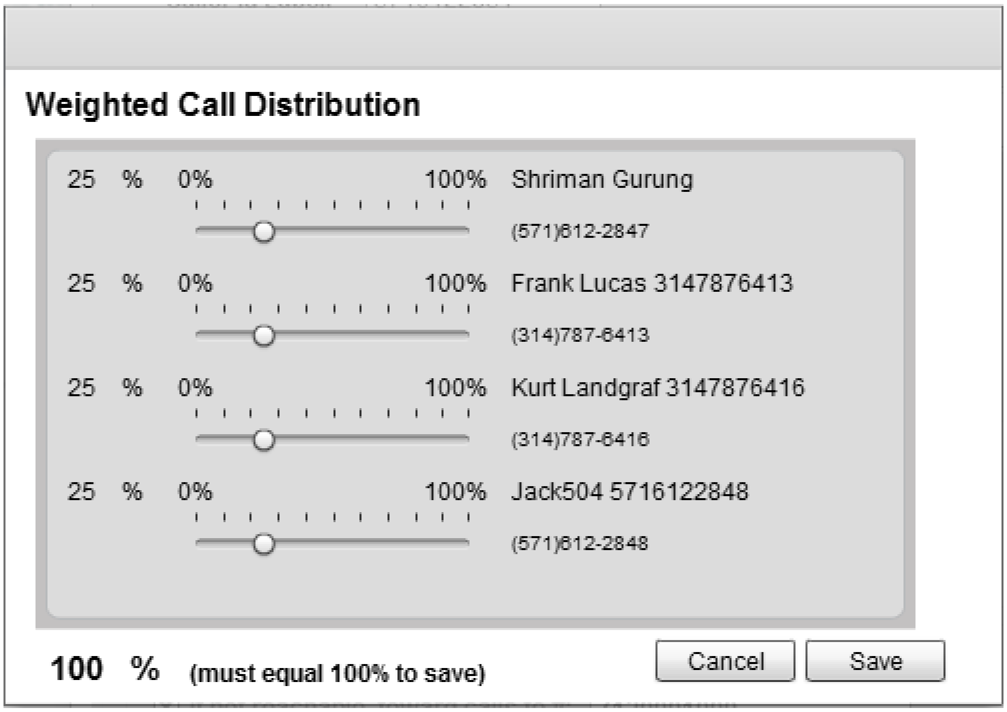


Figure 16: Weighting calls to hunt group members

Drag the sliders according to the percentage of calls that you want each member line to receive. Make sure that the total adds up to 100% and then click *Save.*

Changing Lead Numbers

A lead number or pilot number is a phone number that represents a service such as an auto-attendant, instant group page or hunt group. For example, if you wanted all calls to your business to be answered by an auto-attendant, you would publish the lead number of the attendant on all your business stationery, so that callers to your business would arrive at the attendant menu by default.

To review the lead numbers in use, log in to MySite, navigate to the *Site Services* tab, and choose *Lead Numbers* from the list on the left. You will see a screen similar to Figure 17.

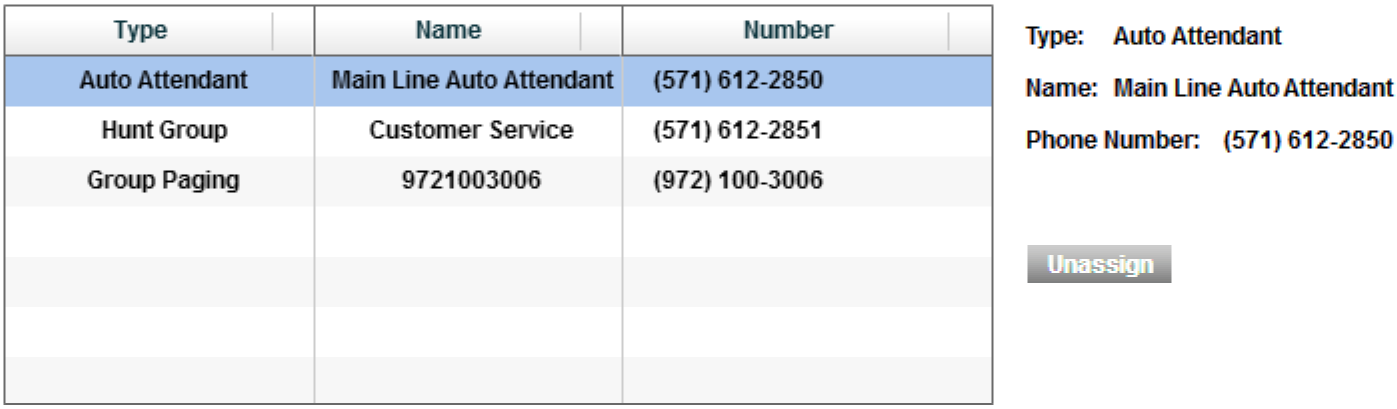


Figure 17: Lead numbers user interface panel

To change a lead number, select the corresponding line and choose *Unassign*. The entry in the number panel will change to a drop-down menu showing all the un-allocated numbers that you can choose from. Select the new number from the list.

You can offer your customers the choice of calling you via toll-free numbers. Contact your XO representative for details. If you use such numbers, however, do not change your lead numbers without contacting XO, or else calls to the toll-free number will not be automatically mapped to the new lead number that you chose.

Configuring Music-On-Hold

Music on hold is the audio that users hear when their call is put on hold. For example, if a customer calls your site and the recipient transfers them to a colleague, the former will hear this audio for a few seconds whilst the transfer is completed.

XO Hosted PBX already includes a piece of music, or you can supply your own. (Of course, it need not be music: you might prefer callers hear a message, such as a promotion for your website or latest offer.) The audio must be in a specific technical format: CCITT u-law, 8bit mono, 8KHz WAV format, and may be no more than 90 seconds in length.

Remember: you are legally responsible for any audio that you upload to the system. For example, you cannot use audio produced by another entity without prior authorization by the copyright owner. Consult an attorney if you are unsure.

To upload your audio, navigate to the *Site Services* tab, and select *Music On Hold* from the list on the left. Check the *Custom* radio button, and then the Music Upload button. Locate and select the audio file that you want to use. Once the file has been uploaded click Save.

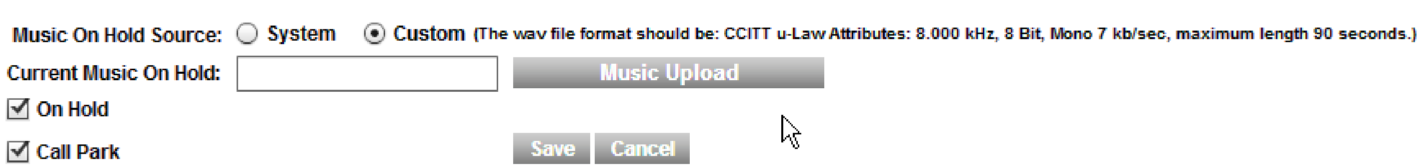


Figure 18: Music on hold upload interface

Call Queuing (Call Center)

Call queuing is an optional service that extends the Hunt Group function into acting like a call center. Where hunt groups allow a group of users, or “agents”, to handle incoming calls received by at the hunt group’s lead number, Call Queuing adds another dimension by providing an automated answer for all calls, with customizable greetings, comfort messages, and hold music for the caller to hear while held in a queue until an agent (assigned user) is available to pick up the call.

Your XO implementation team will have built your call queue(s) when they turned up your service. If you need additional queues, contact your XO representative.

To configure a call queue, select the *Call Queue* item in the *Site Services* tab. You will see one or more queues listed.

Select a row and click the configure button to change the Call Queue settings.

Configuration

Call Queue Name	Lead Phone #	Extension	Policy	Active
Tech Queue	(555) 500-0501	0501	Regular	☑
Sales Queue	(555) 500-0502	0502	Simultaneous	☑

Figure 19: Selection panel for Call Queues

Select the one you wish to modify and press the *Configuration* button. The resulting interface is illustrated in Figure 20.

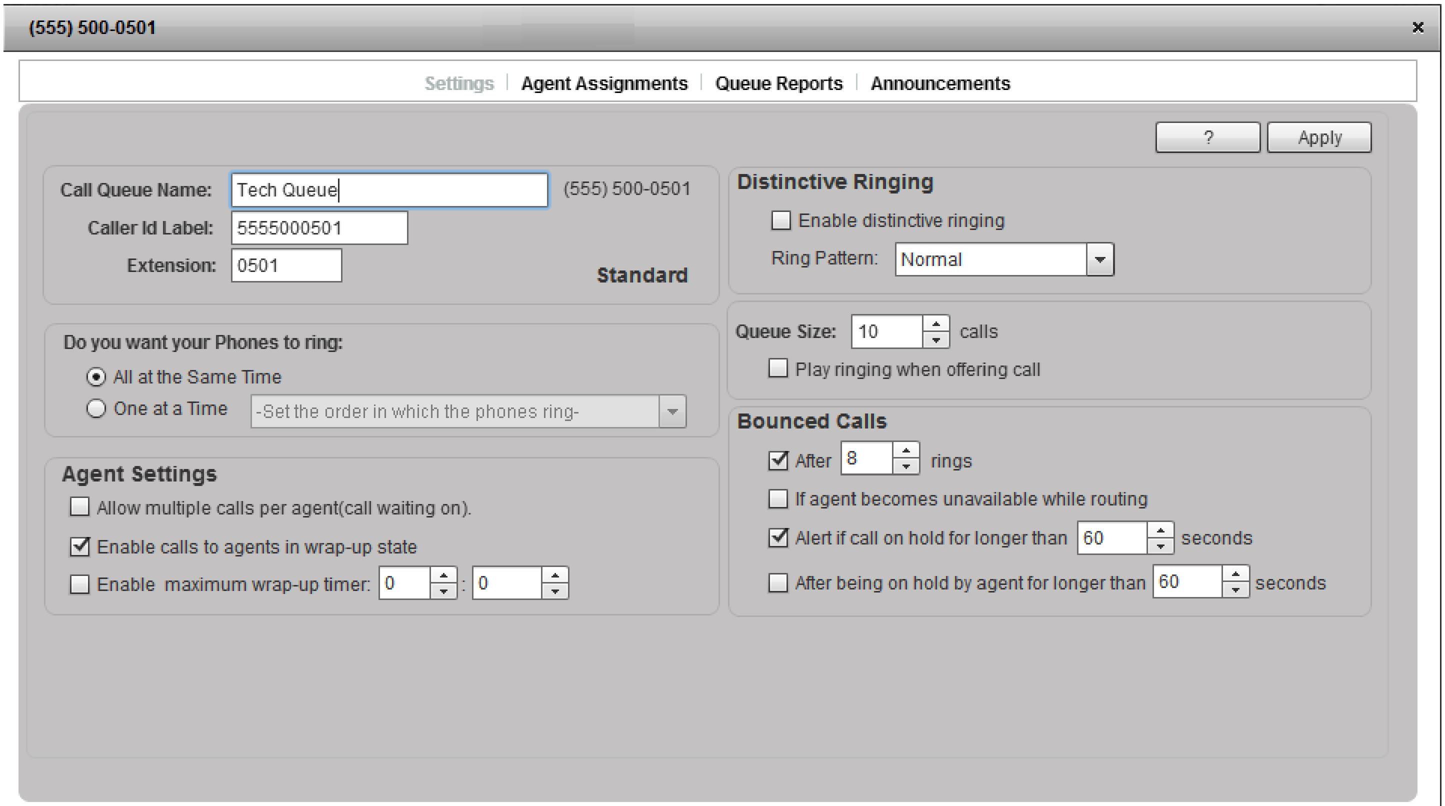


Figure 20: Queue settings (Call Queueing feature)

The interface contains four areas: *Settings*, *Agent Assignments*, *Queue Reports* and *Announcements*. Each is described in the following sections.

Settings

After filling in the name, caller ID and extension number assigned to the queue (which must be a unique, unused value), you can set the options that configure how calls are handled.

Choosing how phones should ring

You can choose how the phones should ring, so that either

- all phones ring simultaneously
- each phone rings in turn, in the order the numbers were assigned in the “Agent assignments” list (“*In order of selection*”)
- each phone rings in turn, in a circular sequence (e.g. “1, 2, 3, 1, 2, …”) (“*Circular*”)
- calls are evenly distributed amongst members
- calls are distributed according to a weighting.

Check the All at the Same Time radio button to make an incoming call ring all phones simultaneously. Otherwise, select *One at a Time* and select the order that the phones ring from the drop down.

Setting wrap-up and call waiting behavior

The wrap-up period is a time following completion of a call when an agent does administrative tasks related to the call. For example, your agents might enter notes about the call into a line of business application in use at your site.

You can choose how long this period is, and whether the Hosted PBX platform will deliver calls to agents in this state, using the interface panel. Check the *Enable maximum wrap-up timer* box and choose the period (minutes:seconds) of the timer to use a wrap-up period. To deliver calls to agents in wrap-up state, check the *Enable calls to agents in wrap-up state* box.

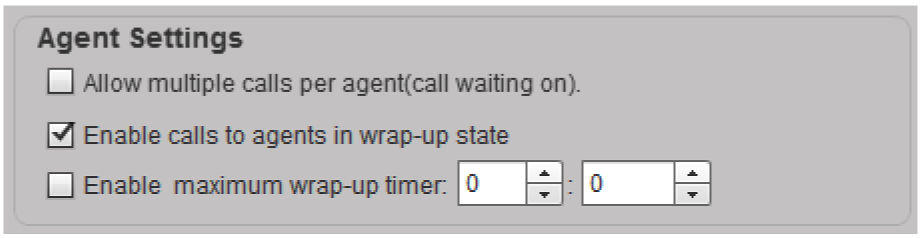


Figure 21: Agent settings for a queue

If desired, you can also enable call waiting for agent lines, so that multiple calls can be delivered to an agent's phone even if they are already busy with another call. To enable this, check the *Allow multiple calls per agent (call waiting on)* box.

Distinctive ringing

Distinctive ringing allows your users to determine when an incoming call is being delivered from a queue versus being direct-dialed. This allows your users to, for example, speak a different greeting when they pick up the phone, such as "Thank you for calling Tech Support, this is John, how can I help?" as opposed to "This is John Doe".

To enable the feature, check the *Enable distinctive ringing* check box and select the ringing pattern from the drop-down box. For example, you might choose *Short-Short-Long*. Figure 22 shows the interface.

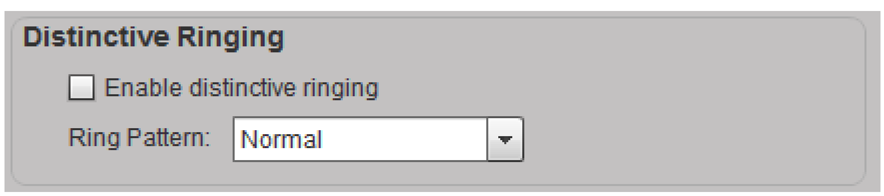


Figure 22: Distinctive ring settings for a queue

Queue size

Queue size is the maximum number of unanswered calls that can be active at any time. There is no single "best" value for this parameter – it will vary by the size of your agent population, how quickly they can process calls, and the willingness of your callers to wait in line. The maximum is 50.

To set the maximum queue size, select the *Queue size* value in the interface. You can also elect to play the caller ringing tone just before delivering a call to an agent, so that they get an audible indication that their call is about to be answered. Check the *Play ringing when offering call* box for this feature.

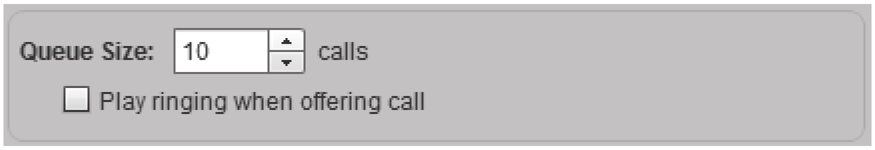


Figure 23: Queue size settings

Bounced calls

A bounced call is one that should have been answered, but for some reason needs to be returned to the queue and another agent found to handle it. This could be because

- the call was sent to the agent, but they did not pick up
- the call was sent to the agent, but the agent went unavailable (they switched on do-not-disturb) just as it was being sent
- the call was delivered, and then put on hold for an unusually long time.

The interface looks like Figure 24.

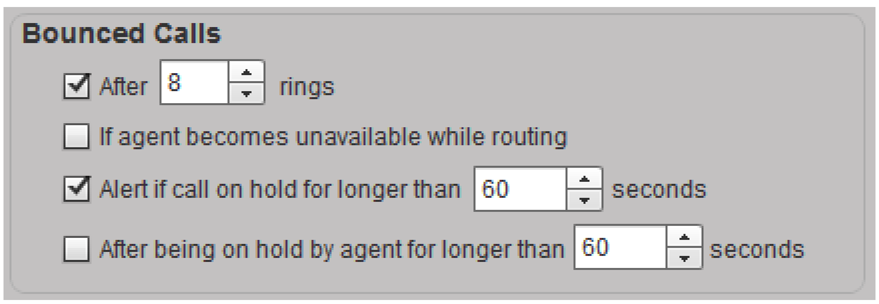


Figure 24: Bounced call settings

Common settings are

- Bounce call: *After* seven or eight rings (this is about 45 seconds)
- Bounce call *if agent becomes unavailable while routing*
- *Alert if call on hold for longer than:* 30 seconds
- *Bounce call: After being on hold by agent for longer than:* 60 seconds

Saving your Settings

Click Apply to save your settings.

Agent Assignments

Agents are users in your site who take calls for one or more queues. An agent can be a member of one or more queues. To make a user an agent in your queue, select the Agent Assignments label at the top of the panel and do the following:

- Check the Agent box to make this user an agent
- Drag and drop the user’s entry from the left hand *Available Users/Agent* panel to the right hand *Assigned Agents* panel.

Figure 25 shows the interface.

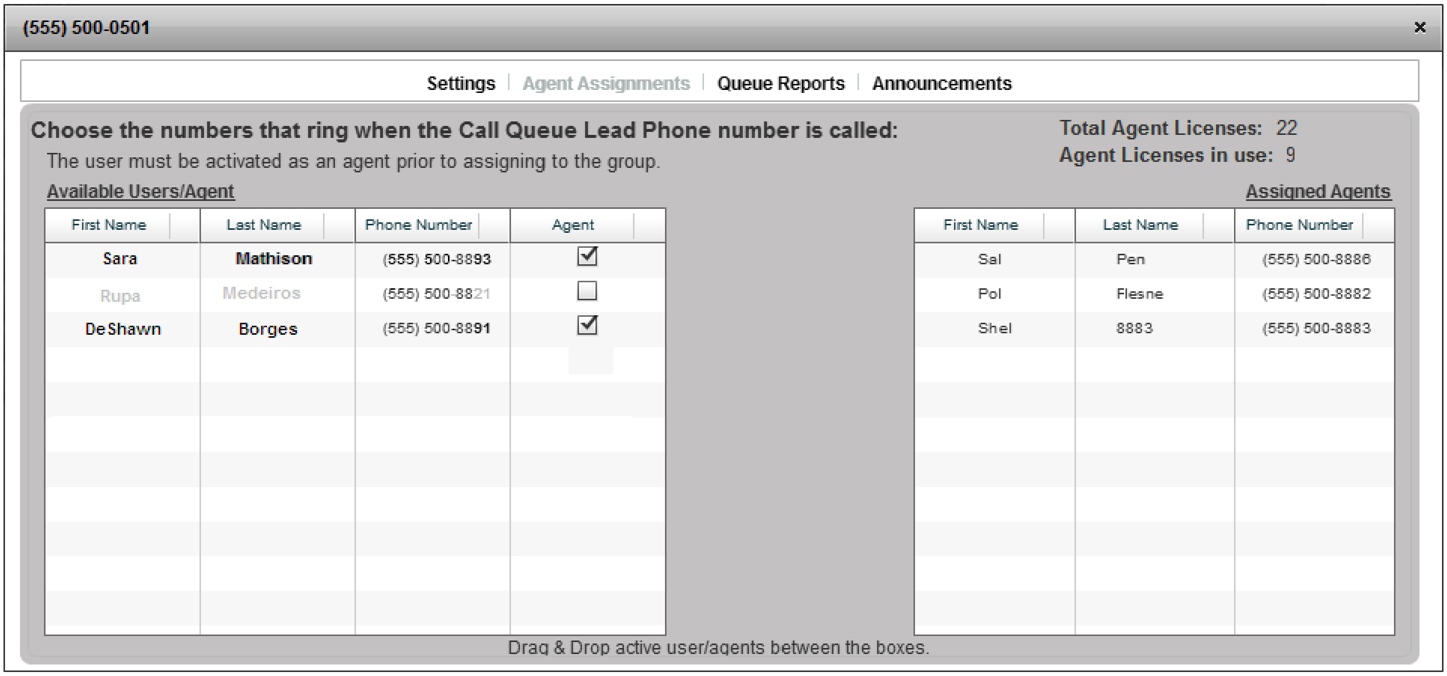


Figure 25: Assigning agents to queues

Remember, call queue agents are licensed separately in XO Hosted PBX. You must have a license for each agent. To order additional licenses, contact your XO representative.

Announcements

Announcements are what callers hear when their call reaches a queue. There are five types:

- Entrance
- Wait
- Comfort
- Hold
- Overflow.

To configure each one, click on the *Announcement* label at the top of the panel and then the button on the left hand side. Each is described in the sections that follow.

Note: The following sections include instructions for uploading your own audio files. The audio must be in a specific technical format: CCITT u-law, 8bit mono, 8KHz WAV format. Remember, you are legally responsible for any audio that you upload to the system.

Entrance Announcement

The Entrance Announcement is what callers hear when their call enters the queue. It is optional, according to whether the *Play Entrance Message* checkbox is selected. If an agent is already available at the point the call enters the queue, you can elect whether to play the message regardless by checking the "*Entrance message is mandatory when played*" box.

XO Hosted PBX includes a default message, but you will likely want to record and upload your own. You can upload up to four messages. To do so, select the *Custom* radio button and then the number "1" in the table shown in Figure 26. You will be prompted to select the file. Repeat for messages 2, 3 and 4 if required, and then hit *Apply*.

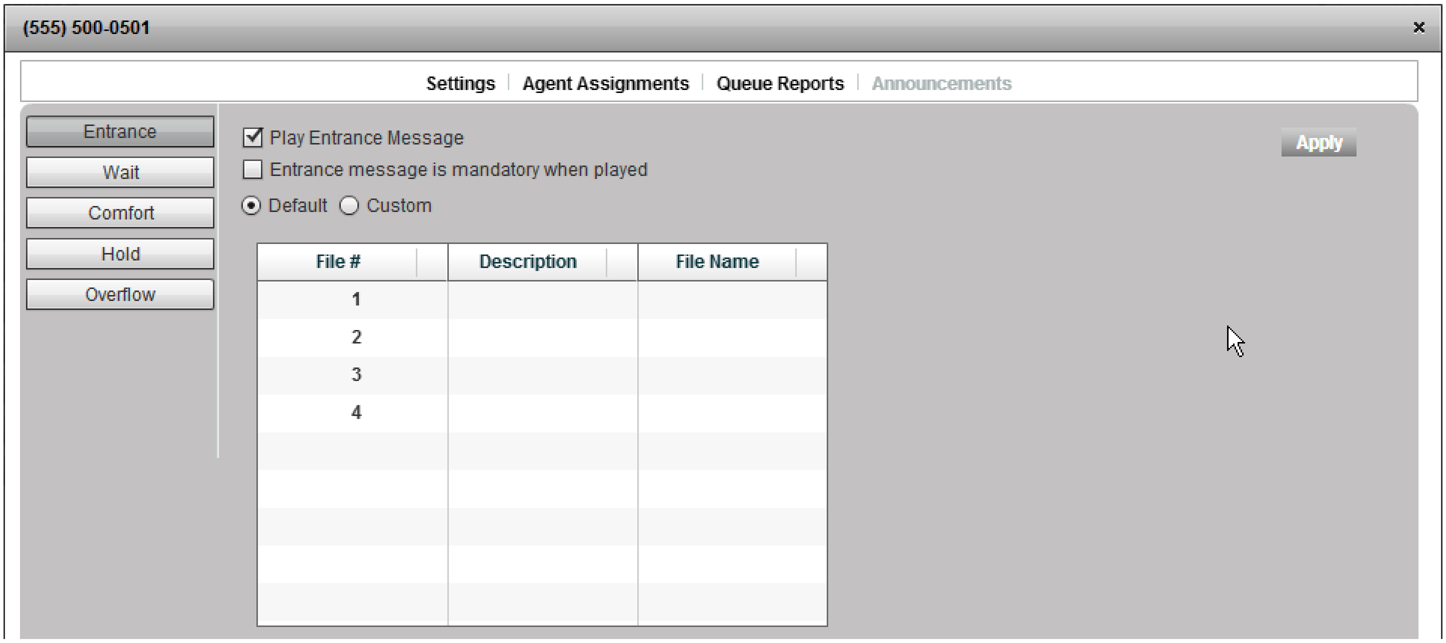


Figure 26: Entrance Announcement configuration screen

Wait Announcement

This optional announcement tells the caller either their estimated wait time or their position in the queue. The interface is illustrated in Figure 27.

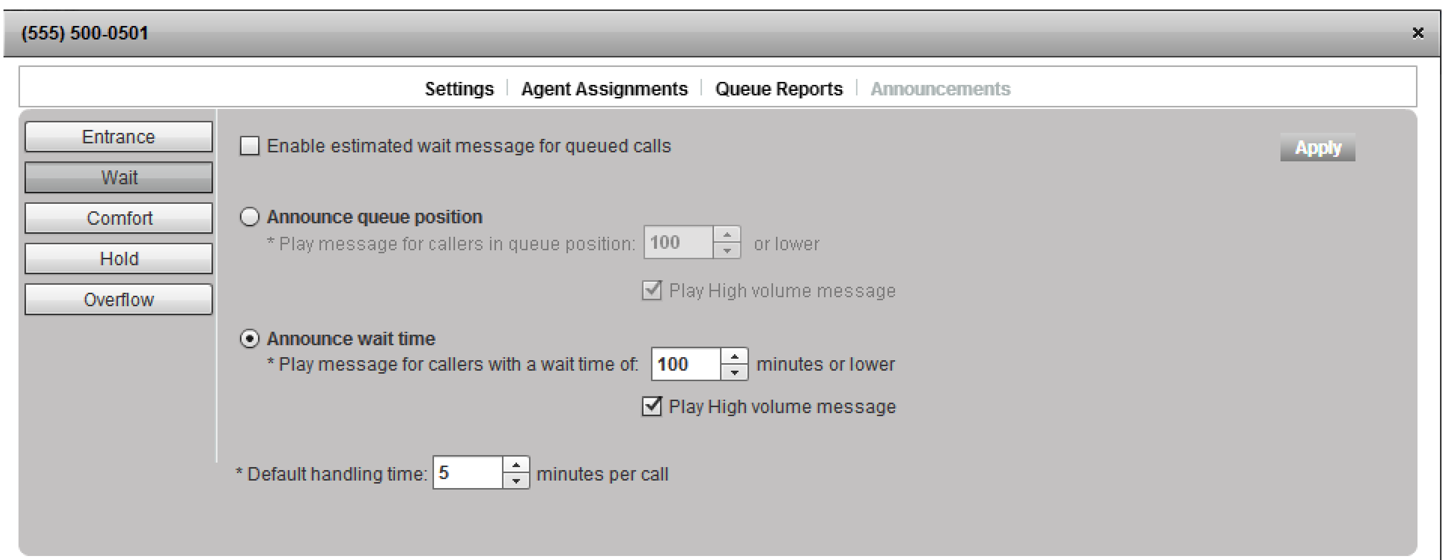


Figure 27: Wait Announcement configuration

- To enable the message, check the *Enable estimated wait message for queued calls* box.
- Choose one of the two radio buttons to announce either the queue position or wait time.
- Use the entry boxes to tune when the announcement is played: either further behind than the Nth position in the queue, or subject to a wait time longer than X minutes.

  In both cases you can also enable a generic “we are currently experiencing high call volumes” message.
- The system’s estimates of wait times depend on the time for an agent to complete a call. Set your best estimate using the input box in the lower part of the panel.
- Click *Apply* to save your changes

Comfort Message

A comfort message reassures the caller that their call is still active. It plays intermittently on a schedule to break up the hold message/music. You can use it to play helpful messages to your callers: for example, a support line might advertise the resources available on your website.

The interface panel is similar to Figure 28.

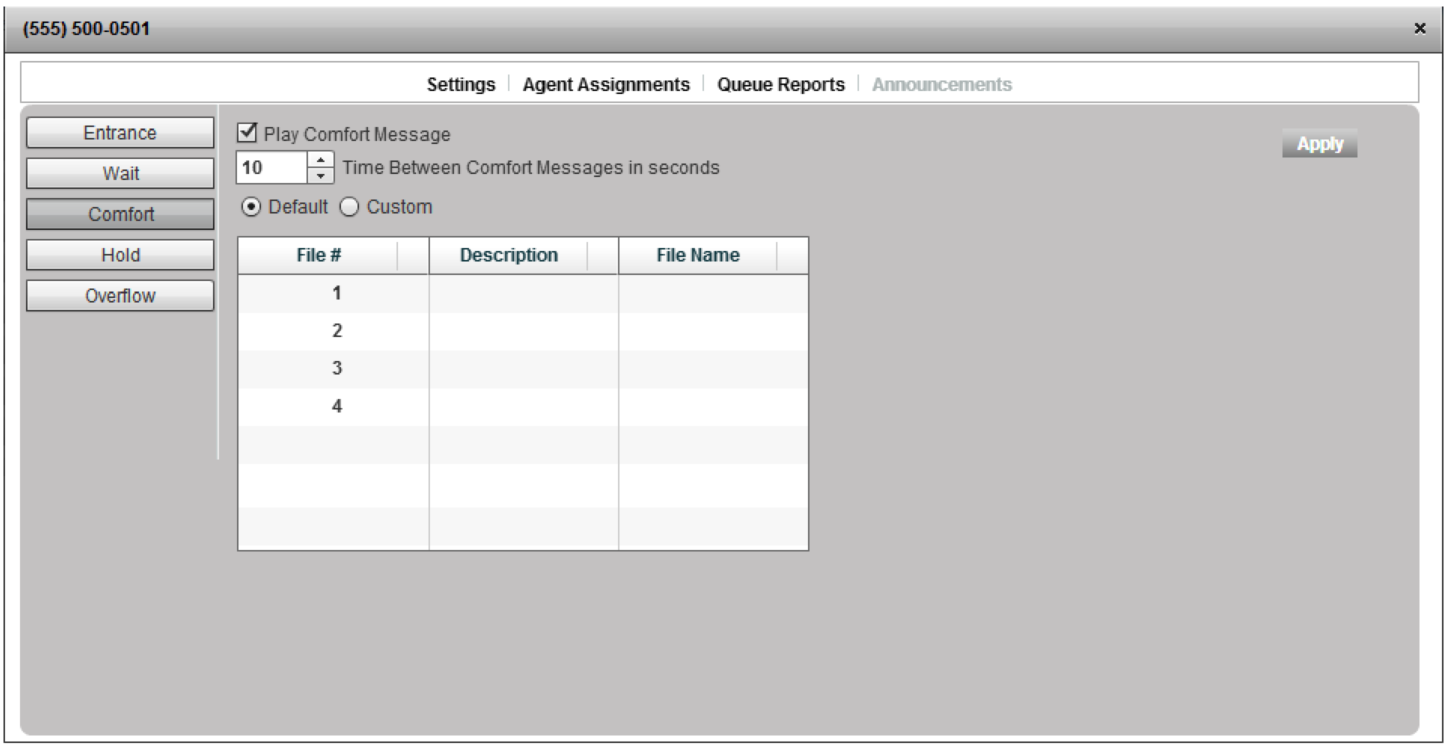


Figure 28: Comfort message selection

Use the entry box to set the period between comfort messages.

You can use the system default message, but this is rare - you will most likely want to record and upload your own. You can upload up to four messages. To do so, select the *Custom* radio button and then the number “1” in the table shown in Figure 28. You will be prompted to select the file. Repeat for messages 2, 3 and 4 if required, and then hit *Apply.*

Hold music and messages

Callers in a queue will typically hear hold music while they are waiting. This is controlled by the *Enable Music on Hold* checkbox, as shown in Figure 29.

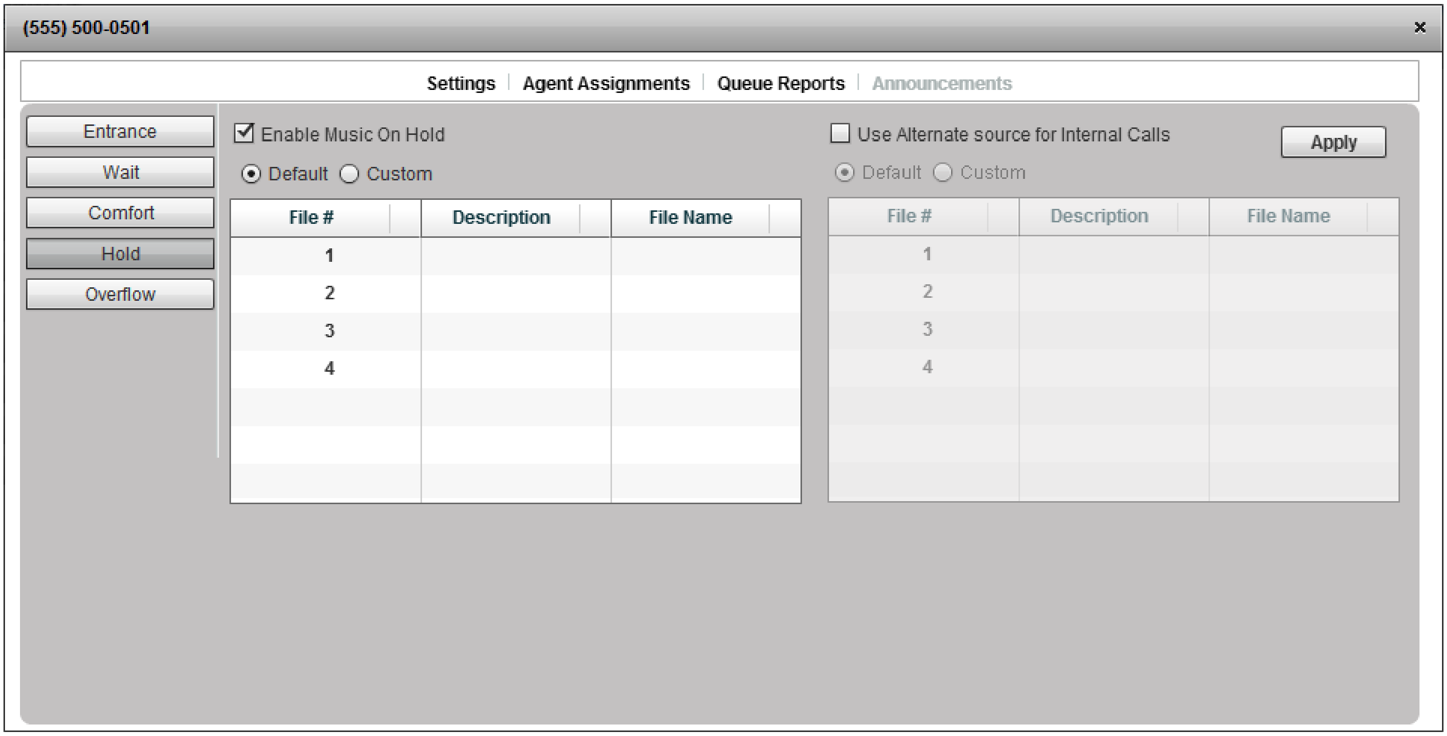


Figure 29: Queue settings for music/messaging-on-hold

You can use the system default music, or upload your own. You can upload up to four pieces of audio. They do not, of course, have to be music, though this is the most common use. To use your own audio, select the *Custom* radio button and then the number “1” in the table shown in Figure 29. You will be prompted to select the file. Repeat for messages 2, 3 and 4 if required, and then hit *Apply.*

You can use different audio for internal callers to the queue. To do so, check the *Use Alternate source for Internal Calls*. Follow the upload process as described above to upload custom audio if desired.

Overflow announcement

Callers will hear the overflow announcement if their call cannot be processed by an agent within a configurable period of time. The configuration interface looks like Figure 30.

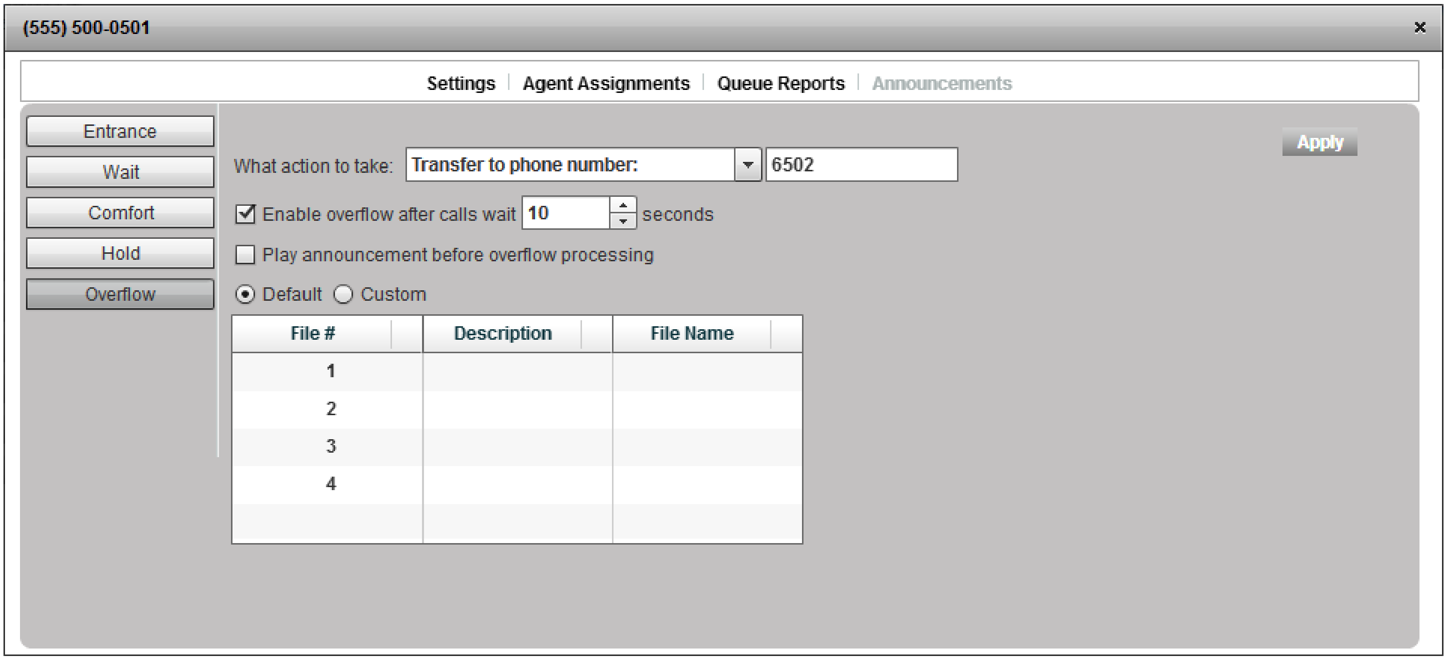


Figure 30: Overflow announcement

The trigger for overflow treatment is the checkbox *Enable overflow after calls wait X seconds*. You can configure values for X according to your site's preference. At this point, you can play and announcement to the caller ("*Play announcement before overflow processing*") and take one of the following actions:

- Play ringing until the caller hangs up
- Perform busy treatment
- Transfer to another phone number

As usual, you can use the system default announcement, or upload up to four of your own. To use your own audio, select the *Custom* radio button and then the number "1" in the table shown in Figure 30. You will be prompted to select the file. Repeat for messages 2, 3 and 4 if required, and then hit *Apply.*

Queue Reports

The Queue Reports panel is not a configuration interface like "Announcements" or "Agent Assignments". Instead, the panel allows you to pull reports and query the status of a queue. Figure 31 shows the interface.

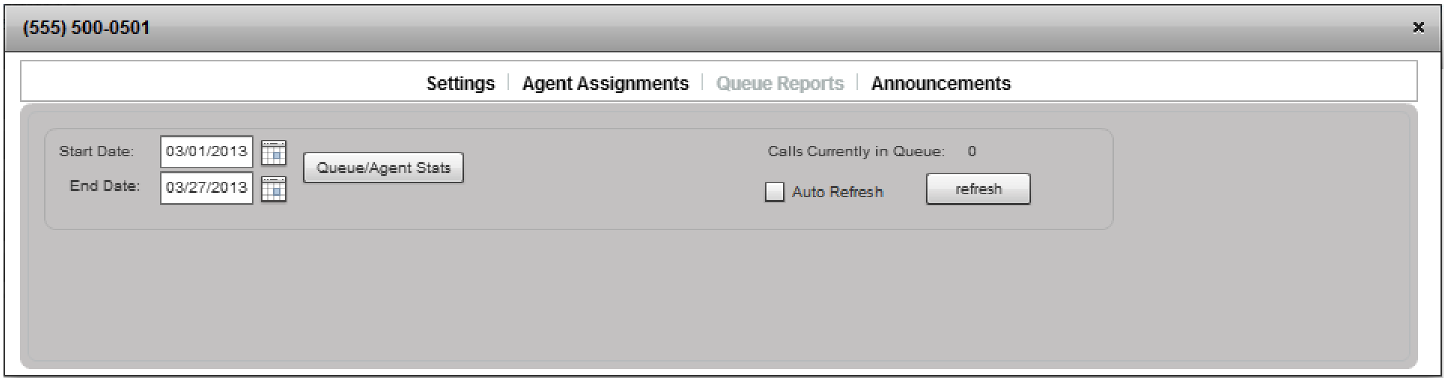


Figure 31: Call Queue Reporting interface

To review the current status of the queue, click the refresh button. The panel will show you the length of the queue.

To pull a report, enter the start and end dates to report in the entry boxes and click the *Queue/Agent Stats* button. A new pane will open, like Figure 32:

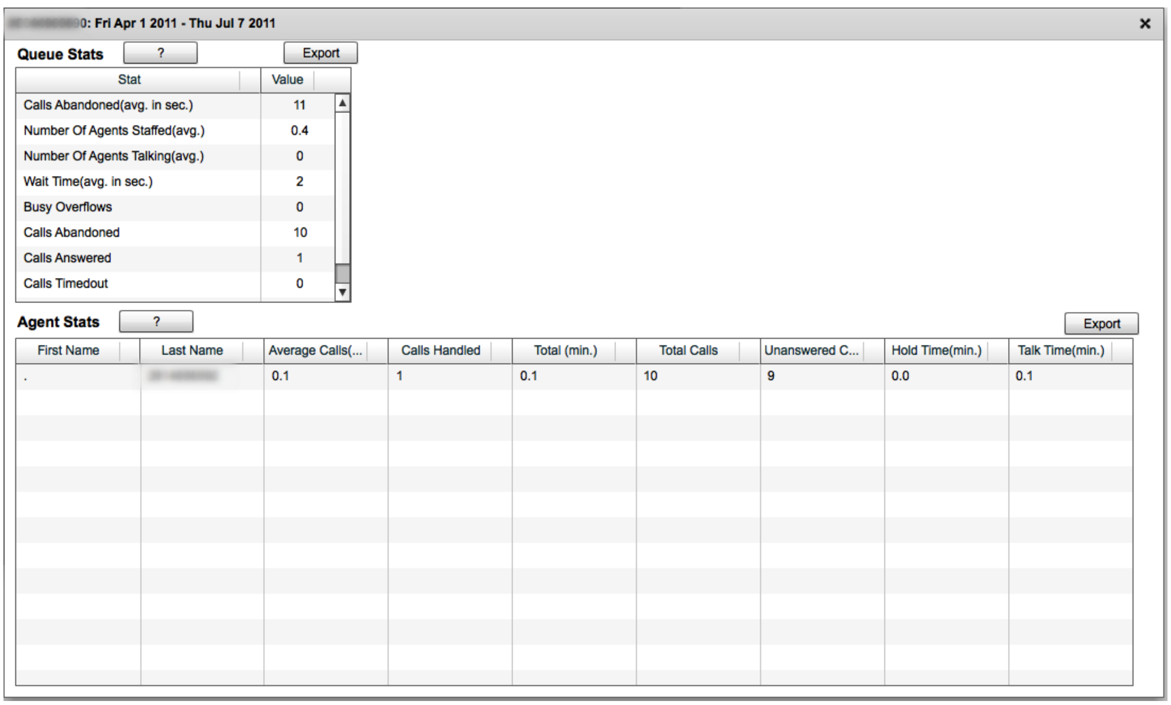


Figure 32: Sample results from queue report

The queue statistics have the following fields:

- **Calls Abandoned (avg.)**. The average time, in seconds, that callers spend waiting for an agent before hanging up
- **Number of Agents Staffed(avg.)**. The average number of agents assigned during the period selected. (Assigned means "configured in the Agent Assignments panel".)

- **Number of Agents Talking(avg.)**. The average number of agents talking (i.e. on the phone, handling queued calls) over the period.
- **Wait time(avg.)**. The average time, in seconds, callers spent waiting for an agent to answer the call.
- **Busy Overflows**. The number of calls that came in after the queue limit was exceeded.
- **Calls Abandoned**. The total number of calls for which the caller hung up.
- **Calls Answered**. The total number of calls answered.
- **Calls Timed Out**. The total number of calls that were unanswered and aged out of the queue (were forwarded upon timeout)
- **Calls transferred**. The total number of calls transferred out of the queue.

The agent statistics have the following fields:

- **Number of calls handled**. The total number of calls that the agent has handled.
- **Average call time**. The average time that an agent spends on each call presented from the queue. *Note:* If they receive a call and then transfer it, only the time spent prior to the transfer is counted.
- **Number of calls unanswered**. The total number of calls that were presented to the agent but were not answered (excepting the case where the agent was already busy). It is possible for this to be greater than the number of calls in cases where a call is presented multiple times to the same agent (e.g. in circular routing).
- **Total talk time**. The amount of time that the agent was busy handling calls.
- **Total staffed time**. The amount of time that the agent was assigned to the queue (in the sense of being allocated in the *Agent Assignments* panel).

You can export agent and queue statistics as CSV-format files by clicking the appropriate *Export* button. You will be prompted for the location to save the results. The CSV file can be imported into another program such as Microsoft Excel.

Call Waiting, Call Forwarding and Business Continuity

You can configure a number of call handling features on behalf of your users using the MySite portal. For example, you might configure Business Continuity on all your user's lines to save them having to do it themselves in the MyPhone portal. Settings you make here are automatically reflected in the user's own settings in MyPhone.

To configure user features, log in to the MySite portal and select the *User Features* tab. Select the feature you wish to configure from the list on the left.

The following features can be configured in this way:

- Call Waiting
- Call Forwarding Always (known as *Forward All Calls* in the MyPhone portal)
- Call Forward No Answer (MyPhone: *Forward Unanswered Calls*)
- Business Continuity.

Call Waiting

Call waiting allows a call to be delivered to the user's phone even if they are already in another call (as opposed to treating the caller as if the recipient was busy, and sending their call to voicemail or giving them some other busy treatment).

To enable call waiting for a user, select the *Call Waiting* button on the left of the *User Features* panel. Check the box in the Call Waiting column for each user that requires the feature.

Click on a checkbox to enable call waiting for that phone number.

First Name	Last Name	Phone Numbers	Extension	Call Waiting
Jack504	5716122848	(571) 612-2848	2848	☑
XO Product	Training 5716122846	(571) 612-2846	2846	☑
Conference	5716122844	(571) 612-2844	2844	☑
Jack1	5716122843	(571) 612-2843	2843	☑
Jack335	5716122842	(571) 612-2842	2842	☑
Jack3	5716122845	(571) 612-2845	2845	☑
Steve	Carter	(571) 612-2857	2857	☑
Joe	Smith	(571) 612-2856	2856	☑
Jack500	5716122858	(571) 612-2858	2858	☐
Jack	5716122860	(571) 612-2860	2860	☐
Bud Gibson	5716122854	(571) 612-2854	2854	☑
Shriman	Gurung	(571) 612-2847	2847	☑
XO Product	Training 5716122859	(571) 612-2859	2859	☑

Figure 33: Call Waiting configuration

Call Forwarding Always

The Call Forwarding Always interface looks like Figure 34.

Click on an entry to view/configure Call Forward Always for that phone number.

First Name	Last Name	Phone Number	Extension	Forward Always To
Jack504	5716122848	(571) 612-2848	2848	Not yet forwarded to a number.
XO Product	Training 5716122846	(571) 612-2846	2846	Not yet forwarded to a number.
Conference	5716122844	(571) 612-2844	2844	Not yet forwarded to a number.
Jack1	5716122843	(571) 612-2843	2843	Not yet forwarded to a number.
Jack335	5716122842	(571) 612-2842	2842	Not yet forwarded to a number.
Jack3	5716122845	(571) 612-2845	2845	Not yet forwarded to a number.
Steve	Carter	(571) 612-2857	2857	Not yet forwarded to a number.
Joe	Smith	(571) 612-2856	2856	Not yet forwarded to a number.
Jack500	5716122858	(571) 612-2858	2858	Not yet forwarded to a number.
Jack	5716122860	(571) 612-2860	2860	Not yet forwarded to a number.
Bud Gibson	5716122854	(571) 612-2854	2854	Not yet forwarded to a number.
Shriman	Gurung	(571) 612-2847	2847	Not yet forwarded to a number.
XO Product	Training 5716122859	(571) 612-2859	2859	Not yet forwarded to a number.

Figure 34: Configuring Call Forwarding Always

To configure forwarding for a line, select it, choose the forwarding action (forward to an internal number, an external number or do not forward) from the drop down. Then, enter or select the number to forward to before clicking *Save*.

- Choosing the *Ring reminder* radio button will play a short tone on the user's phone when a call arrives and is forwarded.

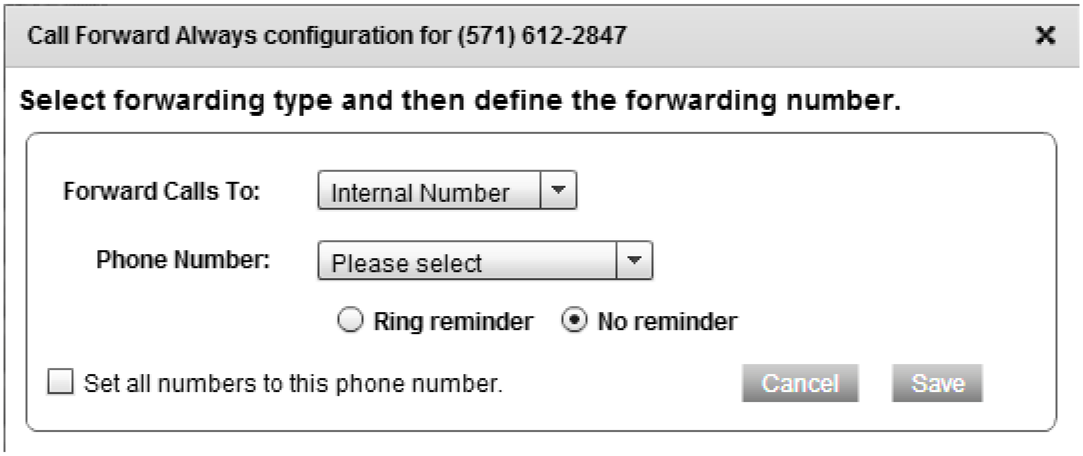


Figure 35: Call Forwarding Always selection screen

Call Forward No Answer

Call Forward No Answer allows unanswered calls to be forwarded to a destination other than voicemail if the call goes unanswered for a number of rings. If you want your users to have unanswered calls roll to voicemail, you do not need to use this feature.

Configuring Call Forward No Answer is very similar to configuring Call Forwarding Always. Select Call Forward No Answer from the left hand pane in the User Features tab and select the user in question. The interface will look like .

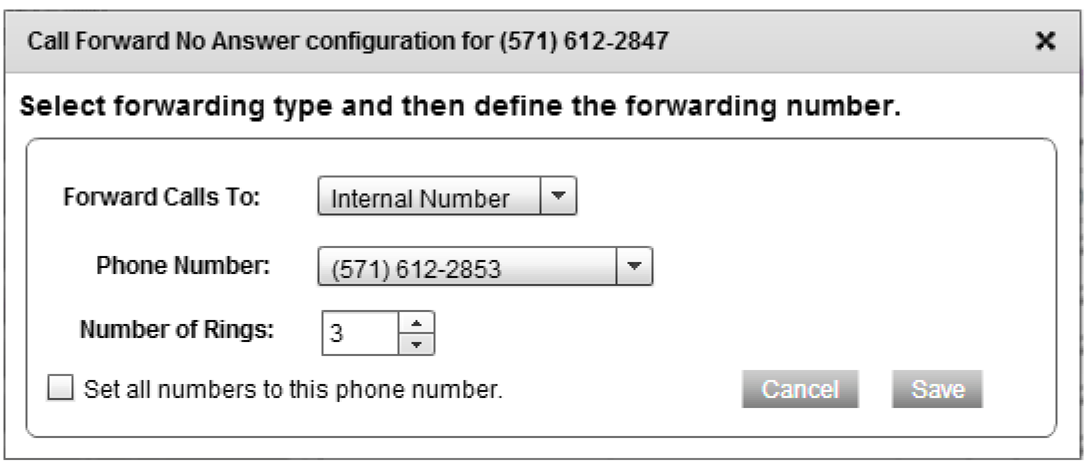


Figure 36: Call Forward No Answer settings

Select the action required, the number of rings that should pass before the call is forwarded, and click *Save*.

Call Forward Busy

Call Forward Busy allows calls for users who are already in a call to be forwarded to a destination other than voicemail. If you want your users to have these calls roll to voicemail, you do not need to use this feature.

Configuring Call Forward Busy is very similar to configuring Call Forwarding Always. Select *Call Forward Busy* from the left hand pane in the *User Features* tab and select the user in question. The interface will look like Figure 37.

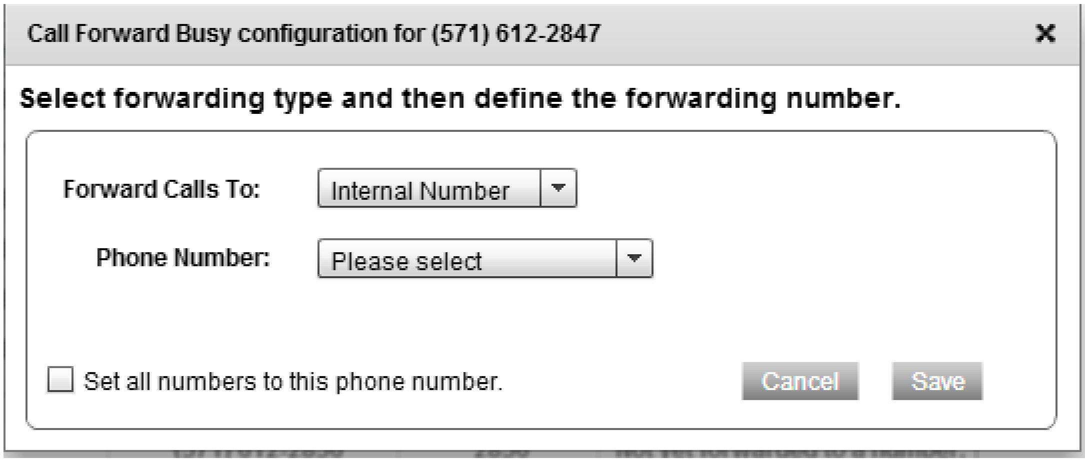


Figure 37: Call Forwarding Busy settings

Select the action required and click *Save*.

Business Continuity

This feature allows the Hosted PBX platform to forward calls when your users' phones or site are unavailable. For example, if your site were to experience a power outage, the Business Continuity feature would allow calls to your business to be sent to a backup location such as another branch office within your organization.

To configure Business Continuity, select the feature from the list on the left of the *User Features* panel. Select the user in question and select and enter the target number to forward to. This is typically an external number, to guard against site failure. Remember to click *Save* when you are done. Figure 38 shows an example.

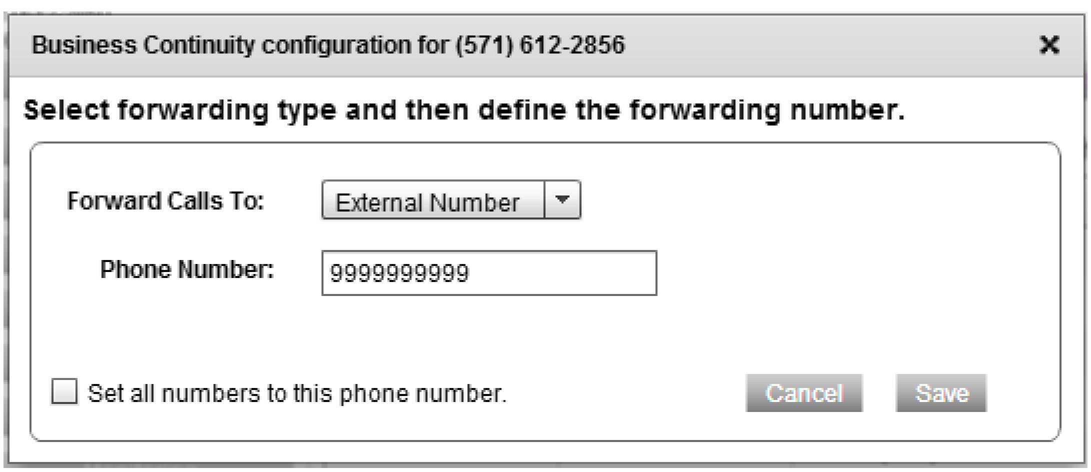


Figure 38: Business Continuity settings

Voicemail and Unified Messaging

You can use the MySite portal to configure the voicemail and unified messaging features of your service on behalf of your users. For example, you might decide for forward a copy of each user's voicemails to their email accounts to save them having to do it themselves in the MyPhone portal. Settings you make here are automatically reflected in the user's own settings in MyPhone.

To configure messaging, log in to the MySite portal and select the *User Features* tab. Select *Unified Messaging* from the list on the left, and select the user whose messaging service you wish to configure.

A new window will open where you can configure the settings for the user's voicemail and unified messaging service. Figure 39 shows the interface.

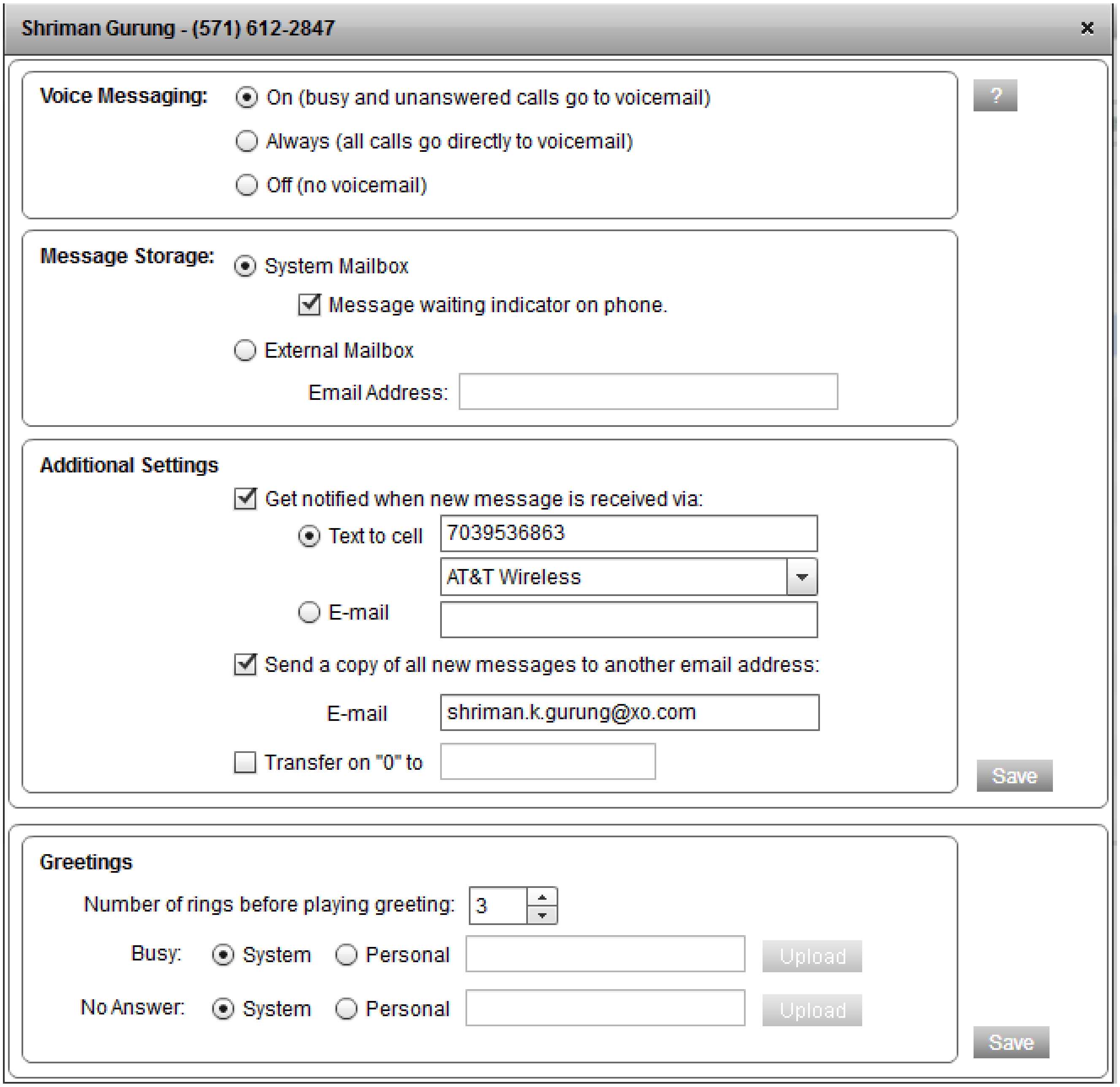


Figure 39: Unified Messaging interface screen

- Use the **Voice Messaging** radio buttons to choose whether to send all calls to voicemail, no calls, or only unanswered ones. Most users select the last choice, but if your users are part of a call center, for example, you do not want calls delivered to their line to end up in voicemail, so you would disable it.
- **Message storage** defines where voicemails are stored. XO Hosted PBX includes an easy to use voicemail system, but you can alternatively choose to forward a user's voicemails to another system as email attachments.
- You can combine the two, by choosing the *System Mailbox* checkbox and the **Additional Settings** choices to notify the user of new message arrival – or even send them a copy of the message – to their email account or cell phone.

- You may want to set a **Greeting** that callers will hear when users are busy or unavailable. Typically users set this up themselves, either using the phone of via the MyPhone portal, but you can upload a pre-recorded greeting if you prefer, or use system-defined ones.

Tip: If you want to give callers more choice, use the *Transfer on "0"* option in the **Additional Settings** section to set a number that they will be transferred to if they hit 0, then ask users to refer to that choice in their greeting. For example, "Hi, this is Jane in Sales. Please leave a message at the tone, or press 0 to be transferred to the account desk."

- Click *Save* to save your changes.

Outbound Call Control

Many enterprises have policies restricting who may make certain types of outbound calls. In the MySite portal, the *Outbound Calling* panel in the *Site Services* tab allows you to restrict the following types of outbound calling:

- International
- Operator assistance
- Chargeable directory assistance.

The interface panel looks like Figure 40.

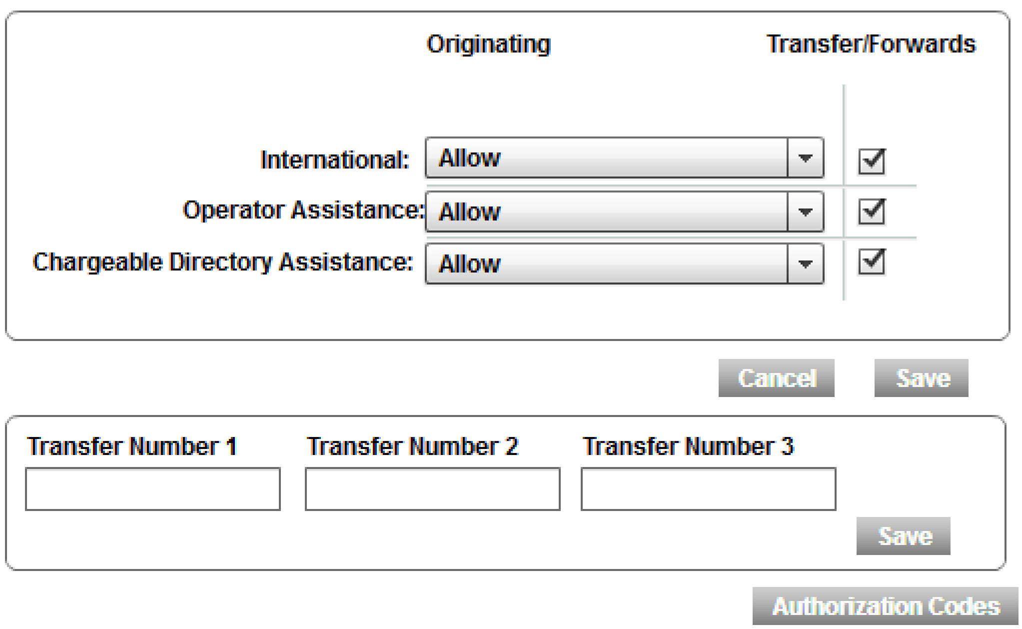


Figure 40: Outbound Calling interface panel

For each call type, you can choose to

- Permit
- Deny

- Automatically transfer the call

  You can define up to three numbers to use for transfers, using the *Transfer Numbers* boxes. Remember to click *Save* after entering a number.

- Require an authorization code to be entered by the user before the call is placed.

  To define an authorization code, click the Authorization Codes button and enter a code. When users place a call, they will hear a message asking for their code after dialing the "restricted" number. If they enter the right code, the call will go through.

Customized settings for outbound calling

You may have users for whom the site-wide outbound calling restrictions are not appropriate. In this case, you can override them on a per-user basis by selecting *Outbound Calling* from the *User Features* tab (and not Site Services). Select the user and check the Custom check-box.

For each call type that you want to permit, check the appropriate box. Figure 41 shows an example.

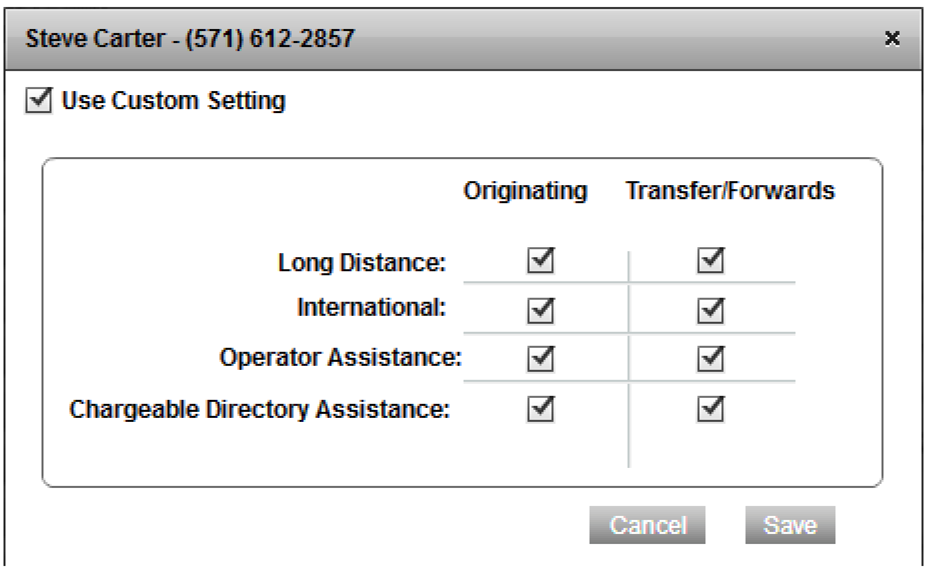


Figure 41: Per-user outbound calling settings

Push-To-Talk, Group Paging and Instant Group Call

Push-to-Talk

Push-To-Talk allows a user to talk to another without dialing their full number and waiting for the other party to pick up. A user dials a star code (`*50`), then the user's extension. The receiving user's phone automatically answers and the users can communicate. Speech can either be one-way or two-way depending on configuration.

In practice, this is a rarely used feature because it is already very easy to dial another user via their extension. However, some sites require it to mimic function on legacy PBXes.

To configure the feature, select *Push To Talk* from the list in the *User Features* panel. Select a line to configure it. The interface will change to look like Figure 42:

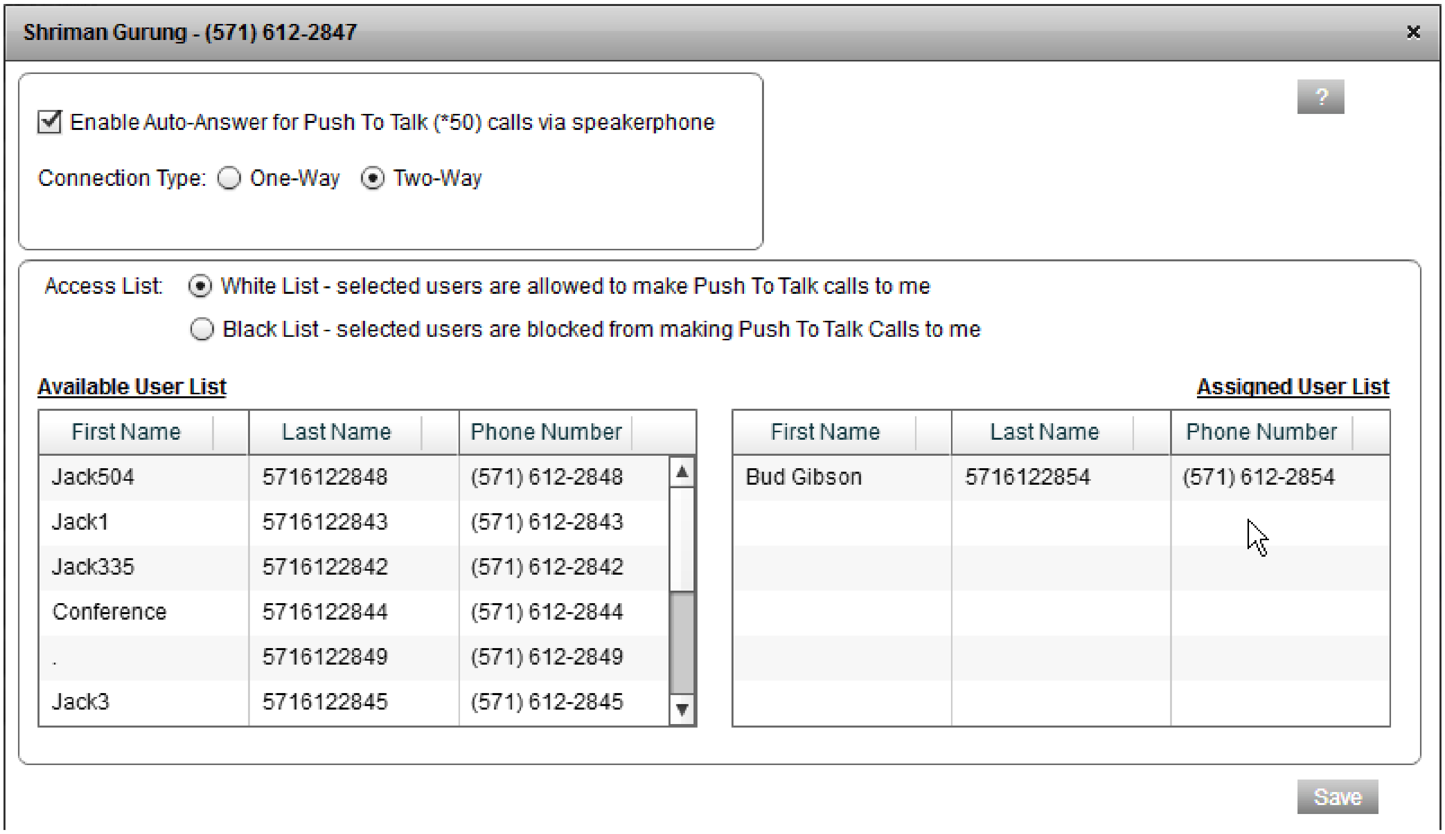


Figure 42: Push to talk configuration for a line

- If you want your user's phone to automatically answer the push-to-talk calls that it receives, check the *Enable Auto-Answer for Push To Talk (*50) calls via speakerphone* box.
- Select the *Connection Type* radio button according to whether calls to this line should be one-way (the user can hear the voice of the invoking user, but not the reverse) or two-way (both invoker and user can hear one another).

In the lower part of the interface panel, you must select who is allowed to make push to talk calls to this user. There are two modes. You can either allow only a named set of users to make such calls ("*White List*"), or the inverse: anyone can make calls save for selected users ("*Black List*"). Select the appropriate radio button.

Now, drag the users from the *Available User List* to the *Assigned User List* in the panel and click *Save*. In the example shown in Figure 42, the administrator has selected *White List* and assigned user "Bud": this combination means that only "Bud" may make push to talk calls to this user.

Group Paging

Group paging lets a user call a number to broadcast their voice to a group of phones. When the user dials the number, the Hosted PBX system alerts all the recipients, then transmits the user's speech until the latter hangs up.

Group Paging is not enabled by default. If you do not see the option listed in the *Site Services* panel, contact your XO representative to enable it and assign a phone number to the feature.

Group Paging significantly increases the amount of call traffic on your network. Before using it, make sure that your network has adequate bandwidth and quality of service to support this feature. Your XO representative can assist you.

To configure the feature, select *Group Paging* from the list on the left of the *Site Services* panel. The interface will look similar to Figure 43.

Group Paging

Active	Name	Phone Number	Extension	
☑	9721003006	(972) 100-3006	3006	

Figure 43: Group Paging line selection screen interface

Select the group paging number to configure by clicking on the green “pencil” icon. You will see the configuration settings for the line, similar to Figure 44:

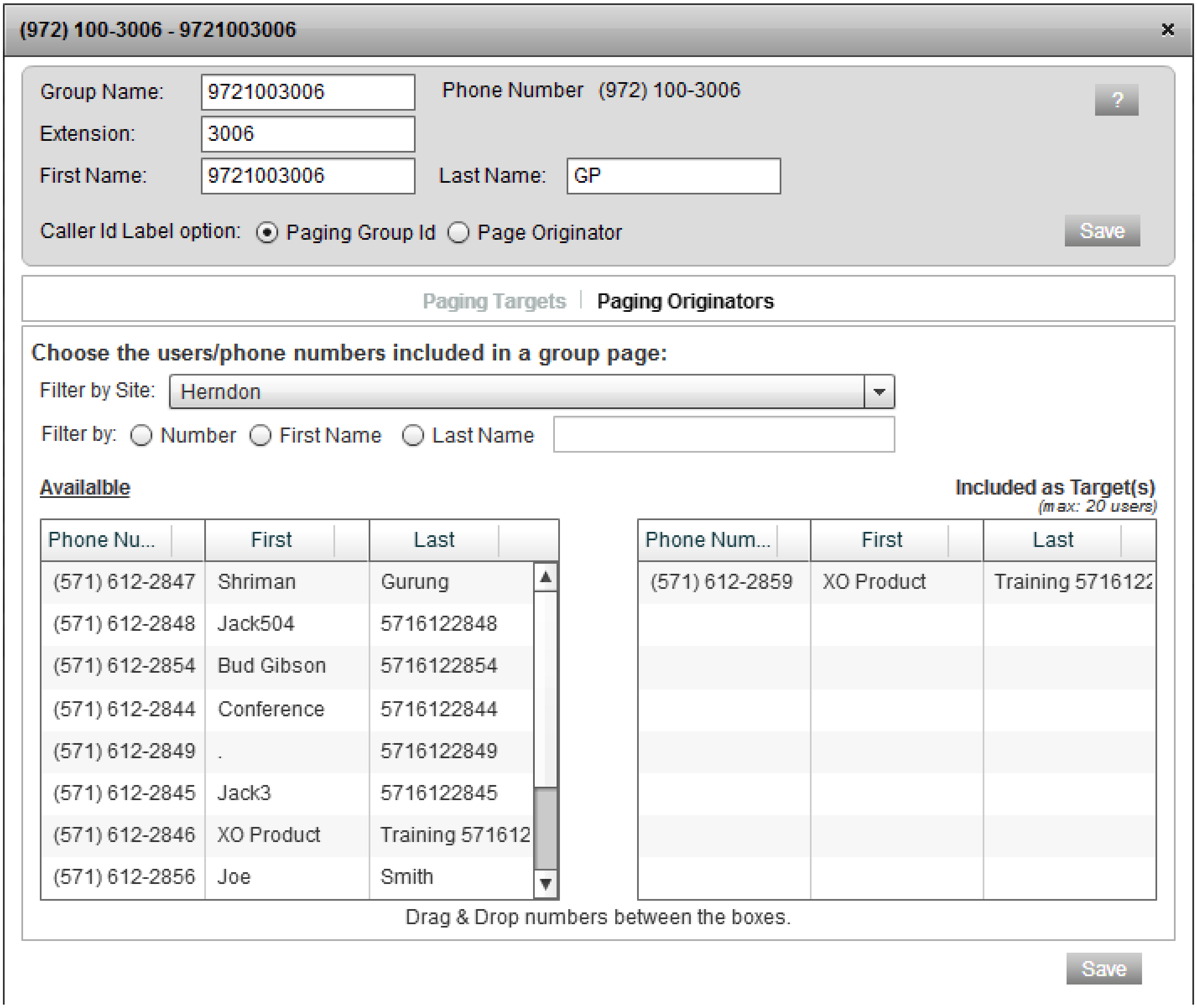


Figure 44: Group Paging configuration for a line

At the top of the panel are the basic settings: the group paging number, an extension number, identifying name and caller ID options:

- Extension number can be any value of your choosing, subject to the following rules:
    - The extension must not already be in use
    - Extensions can be between 2 and 6 digits long (the default is four – the last four digits of the phone number)
    - Extensions with these values will not be allowed:
        - 0311, 1311
        - 0911, 1911, 911
        - 00, 011
        - N11, N11X, N11XX, N11XXX, where N is 2-9 and X is 0-9. For example, 311 and 411 would be barred.
- The radio buttons let you choose what appears on the caller ID when a user invokes a group paging call. By default the recipients of the page will see the caller ID of the user invoking the call (*Page Originator*), but you can create a new ID by filling in the *First Name* and *Last Name* boxes and selecting the *Paging Group Id* button if you prefer.

Click the *Save* button in the upper part of the panel when you are done.

Next, you must define who will be contacted when a user invokes a group paging call. Click on the *Paging Targets* label and drag numbers from the left hand, *Available*, column, to the right hand, *Included as Target(s)*, one. You can include up to 20 recipients. Click *Save* in the lower right hand corner when you are done.

Finally, select who will be allowed to initiate group pages. Click on the *Paging Originators* label and drag and drop user(s) from the left hand *Available* panel to the right *Included As Originator(s)* one, and click *Save*.

Instant Group Call

Instant group call lets a user call a number that provides a group of members with an instant conference bridge. When the user dials the specific group call number, the Hosted PBX system alerts all members in the group and, as the members answer, they are joined into a multi-way conference.

Instant Group Call significantly increases the amount of call traffic on your network. Before using it, make sure that your network has adequate bandwidth and quality of service to support this feature. Your XO representative can assist you.

To configure the feature, select *Instant Group Call* from the list on the left of the *Site Services* panel in MySite. If this is your first call group, select the number to use for the service from the drop down list and choose *Add*. The interface looks like Figure 45.

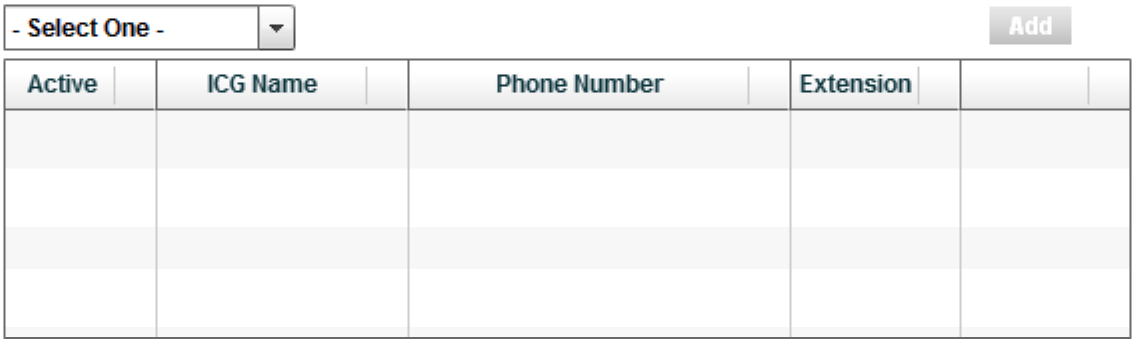


Figure 45: Interface for adding an Instant Group Call number

After clicking *Add*, or if you are editing an existing group call number, select the entry and click on the green pencil icon, as in Figure 46.

Figure 46: Selecting an existing Instant Group Call number

A new window will appear. Typically you will assign an extension to the group call number, to make it quicker to dial. You must also select the users whose phones will ring when the group call number is dialed. Drag and drop the numbers you want from the left hand column to the right, as shown in Figure 47. Click *Save* to save your changes.

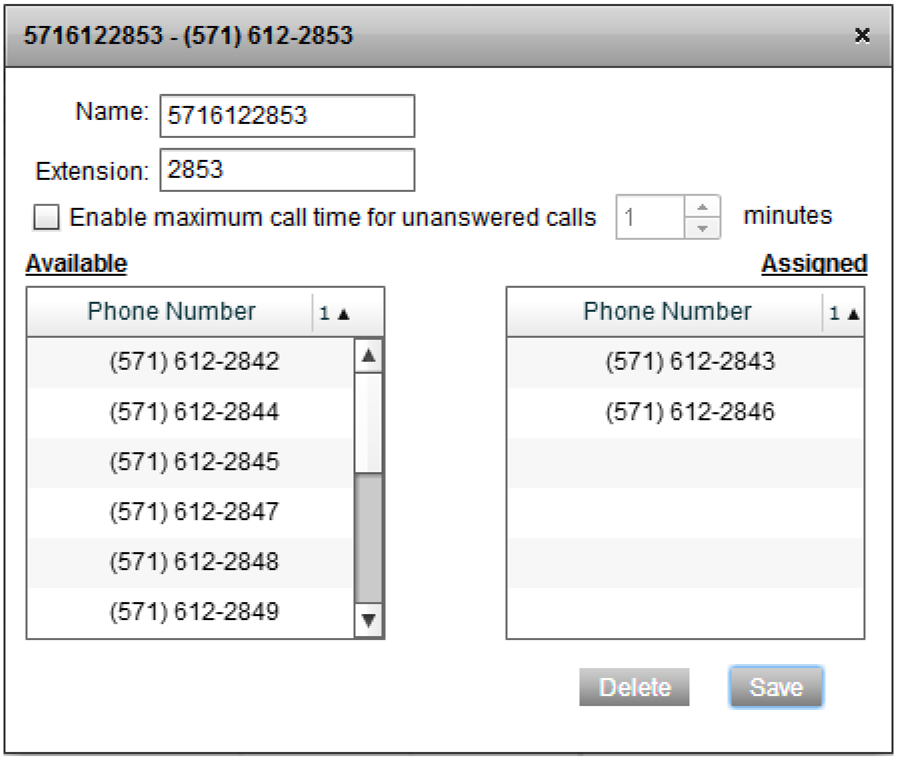


Figure 47: Assigning numbers to an instant call group

Line Sharing and Monitoring

Sharing

Sharing is a technology that allows one phone number to "appear" on multiple phones at the same time. It is commonly used in environments where one user needs to receive and make calls on behalf of a co-worker or manager.

Sharing is different from sim-ring, another technology offered by XO Hosted PBX, in that in sharing, one number is "split" across two phones but one phone number, whereas in sim-ring, one call is split over multiple phone numbers.

To share a user's line across other users' phones, select Sharing from the User Features panel. Click on the user whose line you want to share, and select the user (or users) that will receive this line. In the example shown in Figure 48, the line ending in 2847 is being shared with a user, "Steve".

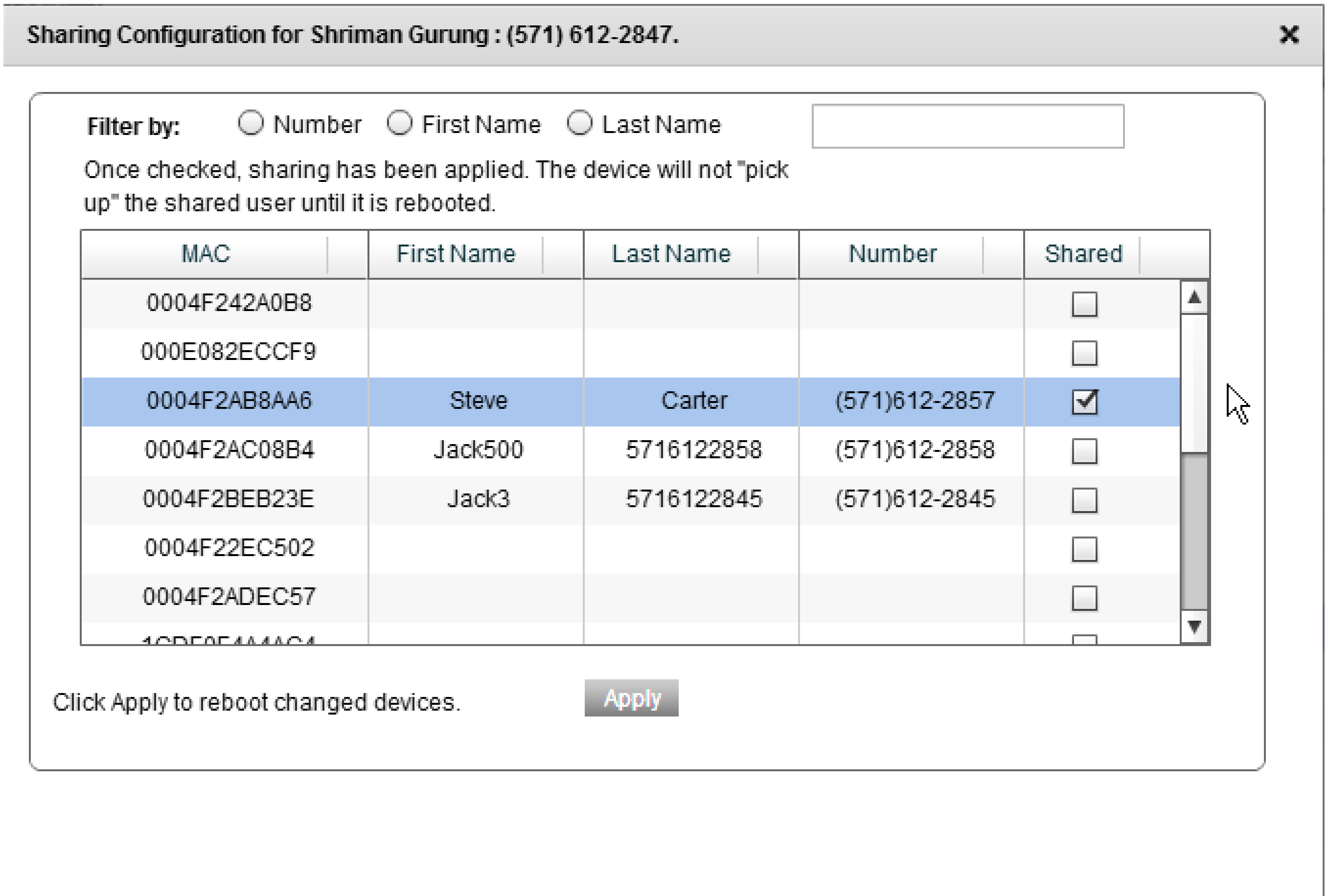


Figure 48: Selecting a user with which to share a line

Click *Apply* to save your changes. You must now reboot the (receiving) user's phone ("Steve" in the example of Figure 48.) for the changes to take effect on their phone.

A line may be shared across up to 35 phones. However, best practice is that you minimize the amount of sharing, in order to make best use of your network's bandwidth. Sharing a line across 2-4 other users is typical.

Monitoring

The Monitoring feature lets a user see who is busy on a call and who is free. It is commonly used by receptionists and other front-desk employees to help them handle incoming calls quickly and efficiently. (In XO Hosted PBX, monitoring is supported on both Polycom and Cisco phones, but the latter require a sidecar.)

Some users may not want their phones to be monitored. This preference can be set in the Privacy panel, as described in the section titled Privacy in this document.

To configure the feature, select *Monitoring* from the list on the left of the *User Features* panel. Select the user that will be monitoring other user(s). The interface panel will look like Figure 49.

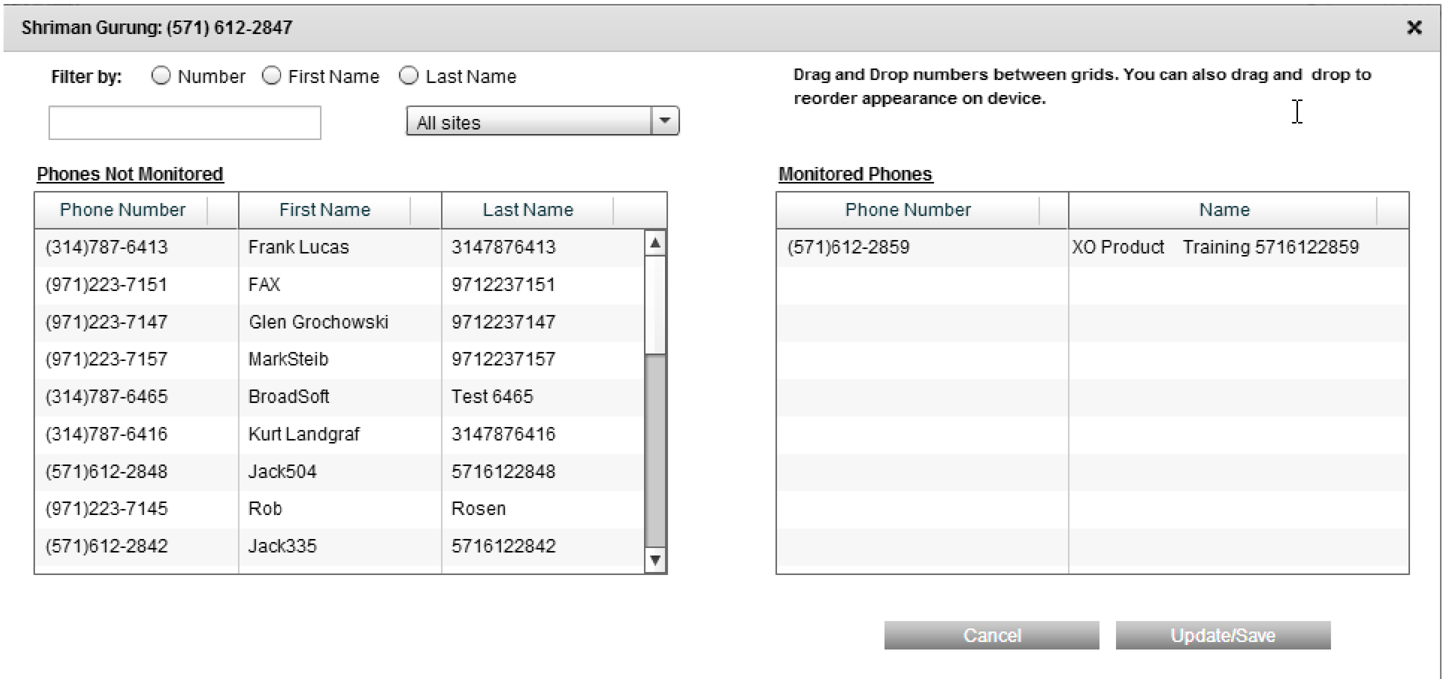


Figure 49: Configuration pane for Monitoring

Next, drag the lines to be monitored from the *Phones Not Monitored* list on the left of the panel to the *Monitored Phones* on the right, and click *Update/Save*.

Privacy

The Privacy feature allows you to exclude users from being

- listed in the Auto-Attendant dial-by-name and dial-by-extension directory
- monitored by other users in your site.

To configure privacy, select Privacy from the User Features panel and click the name of the user that you want to configure. After you select the line, you will see a screen like Figure 50:

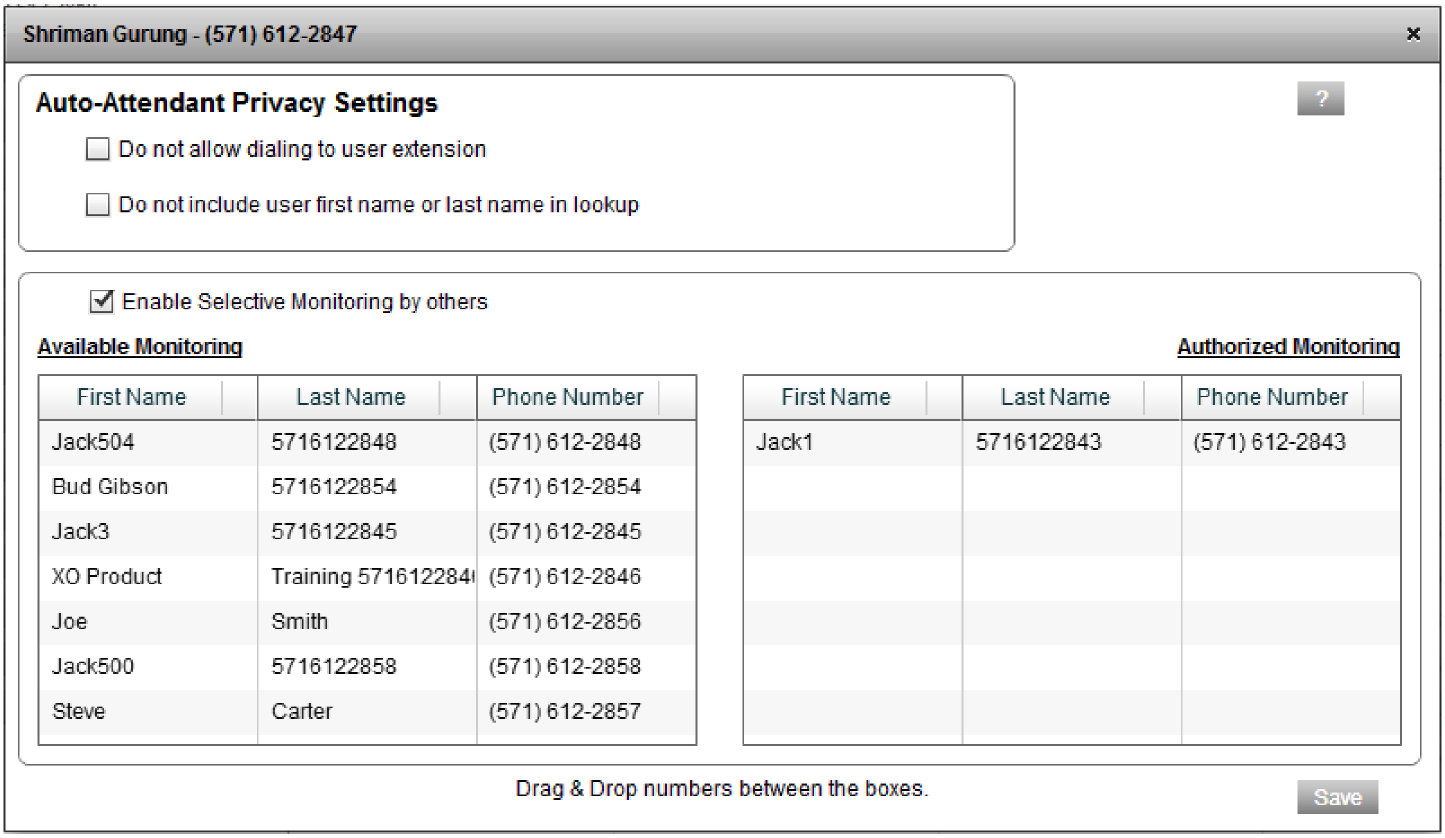


Figure 50: Configuring Privacy

- To hide a user from the auto-attendant directories, uncheck the *Do not allow dialing to user extension* and *Do not include user first or last name in lookup* boxes. The first prevents the user from being found by dialing their extension and the second prevents dial-by-name.
- To restrict who may monitor this user, check the *Enable Selective Monitoring by others* box and drag and drop users from the *Available Monitoring* list on the left to the *Authorized Monitoring* list on the right.
- Click *Save* when you are done.

Call Reporting

The MySite portal lets you easily pull reports on your organization's phone calls, whether they are

- in-bound, out-bound or missed calls
- calls to an incoming call queue (call center)
- calls to an auto-attendant

We'll look at each of these in turn.

General Call Reporting: inbound, outbound and missed calls

The Call History panel provides an easy-to-use interface for reporting on calls. To access it, select the *Call History* label in the upper part of the screen:

The interface panel looks like Figure 51.

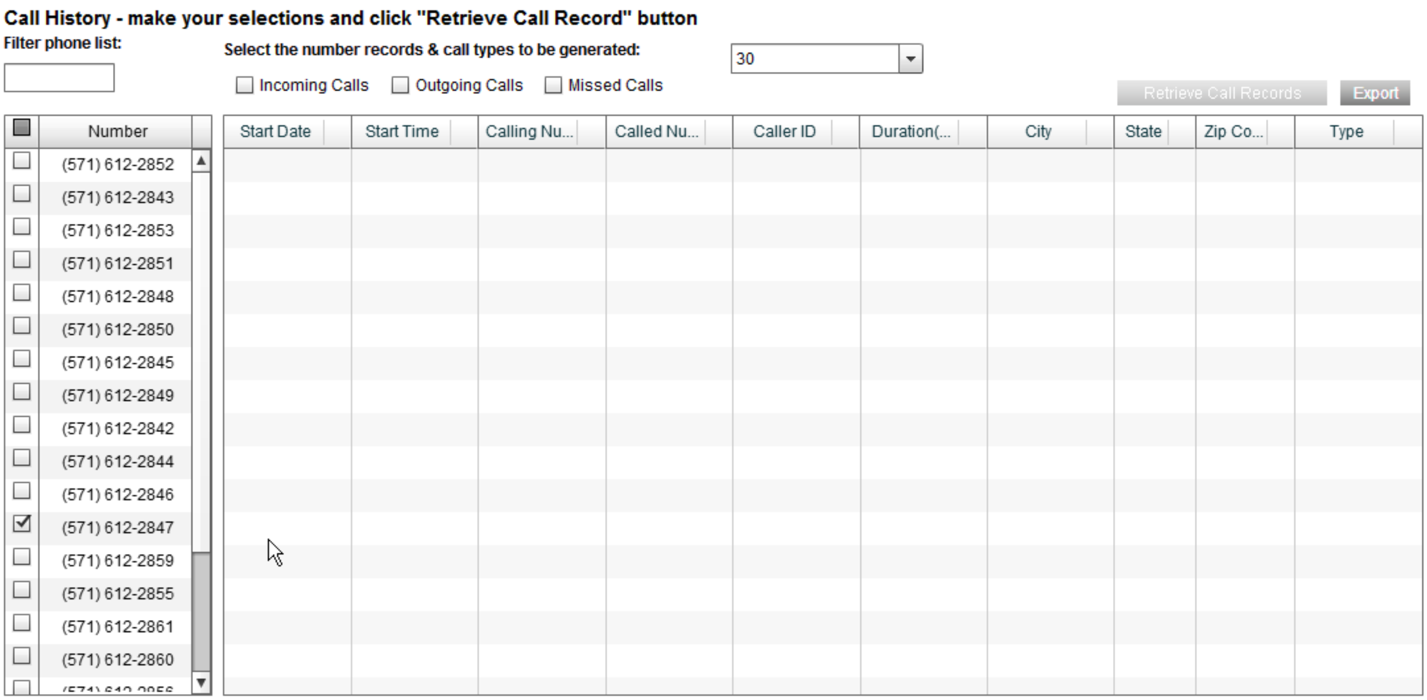


Figure 51: Call History panel interface

Fill out the options in the panel and click the *Retrieve Call Records* button to show the results. For example:

- Check one or more of the *Incoming Calls*, *Outgoing Calls* or *Missed Calls* check boxes
- Select the number of days' worth of records to retrieve using the "*Select the number records*" drop-down list. The default is 30. (Selecting 30 is a best practice: you can choose longer values, but these searches take much longer to complete.)
- If you are only interested in a subset of your users, you can use the checkboxes next to each number to include those and exclude the others. Checking the box at the top of the column next to the word "Number" will select/deselect all the numbers.

After you click the *Retrieve Call Records* button, the results are retrieved and displayed. This takes a few (~30) seconds, with the results looking similar to Figure 52:

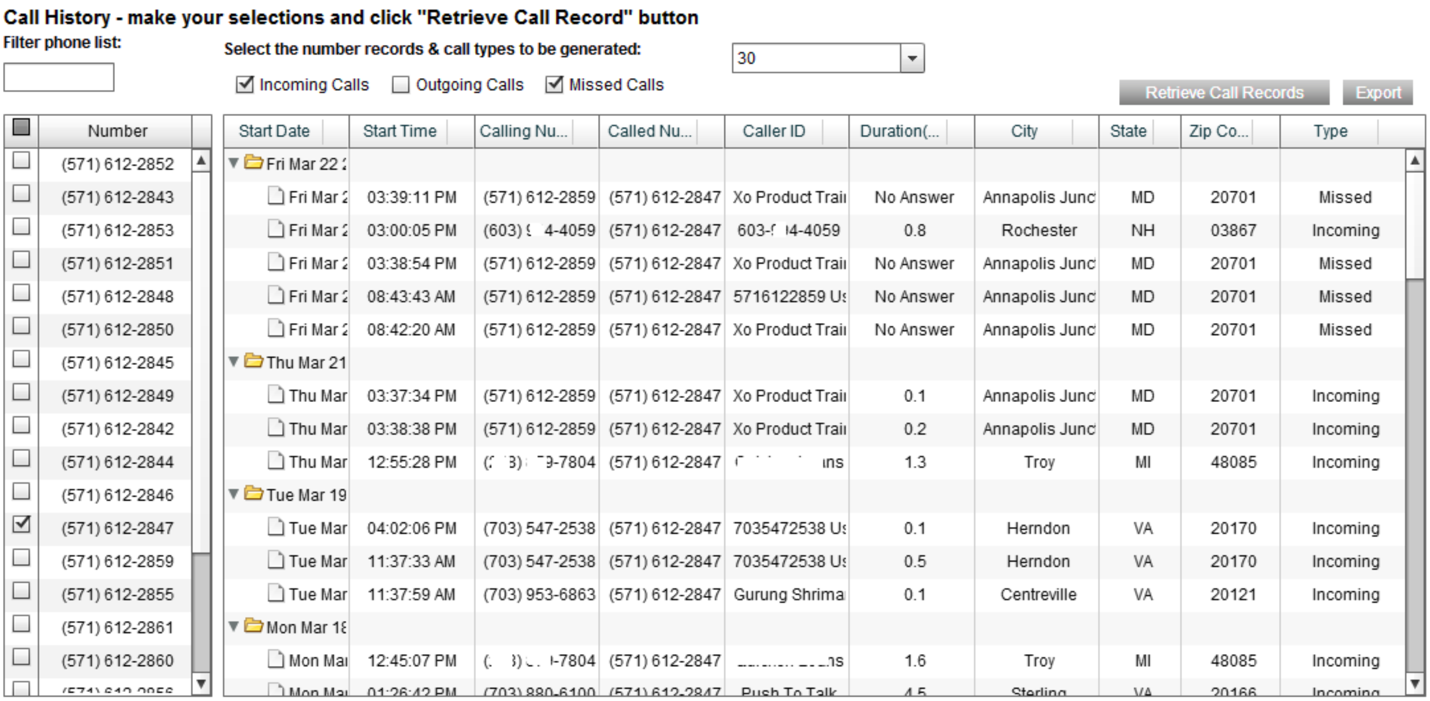


Figure 52: Sample Call Records report

You can export the results as a CSV-format file by clicking the *Export* button. You will be prompted for the location to save the results. The CSV file can be imported into another program such as Microsoft Excel.

Reporting on Auto-Attendants

If your site includes an auto-attendant, you will want to understand how it is being used. For example, "how many callers picked option 1 over option 2 last month?" To answer these types of questions, click on the *My Reports* pane at the very top right of the screen, as illustrated in Figure 53.

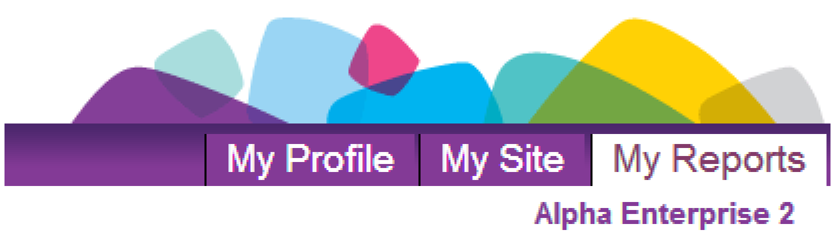


Figure 53: My Reports tab

In the panel that appears, you need to select the site and auto-attendant to report on, and the range of dates that the report should cover.

Select the site that you want to report on by checking the box next to its name in the *Site Name* column. Then, click *Get Auto Attendant List*. The left hand panel of the interface will show you all the attendants for that site. Select the auto-attendant you wish to report on by checking the box next to its phone number. In the illustration shown in Figure 54, the administrator has selected Herndon and one auto-attendant, "Main Line Auto Attendant", has been retrieved.

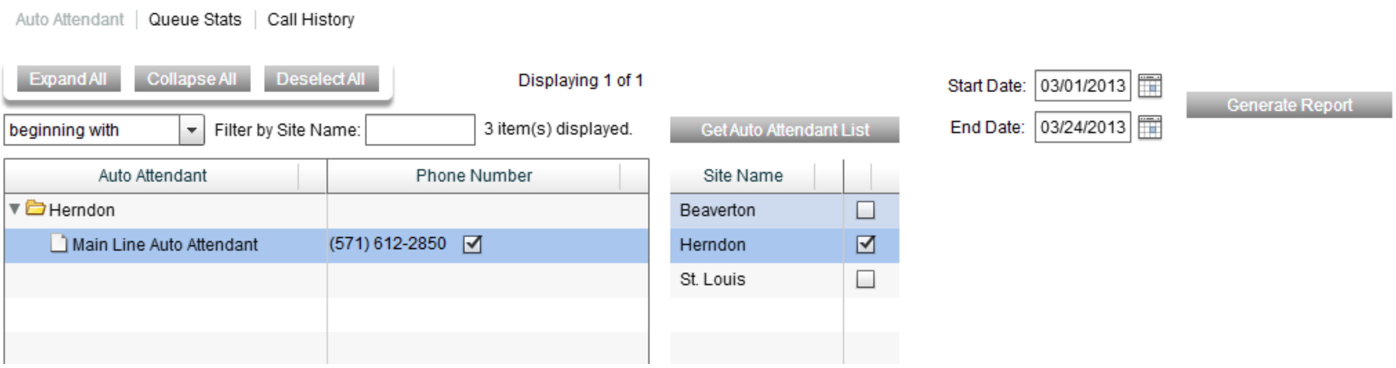
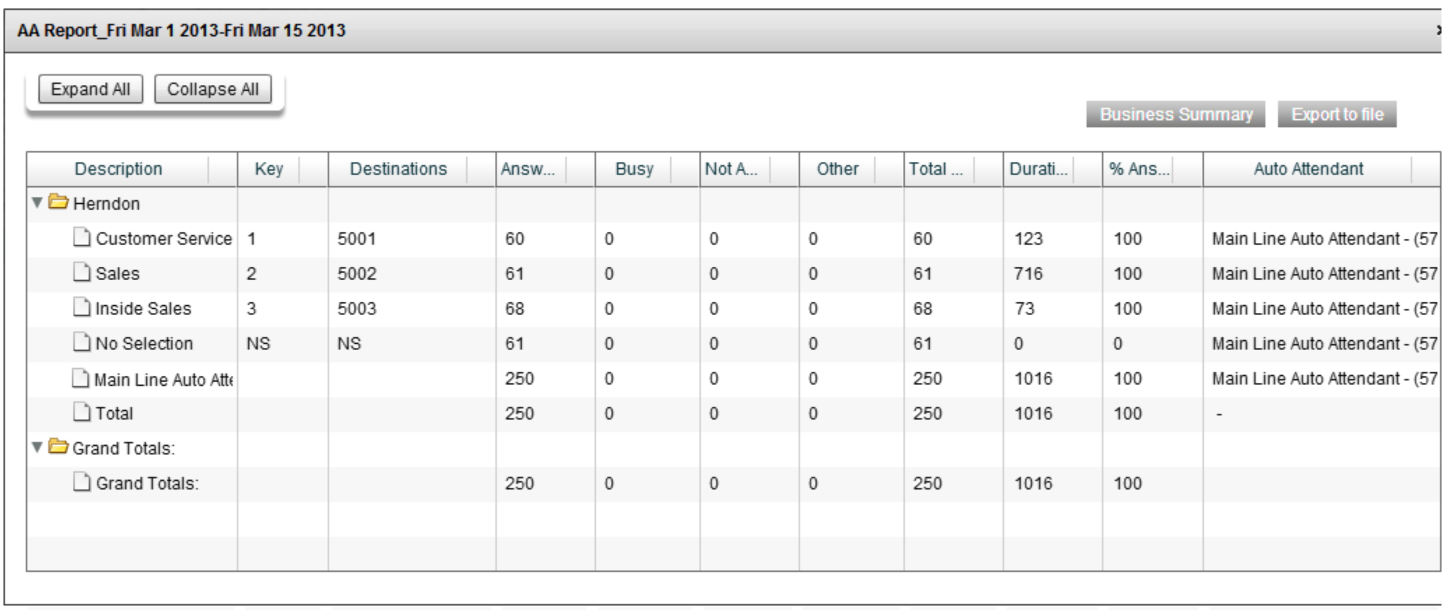


Figure 54: Selection panel for auto-attendant

Using the date entry boxes on the right of the interface, enter the start and end dates in `mm/dd/yyyy` format, then click *Generate Report*. The results will be retrieved and displayed, as in Figure 55.


AA Report_Fri Mar 1 2013-Fri Mar 15 2013

Expand All | Collapse All | Business Summary | Export to file

Description	Key	Destinations	Answ...	Busy	Not A...	Other	Total ...	Durati...	% Ans...	Auto Attendant
Herndon										
Customer Service	1	5001	60	0	0	0	60	123	100	Main Line Auto Attendant - (57
Sales	2	5002	61	0	0	0	61	716	100	Main Line Auto Attendant - (57
Inside Sales	3	5003	68	0	0	0	68	73	100	Main Line Auto Attendant - (57
No Selection	NS	NS	61	0	0	0	61	0	0	Main Line Auto Attendant - (57
Main Line Auto Atte			250	0	0	0	250	1016	100	Main Line Auto Attendant - (57
Total			250	0	0	0	250	1016	100	-
Grand Totals:										
Grand Totals:			250	0	0	0	250	1016	100	

Figure 55: Auto-attendant report

You can export this data in CSV format by clicking the *Business Summary* or *Export to file* buttons. The former summarizes the report whereas the latter provides exactly the same level of detail as shown on screen. Figure 56 shows an example after *Export to file* has been used.

	A	B	C	D	E	F	G	H	I	J	K
1	Description	Key	Destinations	Answered	Busy	Not Answered	Other	Total Calls	Duration(min)	% Answered	Auto Attendant
2											
3	Customer Service	1	5001	60	0	0	0	60	123	100	Main Line Auto Attendant - (571) 612 2850
4	Sales	2	5002	61	0	0	0	61	716	100	Main Line Auto Attendant - (571) 612 2850
5	Inside Sales	3	5003	68	0	0	0	68	73	100	Main Line Auto Attendant - (571) 612 2850
6	No Selection	NS	NS	61	0	0	0	61	0	0	Main Line Auto Attendant - (571) 612 2850
7	Main Line Auto Attendant - total			250	0	0	0	250	1016	100	Main Line Auto Attendant - (571) 612 2850
8	***Total***			***250***	***0***	***0***	***0***	***250***	***1016***	***100***	-
9											
10	**Grand Totals:**			**250**	**0**	**0**	**0**	**250**	**1016**	**100**	

Figure 56: Sample results from Exporting to file

Receptionist Console

The Receptionist Console is not documented here. Please refer to the XO Hosted PBX Receptionist Admin Guide for details on configuring this feature.

Other topics

Alternate Numbers

An alternate number is a secondary number that can be dialed from the outside world to reach a user's line. This is a convenient way of exposing different telephone numbers to the outside world that route to a single person. You can even choose to have calls to an alternate number ring the phone differently, so that the receiving user can easily determine which number the caller used.

To configure an alternate number, select *Alternate Numbers* from the list on the left side of the *User Features* panel. Select the user that you want to configure with an alternate number. The interface will look like Figure 57:

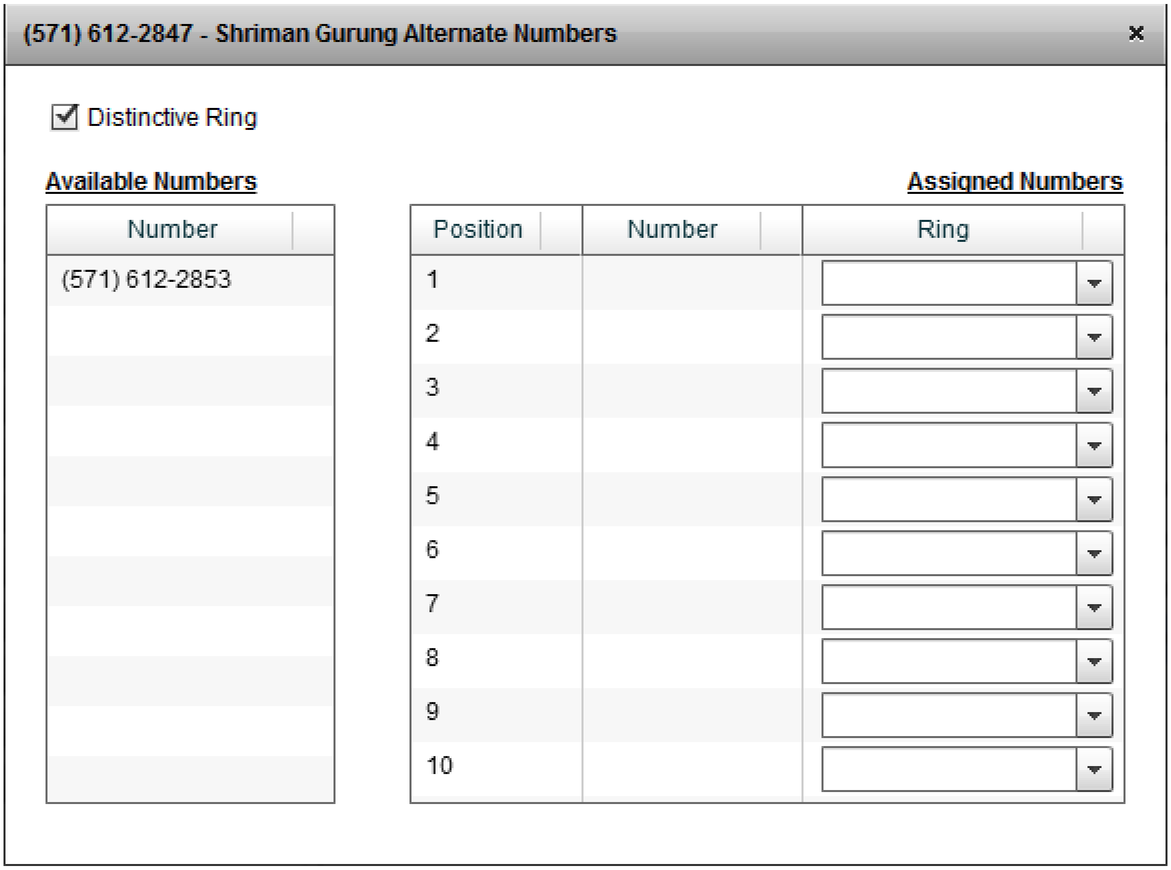


Figure 57: Configuration panel for an alternate number

The numbers that you can assign are in the left hand *Available Numbers* pane. Drag them to any position in the right hand *Assigned Numbers* list. If you want calls to this number to sound the phone's ringer differently, check the *Distinctive Ring* box and select the ring type from the drop-down *Ring* list.

Managing the Voice Portal

The Voice Portal is the phone-based system that lets your users access their voicemail, and administrators configure settings, such as auto-attendant greetings. You will manage the Voice Portal in MySite very rarely. For example, the Voice Portal number is configured when your service is turned up by XO and rarely needs to be changed.

If required, you can change the number of the Voice Portal. This is the number that users can dial to access their voicemail, e.g. when they are away from the office. (Inside of the office, most users do not dial this number to get voicemail: they press a Messages button on their phone, or dial a star code such as `*15`.)

To change the portal number, log on to the MySite portal and click on the Voice Portal button in the Site Services panel. The interface looks like Figure 58.

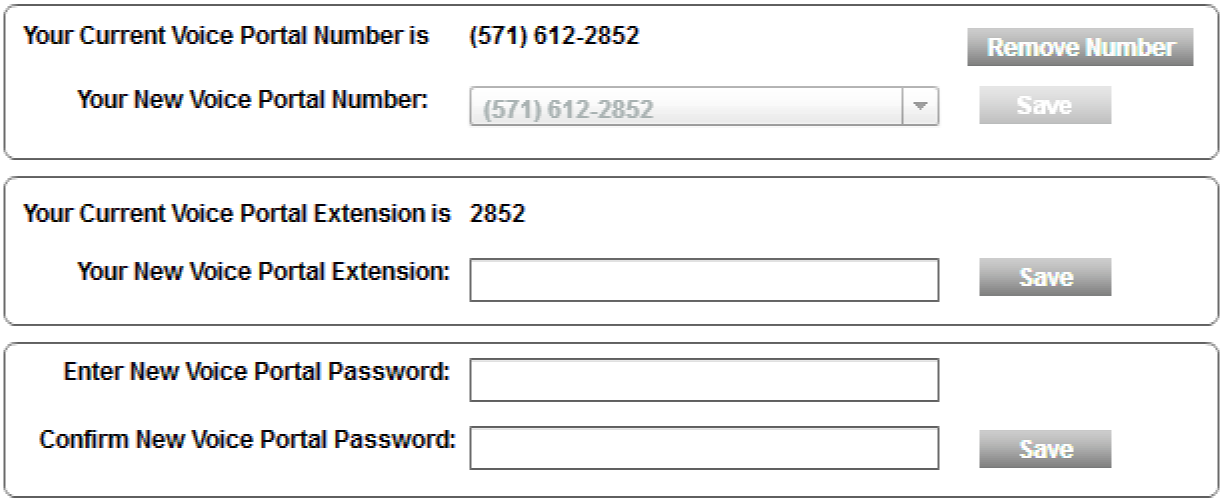


Figure 58: Voice Portal interface panel

Click the *Remove Number* button to remove the number, and then select the new number from the drop-down list before clicking *Save*.

You can also assign an extension number to the Voice Portal: enter the new extension in the *Your New Voice Portal Extension* box and click *Save*.

Finally, you can set a new password (PIN) for administrative access to the Voice Portal (e.g. for updating auto-attendants): simply enter and confirm the new PIN in the *New Voice Portal Password* boxes and hit *Save*.

Selecting an XO Anywhere Portal number

The Anywhere Portal is the number that users dial when they want to use XO Anywhere function, specifically

- to make a call from a mobile device while presenting their business line (Hosted PBX line) ID to the recipient
- to pull a call from mobile to office phone, or vice versa (the "Call Pull" feature).

Like the Voice Portal, the Anywhere Portal is configured by XO when your service is installed and rarely needs to be touched.

If you need to edit the settings, do the following. Select the *Anywhere* button in the *Site Services* tab. The interface will look like Figure 59.

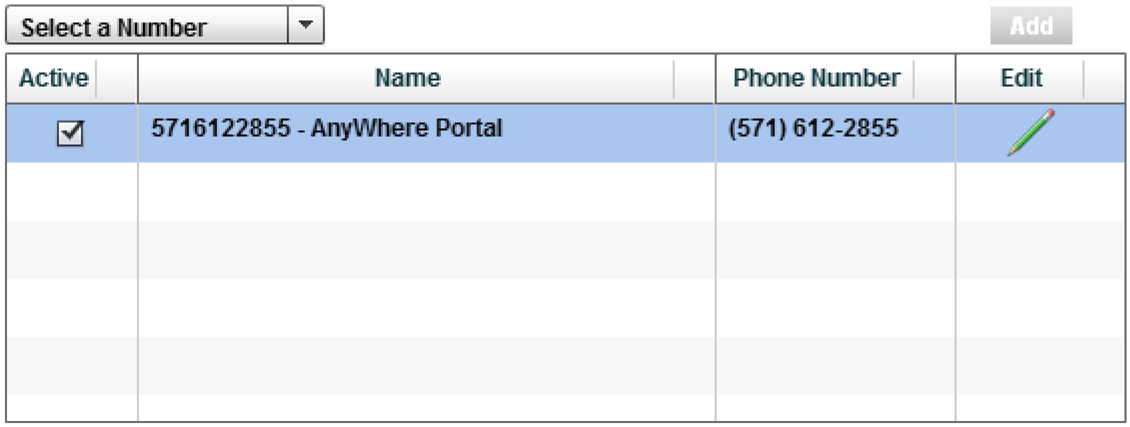


Figure 59: Anywhere Portal number selection panel

- To add a number, select a number from the drop down list and choose *Add*, then *Save* in the window that appears.
- To edit a number, select it and click on the "pencil" icon in the Edit column. The screen will look similar to Figure 60.

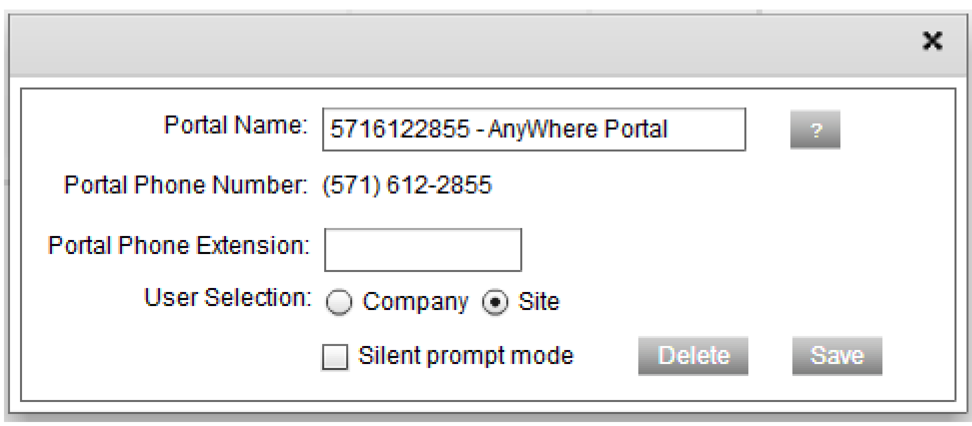


Figure 60: Anywhere Portal - editing settings

- To delete the number, click the *Delete* button.
- To change its description, update the text in the *Portal Name* box.
- To change its internal extension, enter a value. (The extension must not be in use by any other line.)
- The *User Selection* radio buttons control whether access to the portal number is open to all users in the company, or just this site.
- Finally, you can check the *Silent prompt mode* to disable audio prompts when callers dial the Anywhere portal number.

  You might do this if your users intend to program speed dial strings on their mobile devices to access and use the service automatically and do not need to be guided by voice prompts.

- Click *Save* when you are done.

Where To Go Next

This document has covered the fundamentals of the MySite portal. There are many more things that the XO Hosted PBX service can do, including functions such as integrating with Outlook. For more information, please visit the XO Hosted PBX website here.

Revision History

Date	Version	Author	Details
17 Mar 2013	1.01	Gurung	Derived from source documents.
20 Mar 2013	1.02	Gurung	Re-ordered sections by functional area, overhauled content, refreshed screenshots.
24 Mar 2013	1.03	Gurung	Added section on reporting, removed filler
27 Mar 2013	2.0	Gurung	Expanded section on reporting
5 May 2013	2.01	Gurung	Added best practice on PINs
7 May 2013	2.02	Gurung	Improved documentation on permissible extensions
5 July 2013	2.03	Gurung	Updated location of group paging UI and expanded notes on extensions

<End>