

# **Switchvox Administrator Manual**

601-00014-A January 25, 2010

# The Admin Tool Suite

## What is New in version 4.5?

## **Phone Feature Packs: Distinctive Ringtones**

Now you can have Ringtones on your Polycom phones! You can add a Ringtone (.WAV) file to Switchvox, and then SIP extensions can set up their phones to use those Ringtones. For details, see Distinctive Ringtones, or go to the Admin Suite **PBX Features > Distinctive Ringtones**.



If you give permission, each SIP extension can upload its own Ringtones, and set up Ring Rules to be used under particular conditions. For details, see the User Suite's Phone Features.

To use Ringtones and Ring Rules, each phone must have a Phone Feature Packs.

## **Phone Feature Packs: Phone Setup**

The Admin Suite System Setup > Phone Setup page has new features that help you manage your phones:

- Alternate Host for each phone (see Unconfigured Phones)
- Multiple extensions on one phone (see Configured Phones)
- Custom digit map timeout setting for all phones (see Custom Digit Map)
- Custom NTP server setting for all phones (see Custom NTP Server)

To use Phone Setup, you must have a Phone Feature Packs available for each extension.

#### **IVR Menu Actions**

There are new IVR Actions to support Ring Rules:

- Set Distinctive Ring Hint
- Remove Distinctive Ring Hint

#### Language Setting



You can select a language preference for each admin user (including 'admin'), which will also set a Sound preference. For details, see Language and Locale. If you want to use the Soundpack for your preferred language, go to the Admin Suite **Machine Admin > Digium Addon Products** for instructions.

#### **Extension Details & Permissions**



Phone-type extensions now include the following:

- Language
- o Title
- Location
- Picture (image file)
- o Permission to edit Language, Title, Location, or Picture
- o Permission to use the SIP Phone Applications
- Permission to use the Mobile Applications
- o Permission to use Ringtone Rules
- o Permission to upload Ringtones to the extension
- Permission to Record Calls

For details, see Phone and Virtual Extensions, or see the User Suite's Phone Features.

## **Bulk Import Extensions tool**

This tool now includes the fields 'title' and 'location' for each extension. For details, see The Bulk Import Extensions File.

#### **SNMP**

Switchvox's SNMP agent publishes an extensive set of OIDs. Go to **Machine Admin > Network Settings** to set the SNMP agent Login information. For details on what is published, see SNMP Settings.

#### **Event Notifications**

There are two new events that can trigger a notification:

o On Call Answered

Triggered when a call is answered.

On Outgoing Call

Triggered when a call is created.

For details, see URL Manager.

# **Getting Started**

Welcome to Switchvox, the world's most powerful, yet easy to configure IP-PBX.

This documentation describes the Switchvox installation process, how to get started, and how to use the Web-based Switchvox Admin Tool Suite.

Throughout this document, we refer to the Admin Suite menu options as a path, using '>'. Instead of saying "select the Manage Extensions option from the Extensions menu in the navigation bar, we say "select **Extensions > Manage Extensions**."

# **Installing Switchvox**

To install Switchvox, first set up your Appliance according to its instructions. That will include a network connection, and a power supply. Once you have set up your Appliance, the Switchvox software takes a few moments to start up.

When the LCD Panel displays Press any key, Switchvox is ready to be configured.

- 1. Press any key.
- 2. Use the Up and Down arrow keys to select **Configure Network**, then press the green checkmark.
- 3. Use the Up and Down arrow keys to select **Set DHCP/Static**, then press the green checkmark.
- 4. Use the Left and Right arrow keys to select **DHCP** or **Static**, then press the green checkmark.

We recommend using a Static IP address. Because Switchvox is a server, it is important that you assign it an IP address that does not change after it has been configured and users are connecting to it. For details, see Using a Static IP Address.

- If you use DHCP (Dynamic Host Configuration Protocol), we recommend that you configure your DHCP server to reserve an IP address just for Switchvox. For details, see Using DHCP.
- 5. Use the Up and Down arrow keys to select View System Info. This displays the URL of the Switchvox Admin Tool Suite. You can go to any computer on your network, open a Web browser, and go to this URL. For details, see About the Admin Suite.

## Using a Static IP Address

If you are using a Static IP address, you need your IP addressing information: IP address, subnet mask, gateway (or router), and DNS server. If you do not have this information, ask your network administrator.

**Note:** To find an IP address that is not in use on your network and will not be used for another client by the DHCP server or used by some other devices, go to the following web site and look at the information for your particular router: http://www.portforward.com/routers.htm

To select each addressing item, use the Up and Down arrow keys on your LCD Panel.

When you are done setting the IP addressing information, use the Up and Down arrow keys to select **Save Net Settings**. Now the network is configured, and you can complete the installation process (go back to Installing Switchvox).

#### **IP Address**

The IP address provided by your network administrator.

Use the Left and Right arrow keys to place the cursor, and the Up and Down arrow keys to change the numbers.

#### Netmask

By default, the netmask is 255.255.25.0. If your netmask address is different, change it to the correct value.

Use the Left and Right arrow keys to place the cursor, and the Up and Down arrow keys to change the numbers.

#### **Gateway**

The address for your gateway is usually the IP address of your router. A common gateway IP address is 192.168.0.1.

Use the Left and Right arrow keys to place the cursor, and the Up and Down arrow keys to change the numbers.

#### **DNS Server**

This address is for a server inside your network or a server your Internet Service Provider (ISP) runs for you. If your ISP runs your DNS, the network administrator there can provide this address.

Use the Left and Right arrow keys to place the cursor, and the Up and Down arrow keys to change the numbers.

## **Using DHCP**

If your router can assign IP addresses using DHCP, then you can use this method to assign the Switchvox IP address. Your router needs to know the MAC address of the active network card in the Switchvox server. This MAC address is displayed on the Switchvox server screen as a series of six number or letter combinations separated by colons (e.g., 00:11:11:AF:2A:EA). On your router, follow the instructions for entering this MAC address. Each time Switchvox is rebooted, it gets the IP address. Because DHCP works differently on different routers, we can't offer instructions for how to set up your router. We suggest that you contact your router's manufacturer or look in your router's user manual.

After you select **DHCP**, use the Up and Down arrow keys to select **Save Net Settings**. Now the network is configured, and you can complete the installation process (go back to Installing Switchvox).

## Installing the Software from CD-ROM

Place the Switchvox Installation CD-ROM in the Appliance's CD-ROM drive and then reboot. You are prompted to start the installation process by pressing **enter** on your keyboard.

## **Setting the Time Zone**

After you start the installation process, a map is displayed. Select a location on the map that indicates the time zone in which Switchvox will operate.

If you select the incorrect time zone, or you move to a different time zone later, you can change it after Switchvox is installed.

Setting the time zone is particularly important for generating accurate call reports.

## Formatting the Hard Drive and Installing the Software

After you select a time zone, the system formats the hard drive and installs the software. When that is complete, you are prompted to reboot. First remove the installation CD from the drive, and then press **enter** on your keyboard to reboot.

## **Basic Server Functions**

There are several functions that can or must be done at the Switchvox server; those functions are listed here. Use the LCD Panel on your Appliance if there is one, or connect a keyboard and monitor to the Appliance.

1. View System Info

This displays the URL of the Switchvox Admin Tool Suite.

2. Configure Network

This lets you change your network settings. These are the same options that you used when you first set up your Switchvox server (see Installing Switchvox). You can only do this from here at the server.

Reboot PBX

This reboots Switchvox. You can also do this from **Machine Admin > System Reload** (see System Reload).

4. Shutdown PBX

This shuts down Switchvox and turns off the power on the appliance. You can also do this from **Machine Admin > System Reload** (see System Reload).

5. Reset HTTPS certificate

This resets your SSL certificate. You can only do this from here at the server.

6. Tech Support Access

This lets Switchvox technical support representatives log into your Switchvox Appliance. You can also do this from **Diagnostics > TSR Report** (see TSR Report).

7. Restore Web Access

This restores local network access to the Web-based Switchvox Admin Tool Suite. If you turned off this access in the *Local Network* rule in Access Control, you can turn it back on using this option. You can only do this from here at the server.

8. Reset Admin Passwd

This resets the main Admin password (user name 'admin') to **admin**, which is the default password. You can only do this from here at the server.

9. Change LCD Passwd

This lets you set or change the password for the LCD Panel. If you cannot remember your LCD password, you can clear it using the Admin Suite (see Managing Physical Access to the Server). You can only do this from here at the server, and it is only available if you have an Appliance with an LCD Panel.

10Lock LCD

This immediately puts the LCD Panel in password-request mode. You can only do this from here at the server, and it is only available if you have an Appliance with an LCD Panel.

## About the Admin Suite

The Switchvox Admin Tool Suite (the Admin Suite) lets you administer Switchvox via a Web browser.

# **Logging Into the Admin Suite**

On a computer that is on the same network as Switchvox, open a Web browser and enter the URL of the Admin Suite (see The Switchvox Admin Suite URL).

https://YourServerIPaddress/admin

The Login page for the Admin Suite is displayed.

Please log in to use the	e Tool Suite
User Name: admin	
Password:	
Log In	

The first time you use the Admin Suite, enter the default User Name and Password, and click Log In.

User Name: adminPassword: admin

For security, you are prompted to change the default password to a secure password. The strength of your new password is indicated as you type, to help you create a good password.

The next time you go to the Admin Suite, log in using the User Name admin and your new password.

After you log in, the main page of the Admin Suite is displayed.

Note: If you have forgotten your password, see Basic Server Functions.

To log out of the Admin Suite, click **Logout** (in the upper right-hand corner).

## **Navigating the Admin Suite**

The Admin Suite navigation bar contains the following dropdown menus:

#### Extensions

Manage all of your extensions and extension templates, specify extension settings, and manage extension groups and permissions.

#### **PBX Features**

Set up and manage options such as music on hold, time frames, and IVR menus.

#### System Setup

Configure hardware, channels, incoming call rules and routes, and other settings that form the foundation of your Switchvox PBX.

#### **Diagnostics**

See logs and reports, and troubleshoot Switchvox.

#### Machine Admin

Modify your network settings, manage system backups, apply updates, and more.

# **Getting Help**

Click any help link to open a pop-up window with information about a particular tool. Help links are available as question-mark icons and as text links.



If a pop-up window does not display, set your Web browser to allow pop-ups from the domain of your Switchvox server.

# **Registering Switchvox**

After a new install, you are required to register Switchvox to make it fully operational. Select **Machine Admin** > **Product Registration** to display the **Product Registration** page.

Enter your contact information and Registration Code. The contact information you provide is useful if you need to call on our technical support services. Your 16-character Registration Code is located on the Switchvox CD-ROM case. Be sure to enter it exactly as shown, including capitalization, with four characters in each of the four boxes.

Click **Submit** to register Switchvox with the registration management system.

Extensions	PBX Features	System Setup	Diagnostics
Product Registration			
Use the form below to	register your PBX.		
Company Name: (re	quired)		
Phone Number: (req	uired)		
Address Line 1: (req	uired)		
Address Line 2:			
City: (required)			
State: (required)			
- Please select a cou	ntry first —		
Zip Code: (required)			

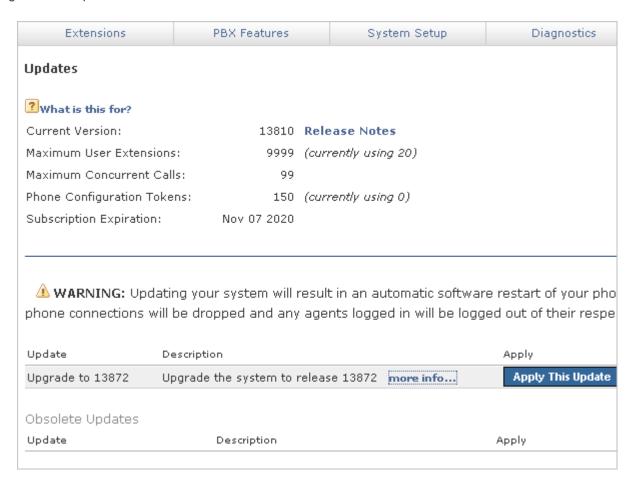
# **Updating Switchvox**

Before you start using Switchvox to make and receive calls, be sure that you are running the newest version of the software. Select **Machine Admin > Updates**.

If the Updates page says that no updates are available for your system, then you already have the newest version of the software.

If there is an update listed on this page, click **Apply This Update**. The update is applied and the system is restarted. You can learn more about this update by clicking **more info...**.

**IMPORTANT:** Applying an update means that the server is restarted, which drops all current calls and logs all agents out of queues.



# **Supported Phones**

Switchvox supports two general types of handsets: SIP/VOIP phones and standard analog phones (as you might use in your home). Voice Over Internet Protocol (VOIP) is the generic name for any technology that allows transmission of a voice telephone conversation over a network, whether it is a local network or over the internet.

## **SIP Phones**

SIP phones are digital phones that convert and send your phone calls to Switchvox through your Ethernet network. Outwardly, they look like typical office telephones. Some SIP phones have two Ethernet connections, so that you can connect your PC to the network through the phone. You can also use a softphone, which is software that acts like a phone.

## **Analog Phones**

Analog phones need an Analog Telephone Adapter (ATA) to connect to Switchvox. The ATA converts the signals sent by the handset into a digital signal then sends it the same way a SIP phone sends its signal. An alternative to using ATAs is to use a channel bank. A channel bank is hardware typically used on legacy phone systems. One channel bank can take a set of 24 analog phone lines and turn them into one T1 line that can be connected directly to Switchvox via an add-in card.

Analog phones require no configuration; however, depending on the method you use to connect the analog phones to Switchvox, you may have to configure either an ATA or a channel bank. If you purchased your ATA with Switchvox, it is preconfigured. If you are using a channel bank, you need to wire all the phones into the channel bank, configure the channel bank to communicate with the phones, then add the lines to Switchvox (see Hardware Setup and Hardware Setup).

### **Video Phones**

Video phones are digital phones that convert and send your video calls to Switchvox through your Ethernet network. Switchvox supports video calling over a SIP provider, and works with a number of video clients including desk phones and softphones.

# Making Your First Call

If you purchased your phones along with Switchvox, then plug each phone into your network and turn it on. When a phone displays the date and time and its extension number, it is ready for use.

If you did not purchase your phones along with Switchvox, then you need to set them up for use with Switchvox. For instructions on how to do this, see Phone Setup.

When your phones and extensions are ready, pick up one of the handsets and dial 800, which rings a sample Interactive Voice Response (IVR) menu. The call should connect and you should hear a recorded voice say "congratulations, Switchvox has been configured properly."

# **About Faxing**

Switchvox lets you send and receive faxes. You can send a fax from your fax machine, or send a file from your computer to a fax machine. Incoming faxes are sent to a Switchvox extension, and can be found in that extension's Mailbox.

**IMPORTANT:** We strongly recommend handling faxes over an analog or PRI channel, or over a SIP provider using T.38. If this is not possible, then you should set up dedicated extensions for incoming and outgoing faxes. If you do not use analog, PRI, or T.38 over a SIP provider, incoming fax detection and fax quality could be less than optimal.

To enable faxing capabilities, you must first install your Fax license and the related Fax software (go to **Machine Admin > Digium Addon Products)**.

## **Incoming Faxes**

You can enter a **Default Fax extension** for each Channel Group or SIP Provider. When a fax comes in, the fax is routed to this extension.

You can set up an Incoming Call Route so that one phone number (DID) can handle someone's voice calls and faxes. This sends incoming faxes to the extension's *Fax* folder, and incoming voice calls to the extension's phone.

You can also set up an Incoming Call Route so that all activity on a phone number (DID) is treated as a fax and sent to the same extension. This is useful if you want or need a dedicated incoming fax extension.

For more details, see Incoming Call Routes.

## **Outgoing Faxes**

If you want to use a fax machine, you can dedicate its extension to just sending faxes. In that case, Switchvox handles all outgoing activity as a fax (prints a fax-file, puts it in the Fax.Outbox folder in that extensions's Mailbox, and sends the fax). For details, see the User Suite This extension only sends faxes. You can also set up a *Feature Code* extension that lets you send a fax from your fax machine, and put the fax into another extension's Outbox. For details, see Feature Code Extensions.

If you want to fax files from your computer, you need to set up a printer driver on your computer. Each Switchvox user can set up a printer driver for a dedicated outgoing fax extension, or for their own extension (or both). With the former, outgoing faxes are found in the dedicated extension's Mailbox. With the latter, outgoing faxes are found in the user's Mailbox. For details, see the User Suite Creating & Sending a Fax.

# **About Chatting**

Switchvox includes an XMPP-based Jabber server for presence and real-time communications. You can use your favorite XMPP-based client to interact with the server, or the Switchboard offers a Chat panel. The main page of the Chat Panel offers a list of people you can double-click to start a chat.

You may want to set up a DNS entry with a hostname for your Switchvox Jabber server. You can set the Jabber hostname in the **Network Settings** (**Machine Admin > Network Settings**). If you are using peered Switchvoxes, you can also set the peer's Jabber hostname in its VOIP Provider information.

# **About Video Calling**

Video calling is available over a SIP provider. For call quality, we recommend using the same kind of phone on both ends of the call. If you are using softphones (software for making telephone calls over the Internet), we recommend using the same software on both ends. Also, make sure that you have installed the latest updates for your phone.

**IMPORTANT:** Video calling is not available over IAX. It is available over peered Switchvox servers, but all servers must be running Switchvox SMB 4.0 (or later). If there is a point in the call where IAX or an older version of the Switchvox software is involved, the video portion of the call will be dropped and only audio will be available.

# **Extensions**

The Extensions tools let you manage the extensions on your system, add new ones, and use templates and settings to customize your extensions.

# **Manage Extensions**

Extensions are the numbers used by Switchvox to send a caller to various destinations. There are many different types of extensions, including extensions for phones, IVRs, call queues, etc.

The Manage Extensions tool shows the extensions that are currently in Switchvox, and lets you create, edit, and delete extensions.

When you first set up Switchvox, there are several predefined extensions. We suggest that you leave those extensions in place until you are sure that you do not want to use them.

# **Viewing Extensions**

There are several ways to view the extensions in Switchvox. By default, you see all of the extensions, listed by extension number in ascending order, with 50 extensions per page.

To limit the types of extensions in the list, select an extension type from the **View** dropdown and click **Go**. The list redisplays with only extensions of that type.

To find a particular extension, use the **Search** box. Enter at least 3 numbers of the extension, or at least 3 letters of the first or last name. The list is narrowed down to just the extensions that match.

To sort the table by a column, click on the column name.

## **Password Warnings**

There are warning icons next to some or all of your extensions because the passwords chosen for these extensions are weak. Weak passwords can make your Switchvox vulnerable to intrusion. If your Switchvox is behind a firewall or not accessible from the Internet, you may wish to ignore these warnings, but we still recommend you choose stronger, more secure passwords where recommended.

## **Level 2 Warning**

The password that a SIP phone uses to register with Switchvox is the same as the extension. (e.g., Extension: 101, Phone Password: 101) Please note that extensions designated with a Level 2 warning could also meet the conditions of a Level 1 warning.

Recommended action: click that extension's **Modify** button, then click the **Show Advanced Options** link under **Phone Settings**. Locate the Phone Password field, type a longer, more secure password, and then be sure to retype it to confirm. Click **Save Extension Settings** at the bottom of the page.

## **Level 1 Warning**

The password to log into the User Portal or to check voicemail is the same as the extension. (e.g., Extension: 101, Voicemail Password: 101.)

Recommended action: click that extension's **Modify** button, and then locate the Numeric Password field. Type a longer, more secure password, and then be sure to retype it to confirm. Click **Save Extension Settings** at the bottom of the page.

# **Extension Types**

There are many different types of extensions in Switchvox.

The three phone and virtual extension types (SIP, Analog, and Virtual) have most of the same settings and they have a corresponding User Suite account. Other types of extensions do not have a user name and password for a User Suite account.

- Phone and Virtual Extensions
- o IVR (Interactive Voice Response) Extensions
- Simple Conference Room Extensions
- Meet-me Conference Center Extensions
- Group Pickup Extensions
- Intercom / Paging Extensions
- Directory Extensions
- Voicemail Access Extension
- Call Parking Extensions
- Dial Tone Extensions
- Feature Code Extensions
- Call Queue Extensions
- Agent Log In Extensions
- Agent Log Out Extensions

# **Creating One Extension**

To create one extension, click Create A New Extension.

Select the type of extension from the dropdown.

If you are creating a phone-type extension, you also need to select the Extension Template to use. Leave this as *Default* if you have not created your own templates.

If you are creating a feature-code type of extension, you also need to select the Feature Type.

Click **Create** at the end of the page to create the new extension and display the settings page. When you have completed the settings, click **Create Extension** to save the extension and go back to the Manage Extensions page.

## **Phone and Virtual Extensions**

The settings are mostly the same for SIP phones, analog phones, and virtual extensions. Phone and virtual extensions have a corresponding User Suite account.

#### SIP Phone or SIP Adapter for Analog Phone (ATA)

A SIP phone, or an analog phone plugged into a SIP Adapter (e.g., Sipura SPA-1001).

#### **Analog Phone**

A regular phone (not VOIP/SIP) that is physically plugged into the Switchvox server.

#### Virtual Extension

An extension without a phone. This type of extension behaves like a SIP Phone extension, except that it does not have a Switchboard and it does not show the phone as **Unreachable** in System Status. This is useful as a general voicemail box, or for call rules that always cascade to someone's personal phone (so that the caller doesn't need to know your personal number).

## **Extension Settings**

The Extension settings include basic information about the extension and the person who uses it.

#### Extension

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **FXS Channel**

This is for *Analog Phone* extensions only. Select the channel that the phone is plugged into. If the only option here is *Unconfigured Channel*, then you need to configure your hardware device channels before this phone can work (see Channel Admin).

#### **First Name**

Enter the first name of the owner of this extension. This is the name that is used in the caller ID name, and as the extension's identification throughout Switchvox. Check the *User can edit* box if you want the extension owner to be able to modify this setting.

#### **Last Name**

Enter the last name of the owner of this extension. This is the name that is used in the caller ID name, and as the extension's identification throughout Switchvox. Check the *User can edit* box if you want the extension owner to be able to modify this setting.

#### **Email Address**

Enter the email address of the owner of this extension. This is used to send voicemail notification messages. Check the *User can edit* box if you want the extension owner to be able to modify this setting.

#### **Numeric Password**

Enter a Numeric Password for this extension. This extension owner uses this to log into their account in the *User Tool Suite*. The strength of the new password is indicated as you type (although in this case you may be setting a generic password for the owner to log in for the first time). Re-enter the password for verification. Check the *User can edit* box if you want the extension owner to be able to modify this setting.

Note: This password is numeric because it will be entered from the numeric keypad on a telephone.

#### Force user to update password on next login

This is a security precaution. It lets you create all extensions with a generic password (such as the extension number), which lets the extension owner easily log in for the first time. But then it forces her to create a new (hopefully secure) password before continuing to use the software.

## **Language and Locale**

The Switchvox web-based tool suite is available in several languages:

- English for United States
- English for United Kingdom
- Spanish for Spain
- Spanish for Mexico
- Italian

To set your preferred language:

- 1. Select a language from the dropdown.
- 2. Select a locale for that language.

If you choose to use a language other than English for US, there are some important issues to be aware of:

Dates

All other languages and locales handle dates with the format of day, then month, then year (as opposed to mm/dd/yyyy).

Sound Packs

There are Sound Packs available for each Language preference. Once you select a language, Switchvox serves you the sounds from that language's soundpack. For example, if you switch to the Italian language, and then you call the voicemail access extension, the prompts will be in Italian. Soundpacks must be installed by the Switchvox administrator (see the Admin Suite Digium Addon Products).

Creating Extensions

By default, new extensions are created as English for US. If you want new extensions to default to your preferred language, use Extension Templates.

#### Force user to confirm language on next login

If you are not sure which language this extension prefers to use, force the owner to choose a language the next time he or she log in. This way, extension owners can decide for themselves which language to use, and you know that they made a deliberate choice.

## **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

#### Voicemail/Fax Mailbox Quota (MB)

Enter a number of megabytes here to set a disk-space quota. This is the maximum number of megabytes that can be used for voicemail messages and faxes. This value can be set for extensions and extension groups.

- If you leave this blank, it indicates unlimited space.
- If you enter a 0 (zero), it indicates that no voicemail or faxes can be stored.
- If a quota is set for a group or groups, and for an extension in the group, the extension's mailbox quota is used.
- If an extension is a member of multiple groups, the lowest mailbox guota is used.
- If an extension is not allowed to have voicemail, this setting is not relevant.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

### **Phone Settings**

This section is only for SIP Phone or SIP Adapter for Analog Phone (ATA) extensions.

Phone Settings	
DTMF Mode  What is DTMF Mode?	rfc2833 (Default) 🔻
Click to Show Adva	nced Options

#### **DTMF Mode**

DTMF tones are the sounds emitted when you press buttons on your phone.

- SIP extensions: Refer to your phone's user manual for the DTMF mode that your phone uses.
- SIP providers: Ask your provider which DTMF mode it supports.

If you are unsure of which DTMF mode to select, use RFC2833 (the most common method).

### **Phone Password**

Enter your phone password and verify it.

This is the Authentication Password that your SIP phone uses for registration with Switchvox.

This password may be different than the password that you use to login to administer your SIP phone through the manufacturer's interface. Also, this password does not change when the user changes the voicemail password.

### **Supported Codecs**

Select the codecs your phone supports. Check your phone's user manual if you do not know them.

For video, H.263 and H.264 are on by default.

For audio, ULAW, ALAW, and G722 generally work with every phone and provide the best sound quality. They are selected by default.

The following table describes the audio codecs. If you have Switchvox SMB and you have entered a License key for the G729 Codec in **Machine Admin > Digium Addon Products**, you will also see G729 in the list of codecs.

**Table 1: Audio Codecs** 

Codec	Bandwidth Usage (kbits/s)	Sound Quality	CPU Usage
ULAW	85	Great	None
ALAW	85	Great	None
G722	85	Excellent	Low
G726	54	Good	Low
SPEEX	33	Great	High

GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

#### **Phone NAT Traversal**

This option controls how Switchvox communicates with the phone if it is behind a router performing Network Address Translation (NAT). The following options are available:

Always. Select if you are unsure if the phone is behind a router performing NAT.

No. Select if Switchvox is behind NAT but the phone is not.

**Never.** Select to prevent attempts to translate even if the handset requests it with the *rport* tag.

Route. Select to send responses symmetrically, but not to include the *rport* tag in the *Via* header.

### **Call API Settings**

The call creation API lets external applications originate phone calls via Switchvox. The API is accessed by requesting a specially formed URL. This request could come from an external program or by typing the URL into your web browser. To use this feature, check with your Switchvox administrator.

#### **Profile Information**



An Extension Profile includes some or all of the following:

- o Image
- First and Last Name
- Title
- Location
- o Extension Number

Profiles can be shown in the Switchboard and on a phone that uses a Phone Feature Pack. For details, see Your Phone Feature Pack.

#### To add a picture:

- 1. Click Add an image.
- 2. Click **Browse**, and then find the image file on your computer. Images must be at least 75x100 pixels, and must be in jpg format.

#### 3. Click Upload.

The image is uploaded and displayed with the crop area that will be used to make a 75x100 image. You can move the crop area around to choose the best portion of the picture. The Preview on the right shows you what the picture will look like once it is cropped.

- 4. Click Save Image.
- 5. Click Save Extension Settings.

To upload a different picture, re-crop the existing picture, or delete the existing picture:

- 1. Click View Image.
- 2. Click ReCrop, or Delete.

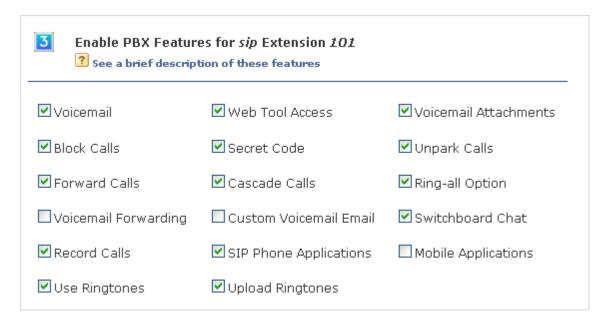
If you want to upload a different image, **Delete** this image now, and then click **Add an image** to start again.

- 3. Confirm your changes.
- 4. Click Save Extension Settings.

Check the appropriate boxes if you want the extension owner (user) to be able to change his or her Location, Title, or Picture.

## **Enable PBX Features**

These settings let you specify which features the extension is allowed to use.



#### Voicemail

Let this extension send unanswered calls to their voicemail Mailbox.

#### **Web Tool Access**

Let this extension log into the Switchvox User Tool Suite. If this is not enabled, this extension cannot connect to Switchvox using the Extend API.

#### Voicemail Attachments

Let this extension attach the voicemail sound file to the voicemail notification email message. Otherwise, the extension-owner has to use his phone or the User Suite to listen to voicemail messages.

#### **Block Calls**

Let this extension set up Call Block Rules for blocking a call by prefix, area code, specific phone numbers, etc.

#### **Secret Code**

Let this extension set up Call Rules with a secret code.

#### **Unpark Calls**

Let this extension retrieve calls that have been parked.

#### **Forward Calls**

Let this extension set up Call Rules that forward a call to another phone number or extension.

#### **Cascade Calls**

Let this extension set up Call Rules that cascade a call to another phone number or extension. When you cascade a call, if the call is not answered then the next Call Rule is applied.

#### Ringall

Let this extension set up Call Rules that ring several phone numbers or extensions at once.

#### **Voicemail Forwarding**

Let this extension automatically forward all of its voicemail messages to another extension or extension group.

#### **Custom Voicemail Email**

Let this extension customize the email message templates used for its voicemail notifications.

#### **Switchboard Chat**

Let this extension open and use the Switchboard Chat Panel.

#### **Record Calls**

Let this extension record its own calls. An extension-owner can record his or her own calls using the Switchboard **Current Calls** panel's Record button. The recording is sent to the extension's voicemail Mailbox.

#### **SIP Phone Applications**

Let this extension use the Switchvox tools on its SIP Phone: Directory, Voicemail Mailbox, Parking Lots, Phonebooks, and Record Call.

#### **Mobile Applications**

Let this extension use the Switchvox Mobile Applications.

For information about Mobile Applications, see the web page Switchvox Add-ons.

#### **Use Ringtones**

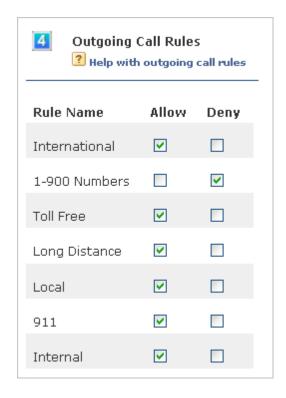
Let this extension create its own Ring Rules. For details, see the User Suite's Distinctive Ring Rules.

#### Upload Ringtones.

Let this extension upload its own ringtones. For details, see the User Suite's Distinctive Ringtones.

#### Outgoing Call Rules

Specify which of the outgoing call rules you want this extension to be able to use. Check *Allow* to allow this extension to use the rule. Check *Deny* to deny this extension use of this rule. Leave both boxes unchecked to let this extension bypass this rule. For details, see Outgoing Call Rules.



If you deny access to a rule, the call will not be completed, and the caller will hear a message that they cannot make that call. If you bypass a rule, the call is passed to the next outgoing call rule, if one is available.

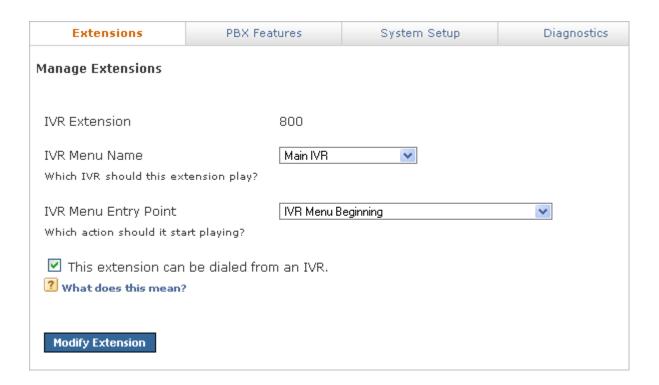
## **Extension Groups**

Select the groups this extension belongs to. To select multiple groups, hold down the CTRL key. For more information, see Extension Groups.

## **IVR (Interactive Voice Response) Extensions**

An extension that presents a menu to the caller, or performs an action. (This is also known as 'Auto-Attendant' in some PBX systems.) For example, it could ask the caller to press 1 for directions, or press 2 to hear the company directory. Or, it could say that everyone is in a meeting, then send the call to voicemail. For informationabout setting up IVRs, see IVR Editor.

You must have an IVR menu to assign to an IVR extension. The IVR extension 800 is preconfigured with the IVR menu 'Example IVR,' which you can use as a basis for your custom IVRs.



#### **IVR Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **IVR Menu Name**

Select the IVR menu that you want use when this extension is called.

#### **IVR Menu Entry Point**

Select an Action that is in the IVR menu. (You don't have to start at the beginning of the menu.)

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

## **Simple Conference Room Extensions**

This extension provides a single conference room for anyone who calls or is transferred to this extension.

Extensions	PBX Features	System Setup	Diagnostics
Manage Extensions			
Conference Room Exte 4 digits in length	ension		
This extension can be dialed from an IVR. What does this mean?			
Play sound when people enter and leave			
☐ Play Music On Hold	when only 1 person is in	the conference	
Conference participant	s may press # to be sen	t to extension 🔃 🛮 🕹	3
Create Extension			

#### **Conference Room Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

#### Play sound when people enter and leave

Check the box if you want this to be true.

## Play Music On Hold when only 1 member is in the conference room

Check the box if you want this to be true.

#### Conference participants may press # to be sent to extension

Enter the extension number if you want this to be true.

### **Meet-me Conference Center Extensions**

This extension provides an access point to each user's conference room. Callers to this extension are prompted for a PIN corresponding to a specific user's conference room. (See the User Suite Conference Setup.)

Extensions	PBX Features	System Setup	Diagnostics	
Manage Extensions				
Meet Me Conference Center Extension  3 digits in length				
This extension can be dialed from an IVR. What does this mean?				
After Unlimited 🕶 invali	d conference room numb	er attempts route caller t	o extension	
Create Extension				

#### **Meet Me Conference Center Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

#### After [] invalid conference room number attempts route caller to extension []

Select the number of attempts and enter the extension number if you want this to be true.

## **Group Pickup Extensions**

A Group Pickup extension can pick up a call ringing on another extension. For example, assume that the Group Pickup extension is 110, the authorized extension is 111, and the group includes extension 112 and 113, in that order. If extension 112 is ringing, dialing 110 from extension 111 picks up the call ringing on 112. If extensions 112 and 113 are ringing, extension 112 is picked up (calls are picked up in the same order as the extensions are listed).

#### **Group Pickup Extension**

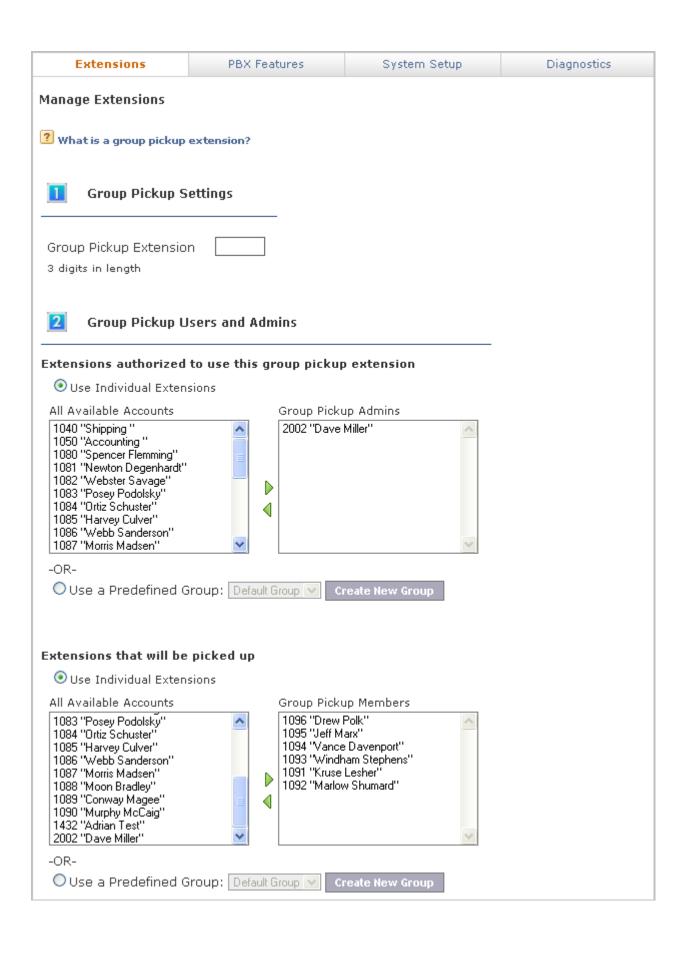
Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **Group Pickup Users and Admins**

Put individual extensions into the Group Pickup Admins box, or select an extension group to give that entire group authorization to use this extension.

#### Extensions that will be picked up

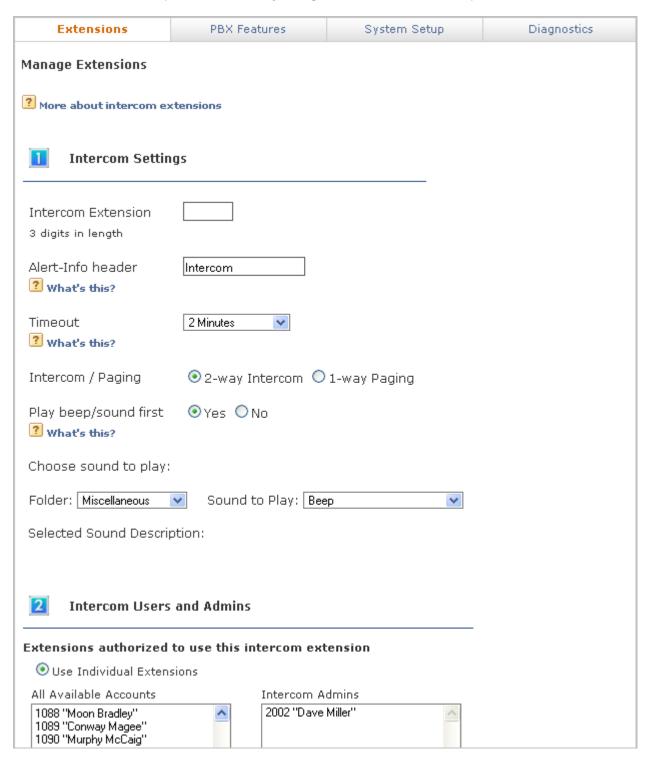
Put individual extensions into the Group Pickup Members box, or select an extension group to make all of the extensions in the group be members.



## **Intercom / Paging Extensions**

An Intercom / Paging extension can make an announcement on a group of speaker phones. Phones in the group do not ring, they answer immediately in speaker-phone mode (the caller can begin talking and be heard over the phone's speaker).

*Note:* You must have compatible and correctly configured handsets to use this option.



#### Intercom Extension

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### Alert-Info header

The value that is sent to the phone in the alert info header. This setting is required for Polycom phones to autoanswer intercom calls.

Use a value of *Intercom* or *SVAutoPickup*. If you use *Intercom*, a ringtone will play before the phone autoanswers the call. With both values, the phone auto-answers and then you hear a tone announcing the call.

#### **Timeout**

Select the length of time that the system attempts to connect an intercom call. After this duration, intercom calls that have not been answered will be hung up.

#### Intercom / Paging

Specify whether this extension makes intercom calls (2-way conversations) or paging calls (one-way announcements).

#### Play beep/sound first

Specify whether or not a beep/sound plays to indicate that the call is beginning. If you choose Yes, then you can also choose a sound to play (select a folder then select a sound file). The **Selected Sound Description** tells you about the sound.

#### **Intercom Users and Admins**

Use this section to set up which extensions can use this Intercom/Paging extension, and which extensions are called when this Intercom/Paging extension is dialed.

#### Extensions authorized to use this intercom extension

Put individual extensions into the Intercom Admins box, or select an extension group to let that entire group use this extension.

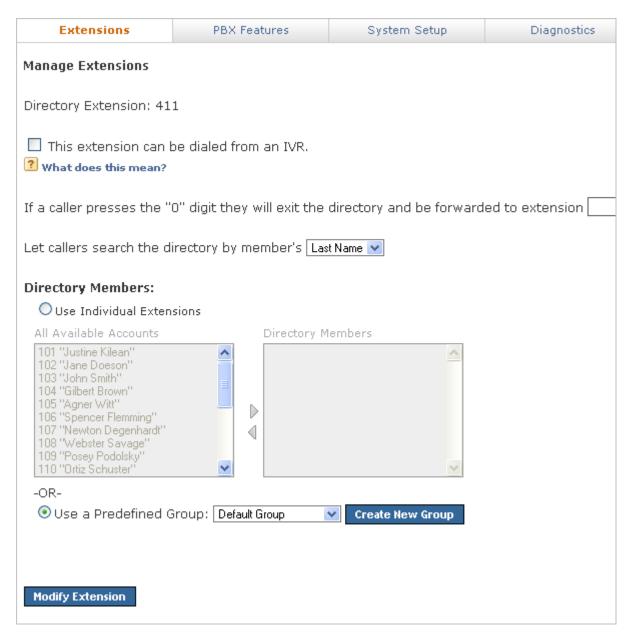
#### Extensions that will be called

Put individual extensions into the Intercom Members box, or select an extension group to make all of those extensions recipients of calls to this extension.

## **Directory Extensions**

A Directory extension presents the caller with a directory of all user-type extensions (extensions with a first and last name). The caller is prompted to type in the first three letters of the party's first or last name and is presented with a list of matching extensions.

The Directory extension 411 is preconfigured, and includes the Default Extension Group (which is all phone-type extensions).



## **Directory Settings**

The Director settings determine how callers can interact with the Directory.

#### **Directory Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

If a caller presses the "0" digit they will exit the directory and be forwarded to extension [] Enter an extension if you want this to be true.

#### Let callers search the directory by member's []

Select from First Name or Last Name.

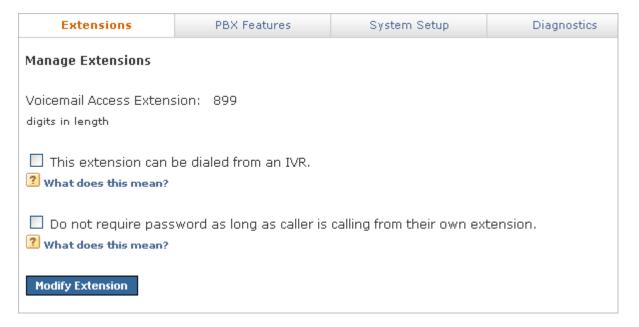
### **Directory Members**

Put individual extensions into the Directory Members box, or select an extension group to make all of those extensions members of the Directory.

### **Voicemail Access Extension**

A Voicemail extension goes directly to the voicemail system where extension owners can check their voicemail. This is not an extension to leave a voicemail message, it is strictly for retrieving voicemail.

Voicemail extension 899 is preconfigured for you.



#### **Voicemail Access Extension**

Enter a unique number for this extension (if you want to use a different extension than 899, first delete the 899 extension, then create a new 'Voicemail Access' extension).

For information about the extension-number length, see Extension Settings.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

Do not require password as long as caller is calling from their own extension Check this box if you want this to be true.

## Call Parking Extensions

A Call Parking extension 'parks' a call so that it can be retrieved from another phone. Users transfer the call to the 'main' Call Parking extension, but the call is parked in one of the extensions in the range specified.

Extensions	PBX Features	System Setup	Diagnostics	
Manage Extensions				
Call Parking Extension: 700				
Parking lot will reserve all extensions between 701 through 799.				
Return parked calls to the extension that parked the call after 90 seconds.				
Modify Extension				

#### Parking lot will reserve all extensions between [] and []

Enter the extension numbers to be used for parked calls. A Call Parking extension 700 is preconfigured, and reserves all extensions between 701 and 799.

For information about the extension-number length, see Extension Settings.

#### Return parked calls to the extension that parked the call after

Enter the number of seconds that a call should remain parked before returning the call to the extension that parked the call. The default is 90 seconds.

### **Dial Tone Extensions**

A Dial Tone extension plays a dial tone when it is called.

### **Dial Tone Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **Feature Code Extensions**

A Feature Code extension lets you perform a certain operation by dialing a \*, the two-digit feature code, then another extension.

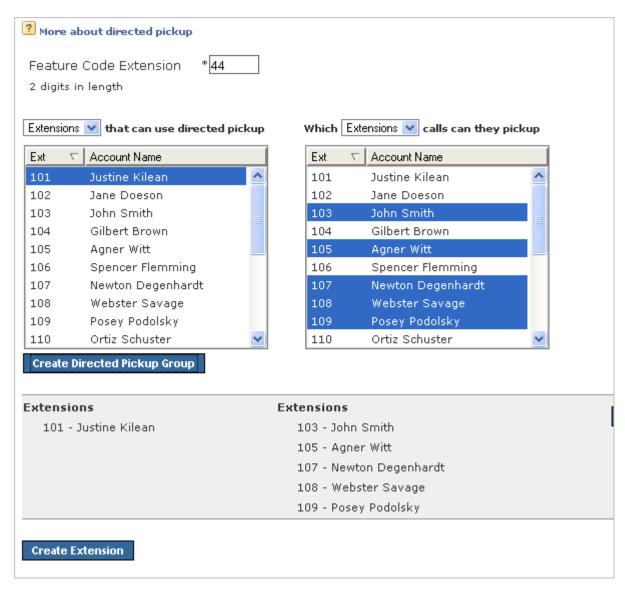
#### Go To Voicemail

Directly dials another extension's voicemail. For example, if you define 56 as the Go To Voicemail feature code, to transfer a caller to extension 202's voicemail, simply transfer them to \*56202.

#### **Directed Pickup**

Answers a call that is ringing a different extension. For example, if you define \*39 as the Directed Pickup feature code, dialing \*39202 will pick up a call ringing 202's extension.

Define which extensions or extension groups can use this feature, and define which extensions can be picked by those authorized extensions.



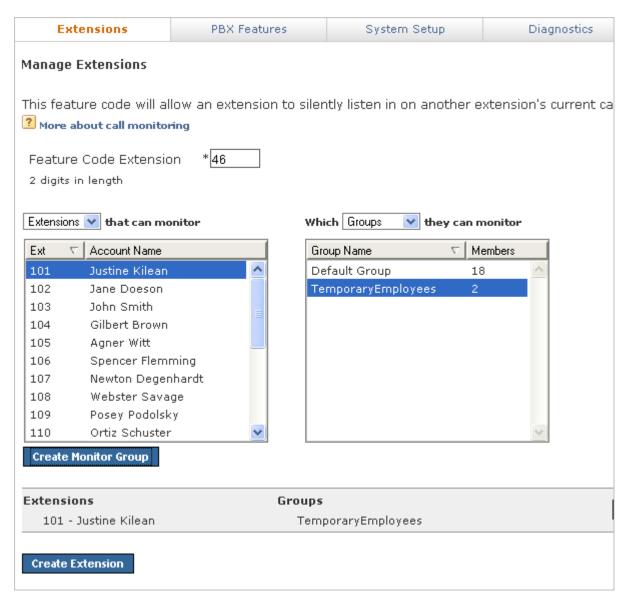
#### **Call Monitor**

Silently listens in on another extension's current call. For example, if you define \*92 as the Call Monitor feature code, dialing \*92202 will listen in on 202's call without either party knowing.

While monitoring a call you can switch to different modes by pressing the corresponding digits on your phone:

- Press **5** for whisper mode. This mode allows you to talk to the extension you're monitoring but does not let the other party hear you.
- Press 6 for barge mode. This mode allows both parties to hear you.
- Press 4 to return to the default monitor mode.

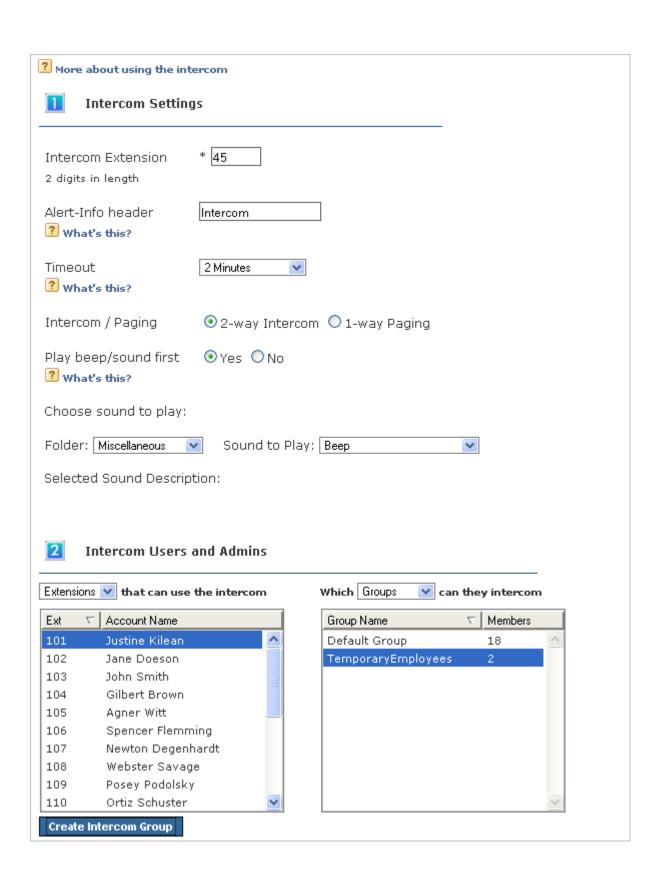
Use the extension boxes to define which extensions or extension groups can use this feature, and define which extensions can be monitored by the authorized extensions.



#### **Personal Intercom**

Directly dials another extension's intercom. For example, if you define \*55 as the intercom feature code, and you dial \*55202, you will now be talking over the speaker on extension 202's phone. *Note:* This feature requires intercom support on the handset receiving the call.

Define the basic Intercom settings. Define Which extensions can use this feature, and define which extensions can receive these calls from the authorized extensions.



#### Fax

Sets a system extension as the sender of the fax, and then sends a fax. The fax is stored in the Sent Faxes of the designated extension. Check **Require extension password** if you want to require the sender to enter the password for the designated extension.

For example: you have a Fax Feature Code extension of 33. You want to send a fax as if it were from extension 200, which has a voicemail password of 111. You must dial 9 for an outgoing line, and you are sending the fax to 1-888-555-1212. You would dial the following from your fax machine:

- \*33\*200\*111\*918885551212. If the password is not required, just leave it out:
- \*33\*200\*918885551212. In both cases, the fax is sent to the number 1-888-555-1212 and is stored in the Sent Faxes of extension 200.

Extensions	PBX Features	System Setup	Diagnostics
Manage Extensions			
This feature code is use designated extension.  More about the fax feat	ed to set a system extens ture code	sion, and then send a fax	. The fax is stored in
Feature Code Extension  2 digits in length  Required extension pages			
Create Extension	333W01 G		

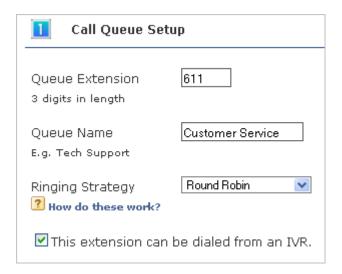
## **Call Queue Extensions**

A Call Queue extension lets your route calls to a group of extensions so that any one of them can pick up the calls.

For example, a sales queue might be a group of extensions that includes all of the sales people. When a caller wants to speak with a sales person, he reaches the queue, and the queue rings the queue members as specified. That means the caller is connected to a sales person as efficiently as possible.

### **Call Queue Setup**

The Setup settings determine the basic information about this queue.



#### **Queue Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **Queue Name**

Enter the name of the queue (e.g., Sales or Customer Service).

#### **Ringing Strategy**

Select one of the following strategies:

Ring All. Ring all queue members simultaneously until someone answers.

Round Robin. Take turns ringing each queue member evenly as calls come into the queue.

Least Recently Called. Ring the queue member for whom the most time has passed since taking his last call.

Fewest Calls. Ring queue member who has taken the fewest calls in the queue.

Random. Randomly ring queue members.

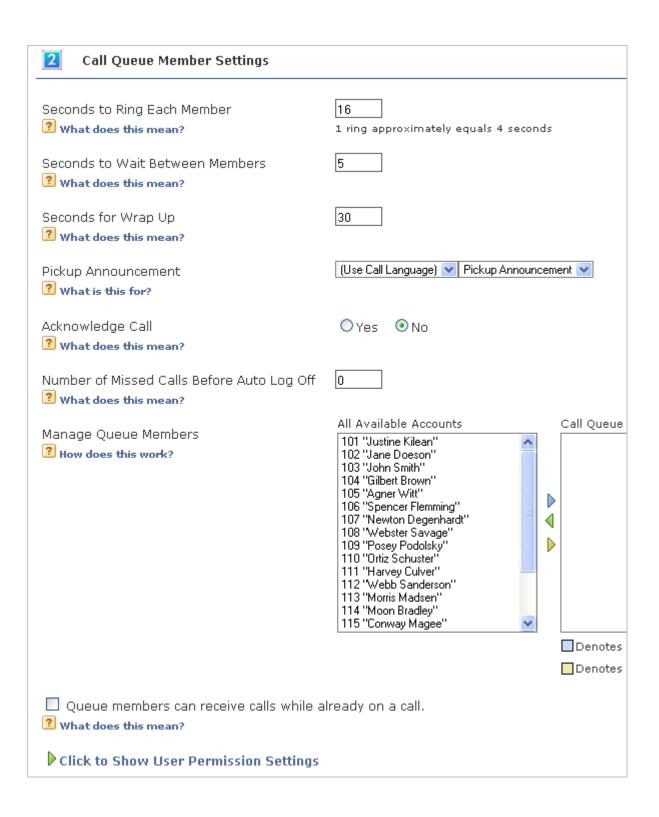
In Order. Ring queue members in order, always starting with the first member in the list.

#### This extension can be dialed from an IVR

Check this option to let IVRs dial this extension. IVR menus must also have the **Allow Extension Dialing** option enabled (see Adding & Modifying IVR Options).

#### **Call Queue Member Settings**

The Member settings determine how the queue interacts with its members.



#### Seconds to Ring Each Member

Enter the maximum number of seconds a queue member's phone can ring before the system stops ringing that member, and goes to the next member of the queue.

There are some issues to keep in mind:

- If a queue member has a call rule that answers sooner than this setting, the call rule is applied. For
  example, if a queue member has a call rule that sends the call to voicemail after two rings, and this
  setting is five, the call goes to that member's voicemail and not to the next member of the queue.
- Some phones may stop ringing and give up after some number of seconds or rings. If this setting is a large value (above 60), and you notice the phones stop before that number of seconds, you might want to try to change your phone to increase this limit.

#### **Seconds to Wait Between Members**

Enter the maximum number of seconds a queue member's phone can ring before the system goes on to ring the next member of the queue.

**Note:** If you set this to a low number and have a large *Ring all* queue, or a queue where no one is logged in, there is a potential for system performance issues. This is because the system keeps trying to find someone to answer the queue call. Unless you have a specific reason for doing so, it is recommended to leave this value at the default of 5.

#### Seconds for Wrap Up

Enter the minimum amount of time (in seconds) after disconnecting a call before the agent can receive a new call from the queue. For example, if this is set to 10, an agent's phone will not ring again for at least 10 seconds after completing a call.

#### **Pickup Announcement**

Select a language and a sound file to play when a queue member answers a queue call. After this sound is played, the queue member is connected to the caller. This sound lets the queue member know that the call is from a queue, and which queue. This is especially useful for people who answer for multiple queues.

To use the language that has been set for this current call, select *Use call language*. If you are only using one language in Switchvox, use this setting. To use a specific language for the pickup announcement for this queue, select that language.

The sound-file dropdown automatically lists all of the sound files in the Sound Manager's Call Queue folder. For more information, see Sound Manager.

#### **Acknowledge Call**

Select Yes to require members to press the **1** key to accept the call. If the member hangs up, then the call goes to the next appropriate queue member.

### Number of Missed Calls Before Auto Log Out

This setting controls when members are automatically logged out of this queue. (Permanent queue members can never be logged out.)

- If a member misses more calls in a row than the number in this setting, he is automatically logged out of this queue. Calls will not ring this member until he logs in again.
- If you do not want members to be logged out automatically, set this to 0 (zero).

#### **Manage Queue Members**

Queue Members are extensions that can log into the queue and receive calls from the queue. Only phone extensions are valid queue members. Permanent members are permanently logged in, so they can always receive calls to the queue.

To add extensions to the queue, select the extensions on the left, then click one of the right-facing arrow buttons: Add Login Member (blue) or Add Permanent Member (yellow). The selected extensions are moved to the **Call Queue Members** box on the right.

To select multiple extensions, hold down CTRL and then select. To select multiple extensions that are all next to each other in a block, hold down Shift and select the first and last extensions.

To remove an extension from the queue, select the extension and click the left-facing green arrow button.

The order of members in the list is important. For example, if *Round Robin* is the ring strategy, it calls the member in the first position, then the second, and so on. If *In Order* is the ring strategy, it always tries to call the members in order, starting with the first member.

To change the order of the member list, highlight an extension and use the up/down arrows to move the extension up or down in the list.

### Queue members can receive calls while already on a call

Check this box if you want this to be true. By default, queues do not ring members who are already on a call. **Note:** Your phone must be able to support multiple simultaneous calls for this feature to work properly.

### **User Permission Settings**

Set the permission level for extensions to see this queue in the Queue Panel on their Switchboard. To set these permissions, click the link **Click to Show User Permission Settings**.

**Filter list by Extension Group** lets you display all phone extensions, or just the extensions in an extension group.

Highlight an extension and select one of the following permission levels:

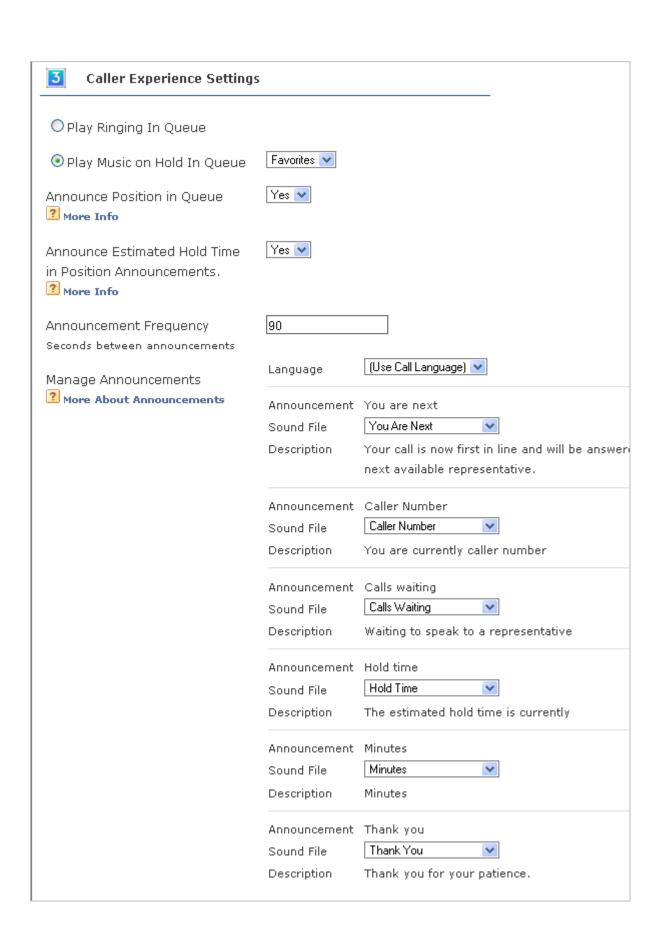
My Status: Lets the extension open this queue panel and see his call information for today.

**Overview**: Lets the extension open this queue panel and see his call information for today, and an overview of all the queue information for today.

**Detailed View**: Lets the extension open this queue panel and see his call information for today, an overview of all the queue information for today, and the members and callers in the queue.

### Caller Experience Settings

The Caller Experience settings offer two options for what happens when a caller is waiting in the queue.



### Play Ringing in Queue

Let the caller hear the call ringing the queue member(s) phones.

#### Play Music on Hold in Queue

Play Music on Hold, and optionally play announcements.

If you select Play Ringing In Queue, then the rest of the settings here are unavailable.

If you select **Play Music on Hold In Queue**, then you can use the rest of the settings here to customize the caller's experience.

#### **Announce Position in Queue**

Select Yes or No. If this is Yes, the caller hears an announcement when they enter the queue, and the announcement says what the caller's position is in the queue. The announcement is repeated depending on the Announcement Frequency setting.

If this is No, the caller hears Music on Hold with no position announcement.

# Announce Estimated Hold Time in Position Announcements Select Yes or No. The caller hears an estimated hold time along with his position in the queue. The estimated hold time is derived from how many other calls are in the queue and the average time it took to answer previous calls.

### Announcement Frequency

Enter number of seconds between announcements.

### **Manage Announcements**

There are various announcement sounds that are played to the caller when he or she is waiting in a queue. All of these sounds can be customized for this queue. Each announcement listed here has a description, and a sound-file dropdown to select the sound that plays when this announcement is triggered.

The sound-file dropdown automatically lists all of the sound files in the Sound Manager's Call Queue folder. For more information, see Sound Manager.

Make the following selections for the announcements:

**Language:** Play all announcements from sound files of this language. To use the language that has been set for this current call, select *Use call language*. If you are only using one language in Switchvox, use this setting. To use a specific language for these announcements in this gueue, select that language.

You are next: Your call is now first in line and will be answered by the next available representative

Caller Number: You are currently caller number

Calls waiting: Waiting to speak to a representative

Hold time: The estimated hold time is currently

Minutes: Minutes

Thank you: Thank you for your patience In Queue Call Routing Settings

The In Queue Call Routing settings determine any alternative call routing. Enter values in the boxes (shown as blanks here) to define queue call routing.

In Queue Call Routing
If a caller has been waiting in the queue for seconds, route the caller to extension Leave this blank to leave callers in the queue indefinitely.
If there aren't any members logged into the queue, route all incoming queue calls to extension Leave blank to keep callers waiting in the queue until a queue member logs in.
If there are unanswered calls in the queue, route all new incoming queue calls to exte Leave blank to have an unlimited queue length.
If a queue call has passed through the Ringing Strategy infinite v times without being answer to extension <u>&amp;</u>
While in a queue allow callers to dial a 0 to be routed to extension &
Create Call Queue

If a caller has been waiting in the queue for [] seconds, route the caller to extension [] Enter a number of seconds, and the extension. If this is left blank, callers remain in the queue indefinitely.

If there aren't any members logged into the queue, route all incoming queue calls to extension [] Enter an extension. If this is left blank, callers are kept waiting in the queue until a queue member logs in.

If there are [] unanswered calls in the queue, route all new incoming queue calls to extension [] Enter a number and an extension. If this is left blank, there can be any numbers of callers waiting in the queue.

If a queue call has passed through the Ringing Strategy [] times without being answered then route that call to extension []

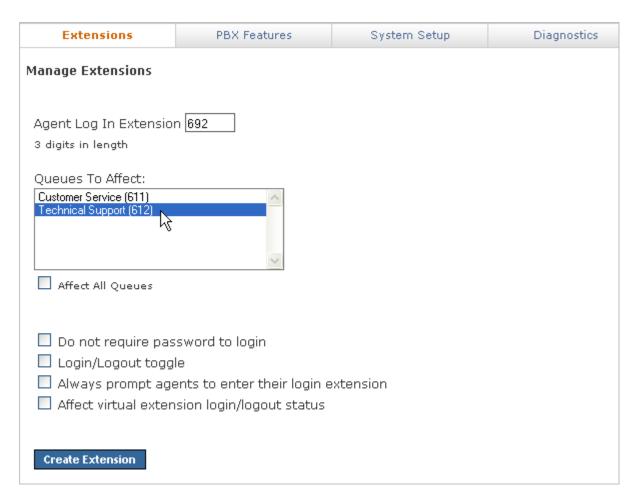
Enter a number and an extension. If this is left blank, callers remain in the queue indefinitely. The effect of this setting depends on the Ringing Strategy. For example, if the Ringing Strategy is *Ring All*, then the call rings all queue members and that is considered one pass through the strategy. If the Ringing Strategy is *Round Robin*, then the call rings each queue member, and once all queue members have been tried, that is considered one pass through the strategy.

While in a queue allow caller to dial a 0 to be routed to extension [

Enter an extension if you want this to be true. **Note:** Click the People icon to select from a list of available extensions.

### Agent Log In Extensions

An Agent Log In extension is used to log an extension into call queues, and/or to log a virtual extension into a phone. This extension can also be used to toggle logging in and logging out, so that you do not have to have a separate *Agent Log out* extension.



### **Agent Log In Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

#### **Queues to Affect**

Highlight the queues that a caller should be logged into when calling this extension. Hold down CTRL to highlight multiple queues. Check **Affect All Queues** if you want that to be true. If you don't want to affect call queues with this *Agent Log In*, do not select any queues.

### Do not require password to login

Check this box if you want this to be true. When people call this extension, they are logged in without having to enter a password.

### Login/Logout toggle

Check this box if you want people to call this extension to be logged in or out. If already logged in, callers are logged out, they are logged in. Using this option, you do not need a separate *Agent Log Out* extension.

### Always prompt agents to enter their login extension

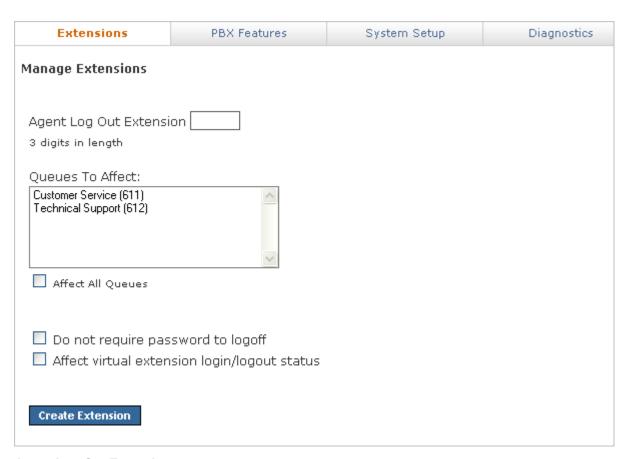
Check this box if you want people to enter their extension to log in. This is useful if you have people who share work areas and phones, or if you have virtual extensions that are used by different people.

### Affect virtual extension login/logout status

Check this box if you want this log-in action (or log out, if toggling) to affect virtual extensions. For example, if a virtual extension is logged into a phone and into a queue, should the action of logging out apply to the virtual extension, or just the queue?

### **Agent Log Out Extensions**

An Agent Log Out extension is used to log an extension out of call queues, and/or to log a virtual extension out of a phone.



### **Agent Log Out Extension**

Enter a unique number for this extension. For information about the extension-number length, see Extension Settings.

### **Queues to Affect**

Highlight the queues that a caller should be logged out of when calling this extension. Hold down CTRL to highlight multiple queues. Check **Affect All Queues** if you want that to be true. If you don't want to affect call queues with this *Agent Log Out*, do not select any queues.

### Do not require password to log out

Check this box if you want this to be true. When people call this extension, they are logged out without having to enter a password.

#### Affect virtual extension login/logout status

Check this box if you want this log-out action to affect virtual extensions. For example, if a virtual extension is logged into a phone and into a queue, should the action of logging out apply to the virtual extension, or just the

# **Modifying one Extension**

To modify an extension, click on **Modify** for that extension. You can modify all of the extension information except for the extension number. If you want to change the extension number, you must delete this extension and create a new one.

### Log in as this user

Use the **Log in as this user** button to open the Switchvox User Tool Suite for an extension. This lets you view and edit the extension's Tool Suite without knowing and entering the password. The extension-user can be logged in at the same time, so be careful when either of you are making changes.

# **Deleting one Extension**

To delete an extension, click on **Delete** for that extension. To keep a copy of the extension's voicemail, click **Download Voicemail Archive** and save a copy of that file. Be sure that you do indeed want to delete the extension and its voicemail, then click **Yes, Delete Extension**.

# Creating & Updating Multiple Extensions

The Bulk Import Extension tool lets you create and update multiple SIP or Virutal extensions.

To do this, you need to create an Extension File that is formatted correctly with specific extension information. Then, you can use the **Bulk Import Extension** tool to identify that file, evaluate the extensions that will be created or modified, and then perform the import.

### The Bulk Import Extensions File

The Bulk Import Extensions File must be in Comma Separated Value (CSV) format.

The first line of the file must be a header line that labels each column. Each subsequent line should be an extension line that contains the values of an extension.

Depending on the type of extensions you are working with (SIP or Virtual) different information is required or accepted. You can use a file that has additional columns of information (Switchvox ignores this information), as long as the required extension information is included correctly, The order of the labels in the header is not important, but the values of each extension must match the order of the labels.

If you are working with SIP extensions, the value of phone\_password cannot be the same as extension.

If you are updating an existing extension and you have some information that you do not want to change, leave that value empty. If you want to make it blank, put a space in that column.

When your Extension File is ready, go to **Extensions > Manage Extensions** and click **Bulk Import Extensions** (see Bulk Import Extensions Settings).

### File Information

This table describes the data for each type of import:

	ext	fname	Iname	email	title	location	password	phone_password
SIP: New	1	1	1	1	1	1	1	✓
SIP: Update	1	1	1	1	1	1	✓	✓



### File Examples

Example: SIP Extension File

ext,fname,lname,email,password,phone\_password 100,Joe,Smith,jsmith@home.com,100,s3CurEp@S\$ 101,Jane,Doe,jdoe@home.com,101,c@nT6ueSS 102,Dan,Jackson,djackson@home.com,102,g0ODP@\$\$

Example: Virtual Extension File with location and other information

fname, Iname, email, dept, ext, password, location Joe, Smith, jsmith@home.com, sales, 100, 100, 3A Jane, Doe, jdoe@home.com, sales, 101, 101, 3B Dan, Jackson, djackson@home.com, facilities, 102, 102, 9A

Example: SIP Extension Updates File without passwords with a location

ext,fname,lname,email,password,phone\_password,location 100,Joe,Smith,jsmith@home.com,,,2A 101,Jane,Doe,jdoe@home.com,,,3B 102,Dan,Jackson,djackson@home.com,,,2C

### **Bulk Import Extensions Settings**

The Bulk Import Extensions settings determine how to manage this bulk import.

When the settings are correct, click **Evaluate Import** to evaluate the Extension File and see what will happen if you import the file.



#### The Extension File

Browse your machine to find your Extension File.

For details about creating this file, see Creating & Updating Multiple Extensions.

### **Extension Type**

This is the extension type that is used for all of the extensions that are created.

### **Extension Template**

This is the Extension Template that is used to create or update all of the extensions. All of the values in the template are used for each of the extensions in the Extension File.

### On Duplicate Extension

This option specifies how to respond when an extension in the Extension File already exists on the system.

**Skip**: Do nothing with the extension data in the file, and skip to the next extension in the file.

**Delete and Recreate**: Delete the existing extension in the system, then create the extension with the values from the file and the chosen extension template. **Note**: only SIP, Analog and Virtual extensions are affected by this. For example, if you have a Call Queue extension of 333 and have a line in your file to create an extension of 333, this line in the file is skipped even if you have Delete and Recreate chosen.

**Update Information**: Update the existing extension with the values from the file and the chosen extension template. **Note**: the update is only applied if the extension types are the same. For example, if you have an existing Virtual extension of 100, and have a line in your file to create a SIP extension of 100, this line in the file is skipped even if you have Update Information chosen.

### **Bulk Import Extensions Evaluation**

When you click **Evaluate Import**, Switchvox opens your Extension File and evaluates it against the extensions that already exist. The results are shown, with how many extensions will be created, how many are duplicates and what will be done with those duplicates, and how many are invalid and cannot be created.

If you have invalid extensions, check your Extension File for formatting problems or errors. Be sure that your extension numbers are the same length as specified in **Extension > Extension Settings**.

If you want to proceed with the import, click Yes, Process Extensions.

Extensions	PBX Features	System Setup	Diagnostics
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### **Manage Extensions**

### **Bulk Import Extensions**

The following is an evaluation of the Extension File import. The actual number of extensions to be less if there are validation errors with the data in the file.

### **Extension Creation**

Extensions that will be created: 22 New Extension Type: Virtual Extension

Extension Template: Default

### **Extension Duplicates**

Duplicate extensions that will be updated: 0

### Creation Exceptions

Invalid extensions that will not be created: 0

Duplicate extensions that cannot be updated: 1

Only extensions that are the same type as the type chosen for the creation process can be updated by



Are you sure you want to process this bulk extension file?

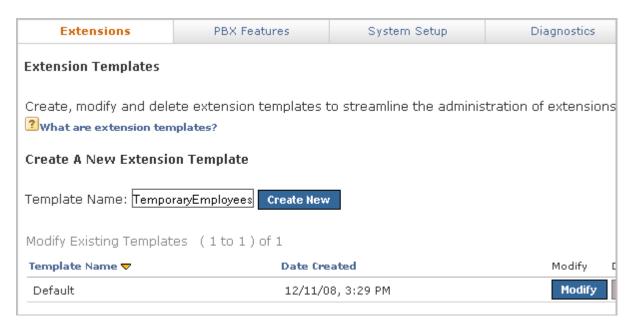


Cancel

# **Extension Templates**

Extension Templates contain most of the same settings that you use when you create phone or virtual extensions, and the templates help you manage the settings for all of those extensions. You can modify the Default template to suit your needs, or create multiple templates for different types of extensions. For example, you might want all sales staff to have a similar setup: a *SIP phone* with *Voicemail Forwarding* that belongs in the call group *Sales*. You can use that template to create all of the sales staff extensions, which saves you time and ensures consistency.

An extension created with a template can always be changed. If you have a special case, you can create the extension with a template, and then modify the extension.



# **Creating an Extension Template**

On the Extension Templates page, enter a unique name for the new template then click **Create New**. The new template is created and the settings page is displayed with the original Switchvox default settings. Make your changes, then click **Modify Template** to save your changes and return to the Extension Templates page.

An Extension Template has all of the same details as Phone and Virtual Extensions.

# **Deleting an Extension Template**

To delete an Extension Template, click its **Delete** button. Be sure that you do indeed want to delete the template, then click **Yes, Delete Template**.

## **Modifying an Extension Template**

To modify an Extension Template, click its **Modify** button. The settings page for the template is displayed. Make your changes, then click **Modify Template** to save your changes and return to the Extension Templates page.

An Extension Template has all of the same details as Phone and Virtual Extensions.

# **Extension Settings**

The Extension settings let you control some basic extension functionality.

When the settings are correct, click **Save Extension Settings**.



#### **Extension Length**

Select 3, 4, 5, or 6 digits. This is the length of the extensions in Switchvox. Changing this number does not change existing extensions.

6-digit extensions are only available in Switchvox SMB.

Keep in mind that if you go to a higher number, and you still have lower extension numbers, there might be a slight delay in internal dialing. For example, if you call a 3-digit extension, Switchvox waits a moment before connecting the call, to see if you are going to dial a fourth digit (or fifth or sixth).

### Beep after an assisted transfer

Check this box if you want to play a beep sound when an assisted transfer is done. For example: Jill puts Jack on hold then calls Jane, and asks her if she can speak with Jack. Jane agrees, and Jill transfers Jack to Jane. When the transfer is complete the beep is played, Jill is off the call, and Jack and Jane are on the call and can begin talking.

# **Extension Groups**

Extension Groups let you group extensions together so that you can apply changes and features to many extensions at a time.

The Default Group includes all of the phone-type extensions that you put in Switchvox. If you add extensions using the extension form, or a CSV file, those extensions are included as **Group Members**. Although you can change the Default Group, you may not want to. If you change the Default Group, then from that point on, new extensions go into the **All Available Accounts** List not the **Group Members** list. That means that you won't have an extension group that automatically includes everyone.

# **Creating a New Extension Group**

Click Create a New Extension Group to add a new extension group.

#### **Group Name**

Enter a unique name for this group.

### **Manage Group Members**

These boxes let you select the accounts (extensions) that belong in this Extension Group. Any extensions in the **Group Members** list is part of this Extension Group.

Click on the column headings to sort by that column.

To select multiple extensions, hold down the CTRL key and then select. To select multiple extensions that are all next to each other in a block, hold down the Shift key and select the first and last extensions.

Use the green right and left arrows to move extensions from one list to another.

The Default Group includes all of the phone-type extensions that you put in Switchvox. If you add extensions using the extension form, or a CSV file, those extensions are included as **Group Members**. Although you can change the Default Group, you may not want to. If you change the Default Group, then from that point on, new extensions go into the **All Available Accounts** List not the **Group Members** list. That means that you won't have an extension group that automatically includes everyone.

### Voicemail/Fax Mailbox Quota (MB)

Enter a number of megabytes here to set a disk-space quota. This is the maximum number of megabytes that can be used for voicemail messages and faxes. This value can be set for extensions and extension groups.

- If you leave this blank, it indicates unlimited space.
- If you enter a 0 (zero), it indicates that no voicemail or faxes can be stored.
- If a quota is set for a group or groups, and for an extension in the group, the extension's mailbox quota is used.
- If an extension is a member of multiple groups, the lowest mailbox quota is used.
- If an extension is not allowed to have voicemail, this setting is not relevant.

# **Modifying an Extension Group**

Click the **Modify** button for the group you wish to modify from the Extension Groups page. This displays the Modify Group page.

# **Deleting an Extension Group**

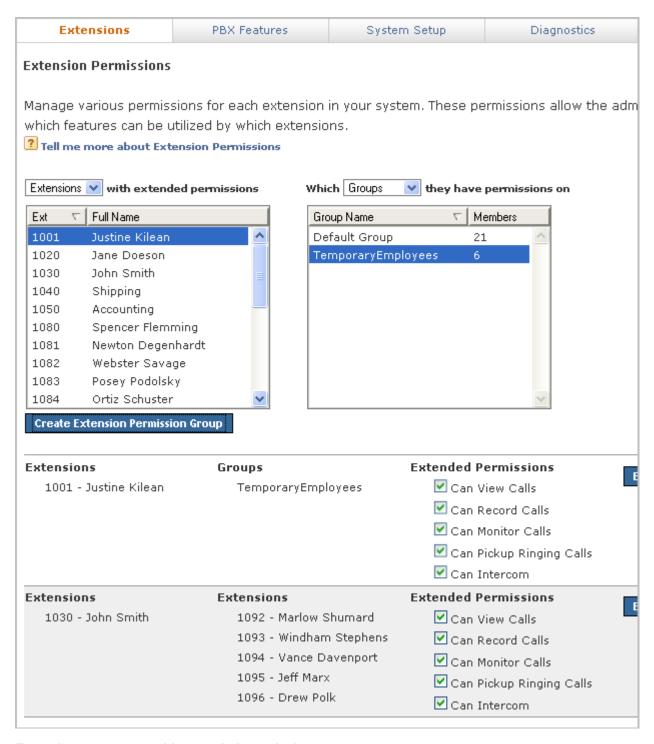
Click the **Delete** button for the group you wish to delete from the Extension Groups page. This displays the Delete Group page.

Click Yes, Delete Extension Group to remove the group.

# **Extension Permissions**

Extension Permissions let you set permissions for the **Extended Entry** features in the **Switchboard Phonebooks**.

After these permissions are set up, the managing extension owners need to add the managed extensions to the **Extended Entries** section of their Phonebooks.



### Extensions or groups with extended permissions

In the left-hand box, select one or more extensions, or an extension group. The extension(s) here are granted permissions to manage the extensions on the right.

### Which extensions or groups they have permissions on

In the right-hand box, select one or more extensions, or an extension group. The extension(s) here are the ones that can be managed by the extensions on the left.

### **Can View Calls**

Let the managing extension view the caller ID of the managed extension's current calls.

### **Can Record Calls**

Let the managing extension record the managed extension's current calls.

### **Can Monitor Calls**

Let the managing extension silently listen in on the managed extension's current call.

### **Can Pickup Ringing Calls**

Let the managing extension answer a call that is ringing the managed extension.

### **Can Intercom**

Let the managing extension initiate an intercom connection with the managed extension.

### **Extended Permissions**

Once you have created a permission-set, check the boxes for each permission that should be granted. If you make changes to the permissions, click **Save**.

To change the permission-set, click **Edit**. The extension boxes are reset with this permission-set, and you save your changes or cancel the editing.

# **PBX Features**

The PBX Features let you set up and manage the features of Switchvox, such as Music on Hold, Time Frames, and IVR menus.

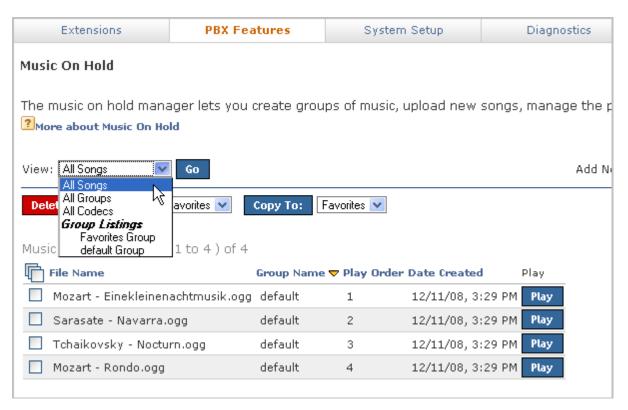
# **Music On Hold**

Music On Hold (MOH) lets you set up audio program(s) for callers to hear when they are not actively on a call (i.e., on hold, waiting in a queue, parked, or being transferred). Although the audio in MOH is referred to as 'songs', you can also include custom announcements, and update them as often as you like. It's a great tool for communicating with your customers.

MOH is organized in groups of songs. Most MOH situations use the default group, which you can customize by adding and deleting songs. MOH in a call queue can use any group. For example, you can create a MOH group with announcements about special offers, and assign that group to your sales call queue.

The default MOH group is a selection of classical songs. You can add songs to this group, and delete any songs that you don't want.

Switchvox supports the Ogg Vorbis codec for MOH files. This means that any file you want to use for MOH needs to end in .ogg, or it won't play properly. However, Switchvox can support other types of sound files if you add a new codec. (For more information, see Adding a Codec.)



# Managing Music On Hold

The MOH Main page lists *All Songs* in MOH. For each song, it lists its group, play order, and date added. To sort by column data, click that column name, and click it again to toggle ascending/descending order. You can also play the files.

To list all of the groups, select All Groups and click Go.

To list the songs in a group, select the group name and click **Go**.

To list all of the codecs that you can use, select All Codecs and click Go.

### **Adding a Group**

To add a new group, select *Group* from the **Add New** dropdown, then click **Go**. This displays the Music On Hold Group page.

Click **Create Group** to save your new group and go back to the Music on Hold page. A success message indicates the group was created, and the group list displays with the new group highlighted in blue.

### **Group Name**

Enter a unique name for this group.

### Description

Enter a brief description of the group.

### **Play Volume**

Select Soft, Normal, Loud or somewhere in between on the adjustable slide bar.

### **Play Order**

Select *Chosen Order*, which plays in the order displayed in the list, or *Random*, which randomly plays all the songs in the list.

### **Modifying a Group**

To modify a group, select *All Groups* from the **View** dropdown, then click **Go**. This displays the list of groups. Click **Modify** for the group you want to make changes to. This displays the Music On Hold Group page.

Make your changes, then click **Modify Group** to save your changes and go back to the Music on Hold page. A success message indicates that the group has been modified.

### **Deleting a Group**

To delete a group, select *All Groups* from the **View** dropdown, and click **Go**. This displays the list of groups. Click **Delete** for the group you want to delete. Be sure that you do indeed want to delete the group, then click **Yes, Delete Group**. This redisplays the Music On Hold main page, and a message indicates that your group has been deleted.

### **Uploading a Song**

To upload a song, select *Upload Song* from the **Add New** dropdown, then click **Go**. This displays the MOH Upload page.

Click **Upload Song** to upload the new file. The MOH Main page is displayed again, with a success (or failure) message, and the new song is available in the song list.



### Song to Upload

Click **Browse**, then use the dialog box to locate the file on your machine. Select the file that you want to upload, then click **OK**. This closes the dialog box and puts the name of the file in the **Song to Upload** box. Be sure that you are uploading an .ogg file (see Adding a Codec if you want to use files of another type).

#### **Part of Group**

Select a group from the dropdown. (You can always move or copy the song to a different group.)

### **Moving or Copying Songs**

To move song files to a different group, check the box for each song that you want to move, select the new group from the **Move To** dropdown, then click **Move To**. The song list is refreshed and reflects this change.

To copy song files to a different group, check the box for each song that you want to copy, select the new group from the **Copy To** dropdown, then click **Copy To**. The song list is refreshed and reflects this change.

### **Deleting Songs**

To delete song files, check the box for each song that you want to delete, then click **Delete**. Review the list of songs that will be deleted, be sure that you do indeed want to delete them, then click **Yes, Delete Files**. The files are permanently deleted, the MOH Main page is displayed, and a success message indicates that the files were deleted.

### **Adding a Codec**

To add a codec, select *Codec* from the **Add New** dropdown, then click **Go**. This displays the **Upload a New Music On Hold Codec** page.

### **Upload Codec Package**

Click **Browse**, then use the dialog box to locate the file on your machine. Select the file that you want to upload, then click **OK**. This closes the dialog box and puts the name of the file in the **Upload Codec Package** box.

Click **Install**. The codec is installed on Switchvox and the MOH Main page is displayed with a message indicating the codec was successfully installed.

**Note**: Because of licensing restrictions, the MP3 codec cannot be included with Switchvox. If you are already licensed to use the MP3 format, or are otherwise allowed to playback MP3 files, you may download the proper codec on this page. When the download is complete, then you can use this page to upload the file and install it.

### **Deleting a Codec**

To delete a codec, select *All Codecs* from the **View** dropdown, then click **Go**. All of the available codecs are displayed. Click **Delete** for the codec that you want to delete. Be sure that you do indeed want to delete the codec, then click **Yes, Delete Codec**. The MOH Main page is displayed, with a message indicating that the codec was deleted.

Note: You cannot delete the default .ogg codec.

# **IVR** Editor

Interactive Voice Response, or IVR, is an automated menu system (also known as an auto-attendant) that you can program to greet and interact with your callers. An IVR can provide information to the caller, collect information from the caller, and send the caller to the right extension.

### Select Starting IVR Menu

Select an existing IVR menu, or Create New IVR Menu.

### **IVR Tree**

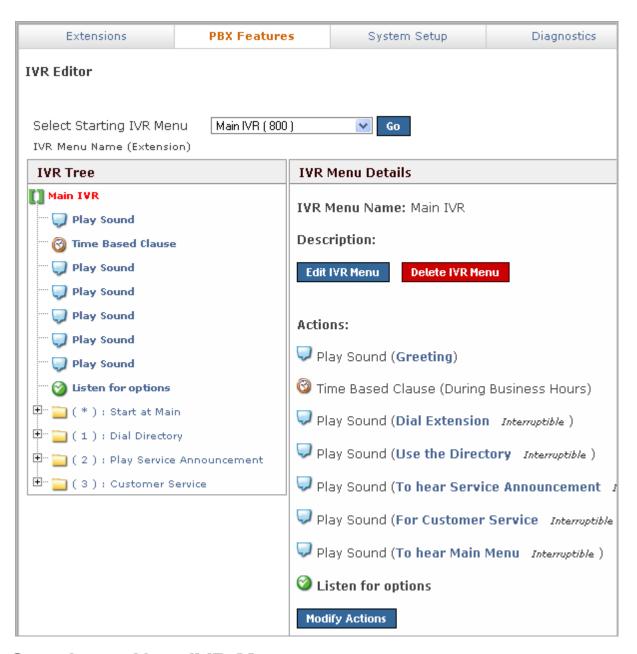
This displays an overview of the selected menu, in tree form. The *Actions* for the IVR are listed at the top, with various icons to indicate what sort of action is taking place. The last action in the list is always *Listen for options*, which cannot be modified or deleted. That is the action that handles the IVR's Options.

The *Options* that a caller can choose are listed at the bottom, with the folder icon. Options are really just an entry point into another IVR, so if you click the *plus* next to an Option folder it expands and displays the actions in that IVR menu.

Click on any action to see the details of that action in the IVR Menu Details section. Click on any folder or option-name to see the menu details in the IVR Menu Details section.

#### **IVR Menu Details**

This normally displays the name and description of the menu, followed by a list of the actions and a list of the options. Or, if you clicked on an action from the IVR Tree, this displays the details for that action.



# Creating a New IVR Menu

Select *Create New IVR Menu* from the dropdown and click **Go**. This displays the Create A New IVR Menu page.

Enter a unique name and a description, then click **Create IVR Menu**. This redisplays the IVR Editor main page with a message indicating the new IVR Menu was created, and the new menu is displayed in the IVR Tree and IVR Menu Details.

To create your main IVR, it's best to plan the entire IVR experience that you want to offer to your callers, then start from the end and work towards the beginning:

 $\circ~$  Determine all of the recorded sounds you want to play for the caller (instructions, information, etc.), and create those sounds in the Sound Manager.

- Determine which extensions the caller might end up being transferred to, and create an IVR menu for each one. (Those menus would have a 'Dial Extension' action.)
- Determine what factors might change your IVR, and build an IVR menu to handle each situation. For example, if your IVR behaves differently during regular business hours than after hours, or if it behaves differently based on a zip code the caller provides, you would create an IVR menu for each situation.
- Once you have your sounds and a set of 'component' IVRs built, then you can build the main IVR menu that answers your calls.

When you are done building your main IVR, you need to create an extension that reaches it. For details, see IVR (Interactive Voice Response) Extensions in "Extensions." Also, if you want all calls from your phone service provider to go to your main IVR, you can set that provider's *Callback Extension* to the extension of your main IVR.

# **Adding & Modifying IVR Actions**

To add an action to an IVR, click **Modify Actions** in IVR Menu Details. This displays the Current Actions page. Click **Add New Action**. Select an action from the dropdown. When you make a selection, the Action Description section shows a short description of the selected action. When you have selected an action, then click **Next**.

There are many actions that can be added to an IVR Menu. Most actions have a configuration page where you can specify the details of the action, then save your changes.

### **Interruptible Actions**

As you build your IVRs, you'll notice that some IVR actions can be interrupted if the caller presses a key on his or her keypad. This can offer a better caller experience, but you must plan for it carefully because it has repercussions throughout your IVRs. If you want to allow interruptions, be careful with how the actions are ordered in your IVR, and where the interruptions are allowed.

Here are the rules that Switchvox follows for interruptions in IVRs:

- 1. If an action is not interruptible, all keys pressed *before* or *during* that action are dropped. If the caller pressed keys during previous interruptible actions, those keys are lost.
- 2. If an action is interruptible, the caller can press any key to interrupt, and the IVR remembers that one key.
- 3. If multiple subsequent actions are interruptible, the IVR remembers each key that was pressed to interrupt each action.
- 4. When an IVR has collected one or more keys through a series of interruptible actions, it uses those keys (in the order they were pressed), in the following manner:
  - 1. If there is an upcoming *Record Digits* action, with no **un-**interruptible action in the way, the keys are reserved for the *Record Digits* action. They will not be used to dial an extension or to select an option.
  - 2. If Enable Extension Dialing is on, and the first key pressed matches an extension number, Switchvox pauses for 3 seconds to see if the caller will press a second key. If a second key is pressed and a valid extension is dialed, the call is transferred. However, if two or more keys are pressed that do not match a valid extension, then the key press is considered an Invalid Option (see Adding & Modifying IVR Options).
  - 3. If *Enable Extension Dialing* is not on, or the caller presses a key that does not match an extension, or the caller does not press a second key within 3 seconds, the option that matches that key is selected and the caller is sent to that option's IVR menu.

### **Play Sound**

This action plays a sound from the Sound Manager to the caller. If you want to play sounds recorded by the caller, use the action 'Play Recorded Sound.'

#### **Folder**

Select the Sound Manager folder where this sound is located.

### Language

Select the language for this sound. *Current Language* is the default and is the language that was previously set in the IVR. The default is English.

### Sound to Play

Select the sound to play.

### **Play Sound: Advanced Options**

If you want to decide which sound to play based on a sound ID stored in a variable, click on the **Show Advanced Options** link. This lets you select variables that have been previously created using the 'Set Variable' action. Make sure that the value in the variable is a sound ID from the Sound Manager, or this action will not be able to play the sound.

### **Record Sound**

This action records a sound from the caller, and saves it in a variable. The sound variable can be used later on in the IVR menu for actions such as playing or uploading.

When a caller reaches a Record Sound action, Switchvox plays a beep and then begins recording. Recording stops when the caller presses the "#" key, hangs up, has recorded too much silence, or has reached the total minutes of recording time.

### Stop Recording after sections of silence

Enter the number of seconds of silence allowed before recording stops.

### Stop Recording after total minutes

Enter the maximum number of minutes to record.

### **Recorded Sound Variable Name**

Select a previously defined sound variable, or enter the name of the new variable.

### **Play Recorded Sound**

This action plays a previously recorded sound to the caller.

### **Recorded Sound to Playback**

Select a sound from the dropdown. The dropdown includes all sound variables that have been previously set in this IVR.

### Send an Email

This action sends an email message. You can include IVR and System Variables in the message, and you can attach a sound file.

### **Email Address to Send To**

Select a variable that contains the email address, or enter an address in the text box.

#### From Email Address

Select a variable that contains the email address, or enter an address in the text box.

### **Email Subject**

Enter a subject in the text box. You can use IVR and System Variables to include text that is specific to this call.

### **Email Body**

Enter the text for the body of the email message. You can use IVR and System Variables to include text that is specific to this call.

#### **Sound Attachments**

Select a sound from the dropdown, then click Attach Sound to Email. The sound name is displayed, with a **remove** button.

### Send Recorded Sound to Voicemail

This action sends a sound to a folder in an extension owner's voicemail Mailbox.

#### **Extension To Send Sound To**

Select a previously defined variable, or enter an extension in the text field.

### Variable of Sound to Send

Select a previously set sound variable.

### Voicemail Folder to Place Sound In

Select the voicemail Mailbox folder that you want to send the sound to.

### Store Recorded Sound in Sound Manager

This action stores a sound in the Sound Manager. This overwrites an existing user sound file in the Sound Manager (it cannot overwrite a system sound). This action can be used to re-record stored sounds, such as information updates.

**CAUTION:** When the IVR reaches this action, it immediately stores the chosen sound in the Sound Manager, overwriting the existing sound. You probably want to do something to limit the access to this action (i.e., require a pass code first).

#### **Folder**

Select the Sound Manager folder that the sound should be stored in.

#### Language

Select the Sound Manager language that the sound should be stored in.

### Sound to Store as

Select the Sound Manager sound file that should be over-written. This dropdown only includes the sounds available in the folder and language that you select. You cannot select system sounds.

#### **Recorded Sound to Store**

Select a previously defined sound variable that should be stored in the Sound Manager as specified above.

### **Upload Recorded Sound**

This action uploads a sound to a remote Web server. Two things are required: the URL of the remote upload processing script, and an existing IVR sound variable.

You can enter a 'static' URL that is always the same, or you can create a 'dynamic' URL by entering variables so that the system builds the appropriate URL.

A dynamic URL can include IVR Action variables or system variables. When using a variable, make sure that a previous action has assigned a value to that variable, otherwise nothing will be inserted into the URL. To add a variable to the URL, select it from the appropriate **Possible Variables** dropdown, then click **Append**. The

variable will appear in the URL field. When the URL is requested, each *%VARIABLE\_NAME%* is replaced with the value of that variable.

#### **URL**

Enter the URL to the remote processing script. Use the **Possible Variables** options to append variables to the URL.

#### Sound to Send

Select a previously defined sound variable from the dropdown.

#### **Timeout in Seconds**

Select the number of seconds to try connecting to the URL. If Switchvox cannot connect with the remote script within that time, Switchvox will quit trying and go on to the next action in this IVR.

### **Merge Sounds**

This action merges multiple sounds together into one sound variable.

### **Sounds To Merge**

Use the **Existing Sounds** and **Sound Variables** options to put sounds into the boxes numbered 1 through 10. The sounds are merged in this order.

### Variable Name for Merged Sound

Enter a unique name for this sound variable (one sound that includes all of the above sounds merged).

### **Play Sound From URL**

This action downloads a WAV sound file from a URL, then plays the sound.

You can enter a "static" URL that is always the same, or you can create a "dynamic" URL by entering variables so that the system builds the appropriate URL.

A dynamic URL can include IVR Action variables or system variables. When using a variable, make sure that a previous action has assigned a value to that variable, otherwise nothing will be inserted into the URL. To add a variable to the URL, select it from the appropriate **Possible Variables** dropdown, then click **Append**. The variable will appear in the URL field. When the URL is requested, each *%VARIABLE\_NAME%* is replaced with the value of that variable.

#### URL

Enter the URL to the remote processing script. Use the **Possible Variables** options to append variables to the URL.

### **Timeout in Seconds**

Select the number of seconds to try connecting to the URL. If Switchvox cannot connect with the remote script within that time, Switchvox will quit trying and go on to the next action in this IVR.

### This sound is interruptible

Check this box if this is interruptible. For details, see Interruptible Actions.

### **Gate Keeper**

This action places a 'gatekeeper' in the IVR that counts how many times a caller has traveled through the gate. This is useful to keep callers from getting stuck in infinite loops, or to trap callers that are trying to abuse the system. Use a **Conditional Clause** action to check the count, and send the caller to another IVR when they exceed the limit.

### **Gate Variable Name**

Enter a unique name for this gatekeeper.

### **Conditional Clause**

This action is a conditional statement used to send callers to another IVR menu if the statement is true. A Conditional Clause action has three parts: the variable to evaluate, the criteria to use, and a value to compare against.

If your variable contains text, you can only use the criteria *Equals* or *Does Not Equal*. The other criteria are only for use with a variable that contains numbers. Also, the text you enter is case sensitive, so 'RINGING' does not equal 'ringing.'

If your variable contains numbers, you can use any of the criteria.

#### If

Select the variable to evaluate, the criteria to use, and a value to compare against.

#### Then Go To IVR Menu

Select the IVR menu the caller should be sent to if the above statement is true.

### **At Entry Point**

Select an action that is in the above IVR menu. You do not have to start at the beginning of the IVR.

### **Time Based Clause**

This action checks the current date, day, and time and matches it against the Time Frame. If the current time is within the Time Frame, then the caller is sent to another IVR. (See Time Frames for instructions on creating and modifying Time Frames.)

For example, if the current time is within your Time Frame *Outside Business Hours*, you can send callers to the IVR that plays a sound asking the caller to leave a message or call back during business hours.

### If the current time is within the time range

Select the Time Frame to compare against.

#### Then Go To IVR Menu

Select the IVR menu the caller should be sent to if the above statement is true.

### **At Entry Point**

Select an action that is in the above IVR menu. You do not have to start at the beginning of the IVR.

### **Change Language**

This action changes the 'Call Language' of this call, and any sounds played after this action are played in this language (or the default language if there is no sound available for the call language).

### Language to Change To

Select a language from the dropdown. The dropdown includes the languages that are available in the Sound Manager.

### Alter Caller ID

This action alters the caller's callerID name or number. You can *Prepend*, *Append*, or *Replace* the name or number with your information. This is useful to include information that a caller has entered, or to show which IVR options the caller had selected.

### How do you want to modify the Caller ID?

Select Prepend, Append, or Replace the callerID with your information.

### Which part of the Caller ID do you want to modify?

Select Number or Name, depending on which part you want to modify.

### Use this text to modify the Caller ID

Enter the text that you want, or use the text from a previously set variable.

### Wait

This action adds a delay before the next action.

#### Wait time in seconds

Enter the number of seconds to wait before continuing to the next action.

### This wait is interruptible

Check this box if this is interruptible. For details, see Interruptible Actions.

### **Give Busy Signal**

This action gives the caller a busy signal for the number of seconds specified. The busy signal is not interruptible, but it does not hang up the call. After it plays, the IVR continues as normal.

### Signal Timeout in Seconds

Select the number of seconds that the busy signal plays.

### Hang Up

This action hangs up the call. There are no settings for this action, just click **Save Settings** to save this action and go back to the Current Actions page.

### **Dial Extension**

This action dials an extension. The caller leaves the IVR and rings that extension. At that point, the extension's Call Rules apply.

### **Extension to Dial**

Select a previously set variable, or enter an extension.

### Send to Voicemail

This action sends the caller to the voicemail box of an extension.

#### **Extension to Dial**

Select a previously set variable, or enter an extension.

### **Send to External Number**

This action sends a caller to an external number.

### **Number To Dial**

Select a previously set variable, or enter an external phone number.

### Outgoing call rules for this number

Check the boxes to allow or deny use of each outgoing call rule (just as you would for Switchvox extensions). See Outgoing Call Rules for details.

WARNING: Use this action with extreme caution because it can be abused by callers.

### Go To IVR Menu / Action

This action sends a caller to another IVR menu. This can also be used to jump or return to an action in the current IVR menu.

### Go To IVR Menu

Select the IVR menu to send the caller to.

### **IVR Menu Entry Point**

Select an Action that is in the IVR menu. (You don't have to start at the beginning of the menu.)

### **Record Digits**

This action records digits (the DTMF tones) pressed by the caller. The digits are saved in a variable.

### Beep before recording

Select Yes if you want a 'beep' sound to play before recording begins. This lets the caller know they should start pressing buttons.

### Stop recording after this many digits

Enter a number from 0 to 255. Recording stops when the caller has pressed the maximum number of digits, or the pound (#) key.

If this is 0, the recording continues until the caller presses #, or presses 255 digits.

### **Recorded Digits Variable Name**

Select a previously set variable, or enter a unique name for this variable.

### Say Digits/Letters

This action plays digits and/or letters to the caller.

Each digit or letter is read individually. For example, "one four four five," or "L I K E." If you want to say "one thousand four hundred and forty five" instead, use the *Say A Number* action. (If you want to say words, use a sound variable.)

### Digits to Say

Select a variable that contains the digits and/or letters to say, or enter the digits and/or letters in the text field.

### Say A Number

This action plays a number to the caller.

The number 1445 is read as "one thousand four hundred and forty five." If you want it to say "one four four five," use the Say Digits/Letters action.

### **Number to Say**

Select a variable that contains the number to say, or enter the number in the field.

### Say Date/Time

This action plays a date and time to the caller. This can be the current date and time, a specific date and time, or the date and time contained in a variable.

#### Say the current date and time when the call enters this action

Check the box Yes if you want this to be true.

### Say a predefined date

Enter the date to be played. Enter a time also, or leave it blank.

### Say date from a variable

Select a variable from the dropdown. This variable must be in epoch time.

### Format to Say the Date in

Select a format for the date to be played.

### **Play DTMF Tones**

This action plays DTMF tones.

### **DTMF** Digits to Play

Select a variable that contains each of the DTMF digits to play, or enter the digits in the field.

### Play each digit for this many milliseconds

Select the number of milliseconds that you want each DTMF tone to play.

### Wait this many milliseconds between digits

Select the number of milliseconds that you want to wait before the next tone is played.

### **Perform Math**

This action performs a mathematical operation on two IVR variables, and stores the result in a new variable.

### Choose the variables for your operands and the desired operation

Select the first variable, the operation, and the second variable.

### Variable Name for Result

Enter a unique name for this variable.

### **Set Variable**

This action sets a variable.

IVR Variables are temporary variables that are set during a call, and are only available to that call. During the call, the variable and its value can be accessed from all IVR menus.

**Note:** If you want to set an IVR Variable where the value can be accessed across calls, see Set Global Variable.

#### Variable to set

Select a previously existing variable from the dropdown, or enter a unique name in the text field.

### Set to value of

Select a System Variable, or enter a value in the field. If you select a System Variable, the value of that variable is now contained in this variable.

### Set Global Variable

This action sets a Global Variable.

Global Variables are variables that can be accessed from all IVR Menus and whose values persist between calls. If a Global Variable's value is changed by an IVR Menu, all IVR Menus and all calls will have access to that value.

### **Global Variable to Set**

Select a Global Variable from the dropdown, or enter a unique name for this Global Variable.

### Set to value of Variable

Select a variable from the dropdown. The value contained in the variable is now also contained in this Global Variable.

### Get Global Variable

This action stores the value of a Global Variable into a normal variable. If you change the value of this normal variable, it does not affect the original Global Variable.

Global Variables are variables that can be accessed from all IVR Menus and whose values persist between calls. If a Global Variable's value is changed by an IVR Menu, all IVR Menus and all calls will have access to that value.

### Variable to set

Select a previously existing variable from the dropdown, or enter a unique name in the text field.

#### Set to value of Global Variable

Select the Global Variable that contains the value that you want.

### **Concatenate Variables**

This action concatenates multiple variables together, then saves the result in a new variable.

### **Variable Concatenation**

Enter a representation of the variable you want to create. Add variables to this field using the **Possible Variables** options, or enter text. Variables must be enclosed with %.

#### Variable Name for Concatenation

Enter a unique name for this variable.

### Send Call Values to URL

This action passes call values to the URL of a processing script. That script can return XML containing a set of values, which Switchvox can save in variables.

### **URL**

The URL can include IVR variables or System variables. When using an IVR variable, make sure that a previous action has assigned a value to that variable, otherwise nothing will be inserted into the URL.

To add a variable to the URL, select it from the appropriate **Possible Variables** dropdown, then click **Append**. The variable will appear in the URL field. When the URL is requested, each *%VARIABLE\_NAME%* is replaced with the value of that variable. For example:

```
http://www.mybiz.com/cgi-bin/app.cgi?act=%ACCOUNT%&pin=%PIN%
```

This script might verify the account number and pin, then return a set of XML that includes the account information.

### XML and Returned Values

If your script returns a set of XML with values that you want to save and use, you must enter the variable names in the boxes labeled 1 through 8. After Switchvox sends the call values to your script, and collects the return set of variables, those variables and their values are available for use on this call, in this IVR menu.

The following is a sample DTD for the XML response from your script:

```
<!-- Switchvox PBX IVR xml DTD -->
<?xml version="1.0"?>
<!ELEMENT ivr_info (variable+)>
<!ELEMENT variable (#PCDATA)
<!ATTLIST variable name CDATA #REQUIRED>
```

The following is a sample XML response from your script:

```
<value>John</value>
</variable>
</variables>
</ivr_info>
</result>
</response>
```

### **Get Extension Status**

This action gets the status of the extension, and saves the result in a new variable. This is useful if you want to keep control of the call. For example, if you use 'Get Extension Status' and the extension is busy, you could try someone else. If you used 'Dial Extension' and the extension is busy, Call Rules might send the caller to voicemail instead.

Get Extension Status returns one of the following results:

- o IDLE
- INUSE
- BUSY
- o INVALID
- UNAVAILABLE
- RINGING

These are case sensitive, so IDLE does not equal idle.

### **Extension To Check Status On**

Select a variable that contains an extension, or enter an extension number in the field. Click on the people icon to look up the extension that you want.

#### Variable Name for Status Result

Enter a unique name for this variable.

### **Get Extension Type**

This action gets the type of the extension, and saves the result in a new variable.

**Get Extension Type** returns one of the following results:

- zap
- virtual
- $\circ$  ivr
- o sip
- conference
- o group\_pickup
- o intercom
- o voicemail
- callpark
- agentcallbacklogoff
- o call\_queue
- o agentcallbacklogin
- autodial
- directory
- o simple\_conference

feature code

These are case sensitive, so ZAP does not equal zap.

### **Extension To Get Type Of**

Select a variable that contains an extension, or enter an extension number in the field. Click the people icon to look up the extension that you want.

### Variable Name for Result

Enter a unique name for this variable.

### **Check User Password**

This action evaluates the extension's password with the password contained in a variable, and saves the result in a new variable. This action returns a 0 (zero), which indicates failure, or 1 (one), which indicates success.

#### Choose variable where the extension is stored

Select the variable that contains the extension.

### Choose variable where the password is stored

Select the variable that contains the password.

#### Variable Name for Result

Enter a unique name for this variable.

### **Set Distinctive Ring Hint**

A Distinctive Ring Hint is used by an extension's Distinctive Ring Rules. These let an extension-owner set up different 'Ring Actions' depending on the call.

To set a distinctive ring hint:

Use the value of an existing variable

The hint is the value of the variable, and this value must be previously set in the IVR actions. Otherwise, the hint is not available to the recipient of the call.

o Enter a value in the text field

The hint is exactly what you enter in this field.

Note: Distinctive Ring Rules are only available on Polycom phones.

### **Remove Distinctive Ring Hint**

A Distinctive Ring Hint is used by an extension's Distinctive Ring Rules. These let an extension-owner set up different 'Ring Actions' depending on the call.

To remove a distinctive ring hint:

- Use the value of an existing variable
- Select a variable from the dropdown. The hint that is removed is the value of this variable.
- Select an existing Ring Hint from the dropdown
- Select a Hint from the dropdown. If you do not have any IVR actions of **Set Distinctive Ringtone Hint**, this dropdown is not available.
- Clear all Distinctive Ring Hints
- Check this box if you want to remove all distinctive ring hints that were previously set in this IVR.

Note: Distinctive Ring Rules are only available on Polycom phones.

# **Adding & Modifying IVR Options**

To add an option, click Modify Options in the IVR Menu Details. This displays the Current Options page.

To add a new option, click **Add A New Option**. Make your selections, then click **Create New Option** to save your changes and go back to the Current Options page.

### **Option Number**

Select a number, or the \* key. This what the caller presses to select this option.

#### **New IVR Menu**

Select the IVR menu the caller is sent to when the Option Number is pressed.

### **IVR Menu Entry Point**

Select an Action that is in the IVR menu. (You don't have to start at the beginning of the menu.)

### **Enable Extension Dialing**

Let the caller dial an extension number, then ring that number. Use this feature for an IVR that says, "if you know your party's extension, please dial it now."

Normally IVR Menus only allow single digit options. However, if this setting is checked, then IVR callers (in this IVR Menu only) can dial extensions. If the IVR caller enters a valid extension, Switchvox rings the extension. If the caller enters an invalid extension, then it is considered an "invalid option," and is handled according to that setting.

Only Call Queues, Phone Extensions, Directory Extensions, Voicemail Access extensions, Virtual Extensions, and other IVR extensions can be dialed. Callers will not be able to dial any other types of extensions.

Extensions must have **This extension can be dialed from the IVR** checked. Even though it may be a valid extension, if this is not checked then it is considered an 'Invalid Option' for a caller. (See Advanced Options.)

Options and extension dialing can coexist in the same IVR Menu. However, if you have an option in the menu (i.e.; 1) that starts with the same number as an extension (i.e.; 105) then the caller experiences a delay. When he presses '1', Switchvox waits a moment to see whether he might also press '05'. If he doesn't, Switchvox proceeds to option 1.

### **Invalid Option Settings**

Specify what should happen when a caller presses a key for an option that doesn't exist. Select a sound to play, then an IVR menu and entry point (you don't have to start from the beginning of the menu).

### **Option Timeout Settings**

Specify what should happen if a caller fails to select any options within a certain amount of time. Set the number of seconds to give the caller, a sound to play, and an IVR menu and entry point (you don't have to start from the beginning of the menu).

#### **Retry Settings**

Specify what should happen if a caller chooses an invalid option or times out several times. Select the number of times to allow, and an IVR menu and entry point (you don't have to start from the beginning of the menu).

# **Sound Manager**

The Sound Manager lets you create, upload, modify, and delete all of the sounds in Switchvox other than Music On Hold. It also lets you organize your sounds in folders, and manage the same sounds that are in

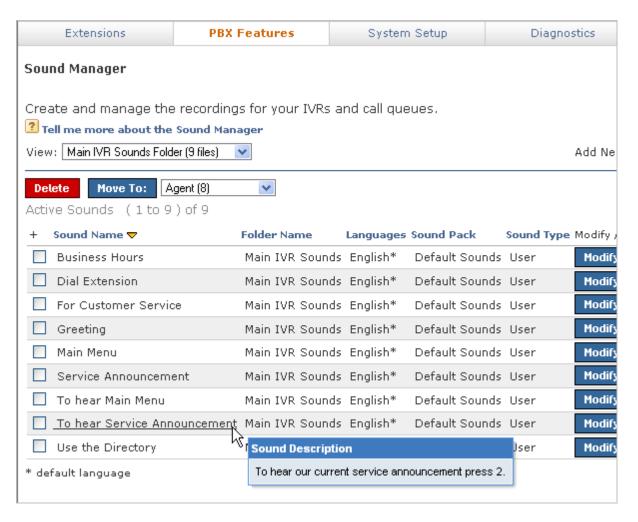
different languages.

Sounds can be uploaded from your computer, or recorded over any phone that's connected to Switchvox.

Use the main page of the Sound Manager to see and manage your sounds:

- View all sounds
- View all of the sound packs, folders, or languages
- View all sounds within a folder
- Add a new sound, sound pack, folder, or language

**Note:** You are not allowed to delete, move, or modify system sound information or system sound files in the English language.

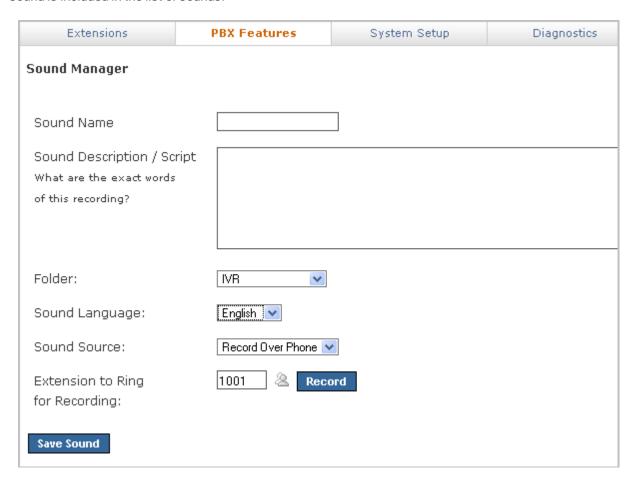


# Adding a Sound

To add new language sound-file to a sound that already exists, follow the instructions in Modifying/Playing a Sound. For example, if you already have a sound for 'hi' in English, and you want to create the Spanish sound 'hola,' then you would modify the English 'hi' by adding the Spanish sound. That lets you keep the same sounds in different languages grouped together.

To add a completely new sound, select Sound from the Add New dropdown and click Go.

When the settings are correct, click **Save Sound**. (If you are uploading a file, the transfer begins). The system redisplays the Sound Manager main page with a message that a new sound has been added, and the new sound is included in the list of sounds.



### **Sound Name**

Enter a unique name for this sound.

### Sound Description/Script

Enter the words of this recording. This is displayed when you mouse over the sound name in the list of sounds.

#### **Folder**

Select a folder for this sound.

### **Sound Language**

Select the language that is used in this sound.

### **Sound Source**

Select Record Over Phone or Upload File.

If you chose *Record Over Phone*, enter the extension to ring, then click **Record**. Make your recording, then hang up the phone or press #.

If you chose *Upload File*, browse your computer to identify the sound file.

# Modifying/Playing a Sound

To play or modify a sound, click its **Modify/Play** button. This displays the Update a Sound page. This page shows the sound, the available languages, and the sound's status, name, sound pack, description, and actions allowed.

When the settings are correct, click **Save Sound**. (If you are uploading a file, the transfer begins). The system redisplays the Sound Manager main page with a message that a new sound has been added, and the new sound is included in the list of sounds.

#### **Play**

Plays the sound on your computer. You must have a sound player application installed to hear the sound. For sounds in the gsm format, Quicktime by Apple is a good player.

### **Play To Phone**

Displays the **Play To Phone** dialog box. Enter the extension to ring, then click **Go**. This rings the extension, then plays the message when you pick up the phone.

### Modify

Displays the Sound Details page. Make your changes, then click **Update Sound File**. The Sound Manager main page is displayed with a message indicating that the sound was modified.

**WARNING:** If you are uploading or re-recording the sound file, the existing file is replaced. You will not be able to retrieve the old version of the file.

#### **Record Over Phone**

Displays the Sound Details page. Enter the sound name and description, and an extension to ring. Click **Record**. Make your recording, then hang up the phone or press **#.** 

### **Upload File**

Displays the Sound Details page. Enter the sound name and description, and browse your computer to identify the sound file to upload.



# **Deleting a Sound**

In the sounds list on the main page, check the boxes for the sounds that you want to delete, then click **Delete** (at the top of the song list). Be sure that you do indeed want to delete these sounds, then click **Yes, Delete Sounds**. The main page is displayed again, with a message that the sound has been deleted.

**Note:** If a sound box cannot be checked, it is because it cannot be deleted. The basic set of system sounds cannot be deleted.

# Moving a Sound into a Folder

In the sounds list, check the boxes that you want to move, select the folder the sound belongs in (the dropdown at the top of the song list), then click **Move To**. The Sound Manager main page is displayed with a message that the sound has been moved. If you have used the sound in IVRs or call queues, Switchvox now accesses the sound from its new folder.

**Note:** If a sound box cannot be checked, it is because it cannot be moved. The basic set of system sounds cannot be moved.

### **About Sound Packs**

Sound packs are pre-configured packages that you can add to Switchvox and use in place of the pre-installed default English sound pack. Only officially certified sound packs can be installed on Switchvox.

# **Managing Folders**

Sounds are organized in folders. Switchvox comes preconfigured with several folders, which cannot be deleted. Typically, the folder name indicates the type of sounds it contains. Any new folders should be named accordingly.

# Adding a New Folder

To create a new folder, select *Folder* from the **Add New** dropdown, then click **Go**. Enter a unique name and a description for this folder, then click **Create Sound Folder**. This displays the Sound Manager main page with a message that a new folder has been created, and the new folder is included in the folders list.

# Modifying a Folder

To modify a folder, click its **Modify** button. Make your changes, then click **Modify Sound Folder**. The Sound Manager main page is displayed with a message that the folder has been modified.

# **Deleting a Folder**

To delete a folder, select its **Delete** button. Be sure that you do indeed want to delete this folder and all of the sounds in it, then click **Yes, Delete Folder**. The folder and all sounds in it are permanently removed, and the Sound Manager page is displayed with a message that the folder has been deleted.

# **Managing Languages**

Switchvox supports multiple languages. Each sound in the Sound Manager can be uploaded or recorded in all the available languages. Your sounds 'hi,' 'hola,' and 'ciao' sounds are grouped together, because they say the same thing to the caller.

The pre-configured default language is English, and there is a set of system files tagged English.

# Adding a New Language

To add a new language, select *Language* from the **Add New** dropdown. Enter the language name, then click **Create Language**. The Sound Manager main page is displayed, with a message that a new language has been added.

# Modifying a Language

To modify a language, click its **Modify** button. Make your changes, then click **Modify Language**. The Sound Manager main page is displayed, with a message that the language was modified.

# **Deleting a Language**

To delete a language, click its **Delete** button. Be sure that you do indeed want to delete this language and all of the sounds associated with it, then click **Yes, Delete Language**. The Sound Manager main page is displayed with a message that the language was deleted.

# Setting a Default Language

To set a language as the delete language, click its **Set as Default** button. The page is refreshed with a message that the sound pack was successfully activated. Now the sounds labeled with that language are played. If there is no sound available in that language, Switchvox uses the English system sound-file.

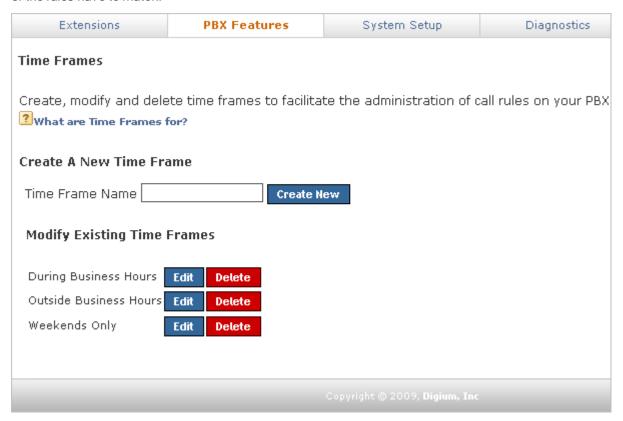
**WARNING:** Changing the Default Language automatically causes the Switchvox software to restart. All current phone connections will be dropped, and members will be logged out of all call queues.

# **Time Frames**

Switchvox can operate differently based on the date, day, and time. You can set up IVR Menus and Call Rules to check the Time Frames, and then act accordingly.

There are several predefined time frames that you can use: *Business Hours*, *Outside Business Hours*, and *Weekends Only*. You can edit or delete these time frames, or create new ones.

A time frame is considered valid when the current date and time match any one of a time frame's rules. Not all of the rules have to match.



# **Creating a New Time Frame**

To create a new time frame, enter a unique name and click **Create New**. Enter any appropriate Start and End Dates, Start and End Days, and Start and End times, then click **Add**.

Continue adding rules until you have completed the Time Frame definition.

The following are sample time frame definitions:

Start Date	End Date	Start Day	End Day	Start Time	End Time
-	-	Monday	Monday	-	-
-	-	Wednesday	Wednesday	-	-

Mondays 12:01AM to 12:00PM, and Wednesdays 12:01AM to 12:00PM

Start Date	End Date	Start Day	End Day	Start Time	End Time
-	-	-	-	3:00 AM	6:00 AM

# **Early Morning**

Start Date	End Date	Start Day	End Day	Start Time	End Time
07/01/2009	07/07/2009	-	-	-	-

# The First Week in July 2009

Start Date	End Date	Start Day	End Day	Start Time	End Time
01/21/2009	-	-	-	-	-

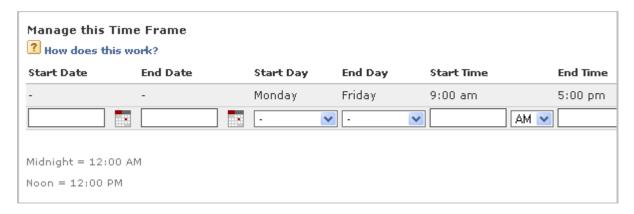
After January 21, 2009

Start Date	End Date	Start Day	End Day	Start Time	End Time
02/01/2009	02/28/2009	Monday	Friday	6:00 AM	7:00 AM

Weekdays in February, 2009 between 6:00 and 7:00 in the morning

# **Editing a Time Frame**

To change a time frame, click its **Edit** button. This displays the Manage This Time Frame page. Create a new rule, or multiple rules, and delete any rules that you do not want to include in this time frame. (You cannot edit a rule.)



# **Deleting a Time Frame**

To delete a time frame, click its **Delete** button. Be sure that you do indeed want to delete this time frame, then click **Yes, Delete Time Frame**. The Time Frames main page is deleted with a message that the time frame was deleted.

Before you delete a Time Frame, be sure you do not have any IVR Menus or Call Rules that depend on it.

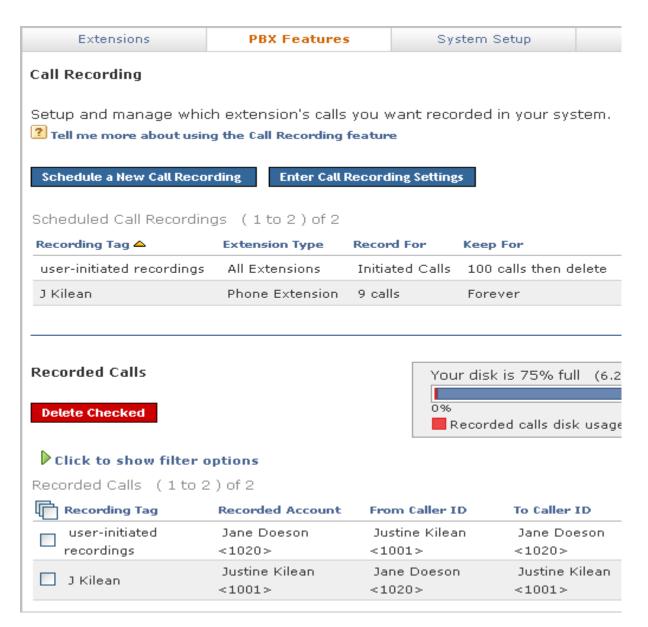
# **Call Recording**

This lets you set up rules for recording calls. Any calls that match these rules are recorded and listed on this page. You can view, listen to, and download recorded calls.

There is one default call recording rule that Switchvox automatically follows, called *user-initiated recordings*. When a user records a call (his own call or another person's), those recordings are included here. These are the same recordings that the user has in his Mailbox, so if you or the user deletes one of these recordings, it is no longer available in either area. However, if the user forwarded the recording to another Switchvox user, that version is unchanged. Users can record calls using the Switchboard, if he or she has permission.

# Viewing & Playing Recorded Calls

The Recorded Calls list shows the information for each call, and lets you play the call. When you click **Play**, you can choose to play the call with your audio player, or download the file (these options may depend on your Web browser's settings).



# **Filter Options**

To view and listen to a specific set of calls, click on the **Click to show filter options** link. The Apply Filter dropdown lets you filter the recordings list.

Click **Run Filter** to limit the Recorded Calls to the calls that match your filter.

#### Date

Enter the surrounding dates of the time period that you would like to hear the recorded calls from.

## **Recording Tag**

Simply choose the recording tag of the recorded calls that you would like to hear from the dropdown.

#### **Duration**

Simply choose the time limit of the recorded calls you would like to hear from the dropdown.

#### From Caller ID

Enter the phone number, extension, or caller ID of the person who called you that you would like to hear the recorded calls from.

#### To Caller ID

Enter the phone number, extension, or caller ID of the person who you called, that you would like to hear the recorded calls from.

#### **Recorded Account**

In this filter option, you must choose either Phone Extensions, Call Queues or Extension Groups from the dropdown, and then choose which account from the lists that are displayed below the menu.

# **About Call Recording Settings**

You can back up your recorded calls to an external FTP server before the calls are deleted. This is highly recommended if you rely on call recordings.

On the main page, click **Enter Call Recording Settings**. Enter your FTP server settings.

When recorded calls are backed up to this FTP server, Switchvox automatically creates a separate folder on the FTP server for each recording rule. Each call recording is automatically stored in its respective folder. Each recorded call has two files, a WAV file and a XML file. The WAV file is the sound, and the XML is the data about the call.

The file names are based on the following structure: [Date]-[Time Stamp]-[From Caller ID]-[To Caller ID].

```
2007-04-06-15-00-41_7771021212_102.wav
2007-04-06-15-00-41_7771021212_102.xml
```

This sample call was made on April 6, 2007 at 3:00:41PM. A call from 7771021212 was received by extension 102.

The XML file includes the following data:

```
<recording_info>
<recording_tag>Steve Smith</recording_tag>
<recorded_call_id>1200</recorded_call_id>
<recorder_cid></recorder_cid>
<recorded_cid>Steve Smith &lt;102&gt;</recorded_cid>
<recorder_account_id></recorder_account_id>
<recorded_account_id>1101</recorded_account_id>
<from_account_id>0</from_account_id>
<from_caller_id>BuyIt &lt;7771021212&gt;</from_caller_id>
<to_account_id>1101</to_account_id>
<to_caller_id>Steve Smith&lt;102&gt;</to_caller_id>
<duration>316</duration>
<date_created_ts>2007-04-06 15:00:41.611474</date_created_ts>
<date_created_secs>1175896842</date_created_secs>
</recording info>
```

# Scheduling a New Call Recording

To schedule a new call recording, click Schedule a New Call Recording.

## Whose calls would you like to record?

Select one or more extensions, extension groups, or call queues. You cannot combine these types, but you may select many of each type.

# What type of calls would you like to record?

Check All Outgoing Calls, Incoming Direct Calls, and/or Incoming Queue Calls.

# How many calls would you like to record?

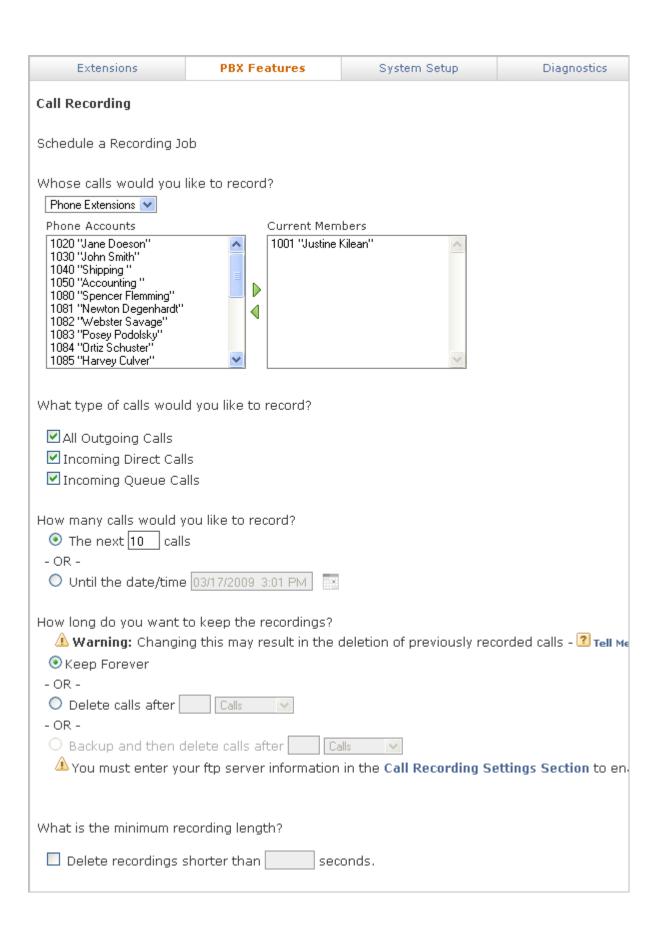
Set a specific number of calls, or the date and time you want the recordings to end.

# How long do you want to keep the recordings?

Set the storage instructions for the recordings. If you want to back up older recordings before you delete them, click the link to **Call Recording Settings Section** (see About Call Recording Settings). **IMPORTANT:** If you enter a number into the **Delete calls after** setting, Switchvox immediately deletes any older recordings to ensure that there are no more than that number of files on disk.

# What do you want to label these call recordings?

Enter a unique name for this scheduled call recording rule.



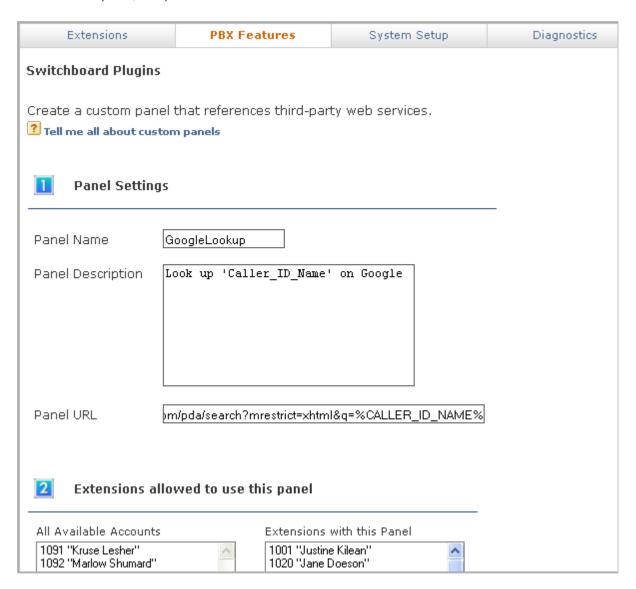
# **Deleting Recorded Calls**

To delete call recordings, check the box for each one that you want to delete, then click **Delete Checked** (at the top of the list). Be sure that you do indeed want to delete the recordings, then click **Yes, Delete**. If you are deleting 'user-initiated recordings,' the recordings are deleted from the user's Mailbox.

# **Switchboard Plugins**

Switchboard Plugins let you add powerful panels to the Switchboard. These panels let you exchange data with external Web applications, then display the results. The panels refresh each time a call comes in, showing caller-specific information.

For each panel, you can specify which phone-type extensions are allowed to use it. If an extension isn't allowed to use a panel, that panel isn't included in the Switchboard's Panel menu.



# Standard Panels

Standard panels provide integration with SugarCRM®, Salesforce.com, and Google® Maps.

# **SugarCRM®**

This panel provides integration with your SugarCRM® application.

# SugarCRM® Web URL

This is the URL to your SugarCRM® application's Web interface (e.g.,

http://crm.example.com/sugarcrm/). If you want external users to have access to this panel, then this URL must be accessible from the outside Internet. If your SugarCRM® application is running behind NAT, you may need to make it accessible by forwarding ports. HTTPS URLs and alternate ports are supported (e.g., https://public.example.com:8080/sugarcrm/).

## SugarCRM® SOAP URL

This is the URL to your SugarCRM® application's SOAP API (e.g.,

http://crm.example.com/sugarcrm/soap.php). This URL must be accessible from the outside Internet, so if your CRM application is running behind NAT, you may need to make it accessible by forwarding ports. HTTPS URLs and alternate ports are supported (e.g.,

https://public.example.com:8080/sugarcrm/soap.php).

### Extensions allowed to use this panel

Select one or many extensions on the left, and click the right-facing green icon to put those extensions into the box on the right. All of the extensions in the box on the right are allowed to use this panel.

### **Test your connection**

Enter your CRM system username and password, and the phone number for a contact in your application. Click **Test Settings** to see the results of your test.

This section is purely for testing your Switchvox's access to the CRM service, this Username and Password are not used outside of this test. In the Switchboard panel, each user must enter their own login information.

# Salesforce.com

This panel provides integration with your Salesforce.com account.

#### Extensions allowed to use this panel

Select one or many extensions on the left, and click the right-facing green icon to put those extensions into the box on the right. All of the extensions in the box on the right are allowed to use this panel.

### **Test your connection**

Enter your CRM system username and password, and the phone number for a contact in your application. Click **Test Settings** to see the results of your test.

This section is purely for testing your Switchvox's access to the CRM service, this Username and Password are not used outside of this test. In the Switchboard panel, each user must enter their own login information.

# GoogleTM Maps

This panel provides integration with Google Maps.

## **Default Zoom Level**

Select the level of zoom to use. The Switchboard user can always zoom in or out.

# Extensions allowed to use this panel

Select one or many extensions on the left, and click the right-facing green icon to put those extensions into the box on the right. All of the extensions in the box on the right are allowed to use this panel.

## **Test your settings**

Enter a phone number to look up. Click **Test Settings** to see the results of your test.

# **Custom Panels**

Custom panels let you build your own panel that integrates with any Web application you choose. You can build the URL with Switchvox variables, which pass caller data to the application.

#### **Panel Name**

Enter a unique name for this panel. The panel name appears as the title of the panel window.

### **Panel Description**

Enter a description of this panel. This is only displayed in the Admin Suite, it is not shown in the panel.

#### Panel URL

This is the URL to be loaded in the panel, for example:

http://news.google.com/news?q=%CALLER ID NAME%

The URL can contain variables (between percent signs) that are substituted with real values during each request. (See URL Variables.)

# Extensions allowed to use this panel

Select one or many extensions on the left, and click the right-facing green icon to put those extensions into the box on the right. All of the extensions in the box on the right are allowed to use this panel.

# **URL Variables**

You can include one or more variables in a Custom Panel URL.

#### **CALLER ID NUMBER**

The phone number of the current caller. Example: 8585551212

#### **CALLER ID NAME**

The text component of Caller-ID for the current caller. Example: BOB JONES

## **EXTENSION**

The extension of the current Switchboard user. This does not apply to who is calling, but is instead the extension who has launched this Switchboard panel.

#### **SESSION ID**

A string of numbers and underscores representing the current switchboard session. (There is no guarantee that this session id is unique across logins or machines.)

#### FIRST LOAD

A boolean integer value indicating if this is the first time the panel is requesting the page. On startup, the panel will request the URL without any associated call information with FIRST\_LOAD set to 1. After subsequent calls, FIRST\_LOAD will be set to 0.

#### PARTIAL REG CODE

The last four digits of your Switchvox's registration code.

## FULL\_REG\_CODE\_DIGEST

A hexadecimal MD5 hash of your Switchvox's registration code.

# **Distinctive Ringtones**

Switchvox can store and play distinctive ringtones that can be used on Polycom phones (each phone needs a Phone Feature Packs). For details about using these ringtones on a SIP phone with a Phone Feature Pack, see the User Suite's Phone Features

You can add as many ringtones to Switchvox as you want. However, each phone has two important limitations for supporting ringtones that do not come standard on the phone:

- Maximum disk-use of 300 kilobytes for all ringtones
- Maximum of 8 ringtones

# Add a Ringtone



### To add a ringtone:

- 1. Click **Browse**, and find the file on your computer.
- 2. In the Name field, enter a name for this ringtone.
- 3. Click Add Ringtone.

The Distinctive Ringtones list is refreshed, with your new ringtone listed.

# **Distinctive Ringtones List**

This list shows the ringtones that you have uploaded to Switchvox, and the ringtones that come with a Polycom phone.

Play the ringtone.

The audio file is downloaded to your computer, then played using your default audio player for .WAV files.

**Edit** the ringtone.

You can upload a new file, or change the Name.

X Delete the ringtone.

You must verify that you want the ringtone deleted. If you delete a ringtone, every Ring-Rule that uses that ringtone is also deleted.

# **System Setup**

The System Setup tools let you configure hardware, channels, network, and other settings that form the foundation of Switchvox.

# **Hardware Setup**

The Hardware Setup tools let you scan for and configure new telephony hardware in the Switchvox server. You can also manage and delete hardware devices.

# **Hardware Devices**

Each device connected to the Switchvox server is listed under Hardware Devices.



To configure a device, click **Configure** for that device. To delete a device, click **Delete** for that device, then verify that you do indeed want to delete it. Devices are listed with Card Type, Device Type, Device Name, Spans (the logical groupings of channels), and Total Channels.

If you have added new hardware, Switchvox detects it when you start the server. If you have deleted a device and want Switchvox to detect it again, click **Scan for New Devices**. Switchvox detects the telephony hardware and includes it in the Hardware Devices list.

# **Hardware Country Settings**

If you need to change the country that your Switchvox server is located in, select it from the dropdown, then click **Set Country**. This setting also sets the call-progress tone scheme, which determines the tones that are sent to your callers.

# **Echo Reduction**

To reduce the echo on your analog lines, click **Run Echo Reduction Tool**. Switchvox runs a series of tests, then the analog lines are automatically calibrated to the results. Click **Remove Echo Reduction Settings** to revert the settings back to their default values.

We recommend running the Echo Reduction Tool if you move lines to other ports, or add new lines.

Running this tool causes your system to go offline for approximately two minutes per analog line. Any current phone calls are dropped, and any agents logged in to call queues are logged out.

# **Troubleshooting**

If Switchvox is not detecting hardware, please be sure that all cards are installed properly, and that the device you are using is supported.

# **Channel Admin**

The Channel Admin tools let you manage hardware device channels to enable multiple signaling methods and configure callback extensions.

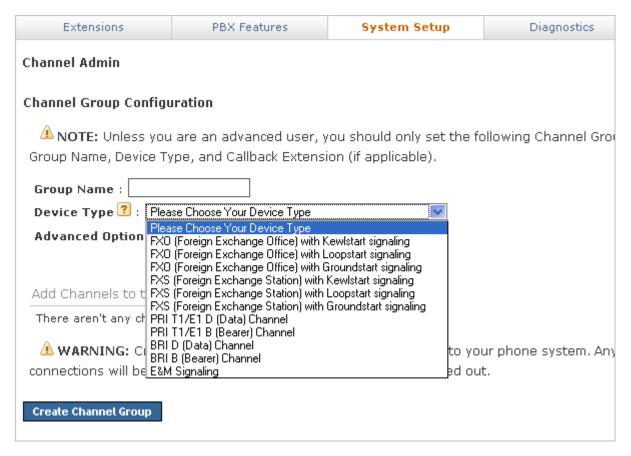
Channel Admin displays all the channels available in your system, and lets you group channels together by signaling type. You can also modify or delete existing groups.



To create a channel group, click **Create New Channel Group**. This displays the Channel Group Configuration page.

Enter a name for the channel group, specify the device type, and select the channels to add to the group. Also enter the Callback Extension and the Default Fax Extension for the channel. (Voice calls that do not match any Incoming Call rules or routes are sent to the Callback Extension. If you have fax installed, Fax calls that do not match any Incoming Call Routes are sent to the Default Fax Extension.) Unless you are an advanced user, do not change any of the other settings.

When you are done, click Create Channel Group to create the group.



**WARNING:** Switchvox restarts after you create a new channel. Any current calls are dropped, and any agents logged into queues are logged out.

# **Device Types**

Device Type describes the type of line that is connected to the PBX through this device.

## **FXO and FXS**

Foreign Exchange Office (FXO) and Foreign Exchange Service (FXS) device types refer to the type of interface on the telephone equipment: an FXO interface receives power (battery) and receives ring signals; and an FXS interface provides power (battery) and generates ring signals. FXO devices can only communicate with FXS devices and vice versa. Each FXO and FXS type listed in the dropdown has an associated signaling type: kewlstart, loopstart, or groundstart. The signaling type is the "language" the interface uses to communicate to the devices connected to it. Some devices have their signaling type preset and cannot be changed (for example, analog cards). Some devices can have their signaling type changed to match the devices they are connected to.

You cannot have both FXS and FXO channels in the same group. However, you can create more than one group per signaling type.

FXO (Foreign Exchange Office) with Kewlstart signaling

Select this option if you are connecting normal analog phone lines from the phone company into your PBX. This option can also be used with certain T1 lines or T1 channel banks in rare instances.

## FXO (Foreign Exchange Office) with Loopstart signaling

Do not use this option. Use FXO with Kewlstart instead. Kewlstart is fully compatible with Loopstart, and will detect hangups on most lines which the Loopstart option does not do.

## FXO (Foreign Exchange Office) with Groundstart signaling

Groundstart is an alternate type of signalling used with analog phone lines from the phone company.

Groundstart phone lines are slightly unusual, so double-check with your provider before selecting this option.

### FXS (Foreign Exchange Station) with Kewlstart signaling

Select this option if you are connecting a normal analog handset to an FXS port of an analog card.

## FXS (Foreign Exchange Station) with Loopstart signaling

It is recommended to use the Kewlstart option instead of Loopstart.

# FXS (Foreign Exchange Station) with Groundstart signaling

Select this option if you are connecting a groundstart FXS device, such as another analog PBX, to the FXS port of an analog card.

# PRI

The Data channel is a single channel used for call control on T1/E1 PRI connections. It is usually on channel 24 for T1, and 16 for E1 connections. The Bearer channels carry the audio for T1/E1 PRI calls. Full PRIs use all channels except for the D channel. Partial PRIs use a subset for the Bearer channels.

#### PRI T1/E1 D (Data) Channel

The Data channel is a single channel used for call control on T1/E1 PRI connections. It is usually on channel 24 for T1, and 16 for E1 connections.

#### PRI T1/E1 B (Bearer) Channel

The Bearer channels carry the audio for T1/E1 PRI calls. Full PRIs use all channels except for the D channel. Partial PRIs use a subset for the Bearer channels.

# BRI

The Data channel is the third channel on every BRI that controls signaling. The Bearer channels are the first two channels on a BRI that actually pass calls.

# BRI D (Data) Channel

The Data channel is the third channel on every BRI that controls signaling.

## BRI B (Bearer) Channel

B channels are the first two channels on a BRI that actually pass calls.

# **E&M Signaling**

The E&M (Ear & Mouth) Signaling device type is a different type of T1 signaling used by some telephone companies.

#### E&M Signalling

E&M signalling describes certain older T1 lines. Select this option if you have an E&M or E&M-wink circuit, sometimes referred to as a Supertrunk.

#### **Callback Extension**

The extension to ring when a voice call comes in over this provider or channel. Callback extensions can be any extension type (your receptionist's phone, an IVR, etc.).

#### **Default Fax Extension**

The extension to ring when a call comes in over this provider or channel, and is identified as a fax. (Switchvox only identifies T.38 faxes over SIP providers. A fax coming in over an IAX provider or a channel, or a fax using a different standard may not be identified as a fax.)

# **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

# **Aggressive Echo Cancellation**

This option enables aggressive echo suppression in the Analog echo canceller. This is done by making the line half-duplex, or by muting the party that is not actively speaking. While this does eliminate echo, it might cause other problems such as not being able to interrupt a speaking party, or clipping on the leading and trailing edge of speech. Setting this option to run for the first 5 seconds of a call is an effective way to eliminate echo that eventually trains out. Otherwise, it's best to keep this option disabled.

# **Apply Noise Reduction Filter**

This option applies a filter on calls to reduce analog line noise. This can improve quality on very old analog lines in rural areas. Before enabling this option, it's best to call your provider to see if they can improve the signal quality or strength from their end, as this option only masks over the noise.

#### **Automatic Gain Correction**

Enabling this will automatically adjust the received volume of calls to a standard volume level. This should be the first thing to try if inbound calls are too quiet, specifically on analog lines.

### Automatically determine call progress

Do not enable this option unless explicitly instructed to. Enabling this option in the wrong environment will result in dropped calls.

#### **Battery Debounce**

This option is **extremely advanced** and only used in very rare circumstances in certain countries.

This option controls how Switchvox looks for a hangup signal on Kewlstart lines. In Kewlstart signalling, hangup is sent by a voltage drop in the analog lines for a certain period of time.

The Battery Debounce controls how longSwitchvox waits until concluding that it has received a hangup signal. If set too low, line noise or lightning or regular speech can be interpreted as a hangup. If set too high, correct hangup signals will not be received. The default value for this setting is chosen to be the perfect balance. Do not change it unless you've physically measured the duration of a battery drop for your country's telephone lines.

### **Battery Threshold**

This option is **extremely advanced** and only used in very rare circumstances in certain countries.

This option controls how Switchvox looks for a hangup signal on Kewlstart lines. In Kewlstart signalling, hangup is sent by a voltage drop in the analog lines for a certain period of time.

The Battery Threshold controls how low the voltage must go before being considered *dropped*. The default setting for this option will work in most countries. Do not change it unless you've physically measured the duration of a battery drop for your country's telephone lines.

#### Caller ID Signaling

Caller-ID is sent over analog lines differently in different countries. This option controls which method to read the caller-ID. The options are Bell (for countries like the USA), v23 (United Kingdom), and DTMF (Sweden). When changing this value, also make sure to adjust the Caller ID Start value to correspond to your country.

#### **Caller ID Start**

Different countries send caller-ID at different times. This option instructs Switchvox when to look for the caller-ID signal. The choices are *ring* (for countries like the USA), or *polarity* (for countries like the United Kingdom).

# **Detect busy signal**

This option will listen for the busy signal to trigger a hangup. It is **strongly recommended** to leave this option disabled and have your telco send a true disconnect signal. This option may be enabled in countries that do not have disconnect supervision. If you are in the USA, UK, Australia, or Canada, you should not need to enable this option. If this option is enabled, it is advised to increase the Busy Count setting to as many tones as possible to avoid false-positives that may result in dropped calls.

#### **Echo Cancel**

This option enables or disables echo cancellation on the system. We strongly recommend setting this option to Yes. Setting this option to No will disable both software echo cancellation, and any onboard hardware echo canceller on the given card.

## Echo Cancel when bridging TDM channels

This option controls echo cancellation for calls that enter Switchvox on one hardware channel, and exits Switchvox on another hardware channel. IE: this affects calls that do not touch any VoIP Providers or SIP handsets.

Calls of this nature do not normally need echo cancellation, so it is safe to leave this option to the default of *No* . Note that this setting does not apply to calls to normal SIP handsets.

#### **Echo Canceller Tail Length**

This controls how long the canceller should listen for echo to return. It is best to use the smallest value possible. For analog lines, echo is usually in the 64 or 128 range. For T1 lines, 128 or greater.

#### **Echo Canceller Type**

Different echo canceller types may perform better in certain situations.

#### Analog

A good general-purpose echo canceller that works best on analog lines, or on T1 lines with shorter echo tails. This is the default echo canceller, and is a good place to start.

#### T1

This echo canceller works best on T1 lines with long echo tails, but may also outperform other types in certain analog environments.

#### **HPEC**

Digium, Inc.'s High Performance Echo Cancellation software. To use this option, you must have an HPEC license installed.

## Acoustic

This experimental echo canceller is tuned for acoustic echo. While good on paper, this method of cancellation hasn't worked very well in practice.

#### **Fax Detection**

Set the Fax Detection for this Channel Group. For more information, see About Faxing.

#### Roth

Detect incoming and outgoing faxes, and behave appropriately.

#### No

Do not detect incoming or outgoing faxes.

# Incoming

Detect incoming faxes and behave appropriately. Do not detect outgoing faxes.

## Outgoing

Detect outgoing faxes and behave appropriately. Do not detect incoming faxes.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

#### **Fax Error Correction Mode**

Error Correction Mode (ECM). The default is to leave this feature enabled.

# Hang up on Polarity Switch

Some countries, most notably Australia, signal remote hangup by switching the polarity of the lines. This is sometimes called ROIC (Reverse On Idle Condition). Only enable this option if you have confirmed with your telco that hangups are signalled with polarity reversal.

#### **Hardware Echo Cancellation**

This option enables the hardware echo cancellation module on BRI cards.

#### Hide caller id

Enabling this option will disable sending caller-id for this channel group.

#### Maximum transfer rate for fax transmissions

Maximum transfer rate used during fax rate negotiation. The default maximum transfer rate is 14400.

#### Minimum transfer rate for fax transmissions

Minimum transfer rate used during fax rate negotiation. The default minimum transfer rate is 2400.

### **Network Specific Facility Code**

This option is rarely used. If you are unsure of what to do, leave it set to None.

NSF, or Network Specific Facility codes are special codes that are sent in PRI facility messages and are used to signal additional data to your telephone company. Do not change this setting unless your telephone company explicitly stated that they require a specific NSF value.

#### Overlap dial

Overlap dialing sends each digit pressed as it's entered, instead of sending the entire number all at once. It is safe to leave this option disabled.

# **Pause Before Dialing**

This option adds a slight delay to allow the off-hook signal to propagate to the telephone company before sending down DTMF digits. This should not need to be changed.

# **PRI Dialplan**

This controls how outbound calls are sent over the PRI. Check with your PRI provider to confirm which method they prefer.

#### **PRI Reset Interval**

The default value for this option, 3600, is usually correct.

This controls how often the PRI B channels are reset when they are unused. For the vast majority of PRI lines, this has no effect on the PRI and is purely done just to be safe. However, some PRIs in countries such as India may have problems with being reset. In that case, you may want to enter a value of *never*.

### **PRI Switch Type**

Your PRI provider should be able to provide you with the preferred PRI Switchtype. The most common option is *National ISDN 2*, sometimes referred to as *NI2*.

## **Relax DTMF recognition**

Enabling this option will make the DTMF interpreter much more permissive. If you are having trouble receiving DTMF key presses in IVRs with calls over a channel group, enabling this option may help. Unfortunately, it also may result in many more false positives, or double keypresses. In cases where poor line quality is hampering DTMF detection, it is highly recommended to improve the line quality first before enabling this option.

### **Ring Debounce**

This option is only used in very rare circumstances in certain countries.

This option controls how long Switchvox waits before confirming a ring on analog lines. The value is in 1/8000ths of a second, and the default is 1024. Larger numbers cause Switchvox to wait longer, potentially making it harder to receive false-positives or ghost calls. Smaller numbers cause Switchvox to be more sensitive to rings.

#### **RX Gain**

We recommend enabling Automatic Gain Correction before altering this option.

This controls the volume of audio received from this channel group. For example, if incoming callers are too quiet or too loud, this option can change the volume. Values are in decibels, and can be positive or negative to make calls louder or quieter respectively. This is usually only needed on analog lines (not T1 lines) to compensate for signal loss in the wires. Normal values range in the -15 to 15, and it's recommended to start low, and increment the gains by a factor of 2 until they reach the correct volume (e.g., 2, 4, 6, 8...).

## **Secondary Device Type**

This option should almost always be set to PRI (Primary Rate ISDN) CPE Side. Consult your T1 provider before selecting an alternate option.

## Train echo canceller at beginning of calls

This option sends a pulse at the beginning of a call in the attempt to pre-train the echo canceller. This can improve situations with echo at the beginning of a call.

#### TX Gain

This controls the volume of audio *sent* from this channel group. It is recommended to not increase this value, although lowering it to -3 may improve call quality and/or echo. If in doubt, leaving this setting at the default of 0.0 is recommended.

## **Use Caller ID**

Enabling this option will enable receiving (and sending) caller-id.

It's almost always recommended to leave this set to Yes. The one situation where it may be prefereable to disable caller-id is when using analog lines in a country (such as the USA) that sends the caller-id between the first and second ring. In order to receive caller-id on analog lines, Switchvox has to wait until the second ring before answering. If caller-id is disabled, Switchvox will answer inbound calls immediately, without waiting for the caller-id.

# **Phone Setup**

If you purchased your phones with Switchvox, then your phones are already configured. It's going to be very easy to get your phones up and running! Plug the phones into your network and wait for them to start up. Once you see the date and time on the phone's display, pick up the handset and dial 800. This extension rings a sample Interactive Voice Response (IVR) menu. The call should connect and you should hear a recorded voice say "congratulations, Switchvox has been configured properly."

If you did not purchase your phones with Switchvox, we offer the **Phone Setup** tool that lets you easily set up phones from Polycom® and snom technology Inc.

**Note:** If you do not have these types of phones, you need to refer to your phone's documentation for information on how to set it up for use with Switchvox.

Select **System Setup > Phone Setup** to display the **Phone Setup** page.

The following must be in place before you set up your phones. Make sure that

- Switchvox is up and running.
- Your phones are on the same network as Switchvox.
- You have a Phone Feature Packs available for each phone.
- Any Polycom phones you are setting up are running SIP firmware 1.6.7 or higher.
- Any snom phones you are setting up are running snom firmware version 6.5.18 or higher.
- You are comfortable with Switchvox updating each phone's firmware as necessary. If you do not want a phone's firmware updated, you need to set up that phone manually (not as part of this Phone Setup process).
- You are comfortable with Switchvox setting each phone's config authentication login and password, and its handset password. The config authentication login and password are randomly generated and can be found in the **Configured Phones** list. The handset password is always 456. If you do not want this information changed for a phone, you need to set up that phone manually (not as part of this Phone Setup process).

**IMPORTANT:** If your phone is not on the same network as Switchvox, or it does not have the required firmware, it cannot be discovered by Switchvox. If this is the case, see Making Phones Visible to Switchvox. If you have a Polycom 300 or 500 phone, you must set up your phone manually (you cannot use the **Phone Setup** tool). For help, see the documentation for your phone.

The Phone Setup tool has three main categories: **Unknown Phones**, **Unconfigured Phones**, and **Configured Phones**. You must confirm each phone for configuration, and then configure each phone with extension details.

To begin setting up phones on Switchvox:

- 1. Place each phone where it should be located, plug it into your network, and turn it on. At this point, it is helpful to note the MAC address of each phone, but it is not necessary.
  - When each phone boots up (after you turn it on), Switchvox detects it and adds it to the **Unknown Phones** list. These are phones that Switchvox has seen on the network, but does not have control over.
- 2. In the Admin Suite, select **System Setup > Phone Setup**.
  - The Phone Setup page opens to Unknown Phones. If a phone does not appear here, try rebooting the phone.

# **Unknown Phones**

Unknown phones are the phones that Switchvox has found on your network, but they are not yet configured for use with Switchvox. (These phones do not receive any configuration files from Switchvox.)

Phones that have not been detected on the network in over 14 days will stop showing up in the **Unknown Phones** list.



- 1. Check the box next to each phone you want to configure, and then click **Confirm Checked Phones**. Switchvox sends firmware and configuration files to the phone, then a command to reboot.
  - **IMPORTANT:** If you do not want to configure a phone, do not check its box. That ensures that the phone does not receive Switchvox configuration files. In particular, if you have more than one PBX on your local network, make sure that you do not confirm phones that are set up on another PBX. This overwrites the configuration of these phones and they will no longer connect properly to the other PBX Phones.
- 2. Be sure that each phone has finished rebooting. This can take as much as 10 minutes, and it is important that it is finished before you move on to the next step. If the date and time are shown on the phone's display, that indicates that it has finished rebooting.

Once your phone has finished rebooting and is back online, it is in the Unconfigured Phones list. Now it needs to be assigned an extension.

# **Phone Model Not Reported**

Switchvox is not able to determine the phone model for some phones. Despite that, these phones can still be configured. During the configuration process, Switchvox updates the firmware on the phone (if necessary), and that helps to determine the model.

# **Unconfigured Phones**

Unconfigured phones are the phones that are registered with Switchvox and can receive configuration files, but these phones do not yet have an assigned extension.

**Note:** The 'Config Authentication' Login and Password shown on this page are used when the phone requests a current configuration file from Switchvox. This is different from the phone's registration login, and the extension's login, and you can safely ignore this information.

To remove a phone from this list and put it back on the **Unknown Phones** list, check the box for that phone and click **Reject Checked Phones**.



To assign an extension to a phone:

- 1. Enter the extension in that phone's row. You can use an existing extension as long as it is a SIP extension, and has not already been configured for use by a phone.
  - If you know the phone's MAC address and the extension that belongs with it, you can enter the extension on this page. Do this for each phone that you know the extension for.
  - If you do not know the phone's MAC address, pick up each phone's handset and follow the prompts to enter the correct extension for that phone. When you are done, go back to the Phone Setup page. The **Unconfigured Phones** list now has an extension listed for each phone.
  - If an extension already existed, the owner's name and other information is displayed.
- 2. If the extension does not already exist, enter the extension owner's first and last name, and email address. You can also select the Extension Template you want to use to create this extension (or just leave it set to **Default**).
- 3. If you have a secondary Switchvox for the phone, enter that IP address in the Alternate Host field.
  - If a phone is unable to reach the primary Switchvox (this one that you are using to setup this phone), the phone will try to connect with a SIP server at this IP address. This should be a secondary Switchvox with the same configuration as your primary Switchvox. You can set up a secondary Switchvox using Backups from this primary Switchvox.
- 4. Once each phone has the correct extension, check its box and click Accept Checked Phones.

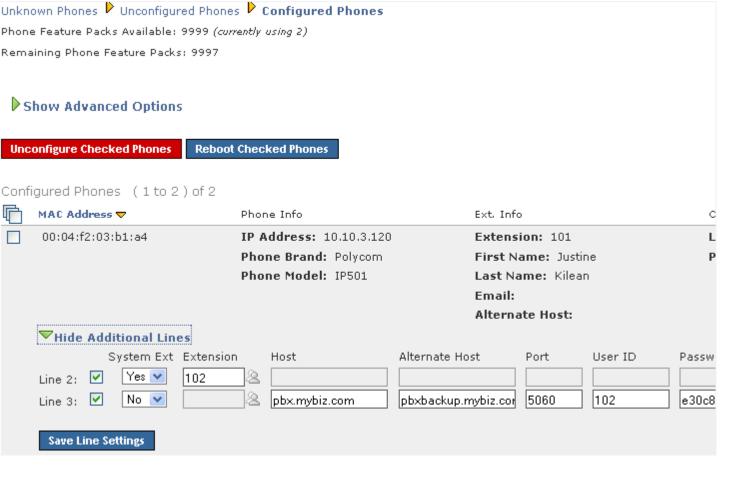
Switchvox creates any extensions that do not already exist, pushes out the final configuration to each phone, and sends a command to each phone to reboot again.

If an extension already exists, this process does not change the phone's registration login, or the extension's password. If the extension is created in this process, then the phone registration login is randomly generated, and the extension password is the same as the extension number.

Once your phone has finished rebooting and is back online, it is in the Configured Phones list. If you want to assign additional lines to this phone, you can do that on the **Configured Phones** page.

# **Configured Phones**

The phones are now on the **Configured Phones** list on the Phone Setup page. Each time a configured phone reboots, it receives configuration updates and its user's specified Phonebook (see the User Suite's Phone Features Options).

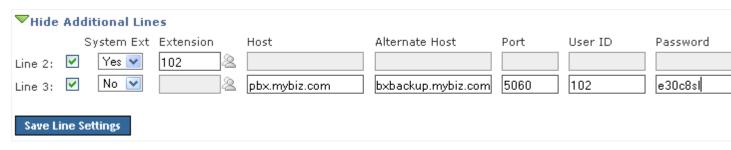


# **Additional Lines**

Click **Show Additional Lines** to add or remove 2 additional lines on this phone.

To add a line, specify whether this is a Switchvox Extension or not, and then enter the extension information.

- o If it is a Switchvox extension, you just need to enter that extension number.
- If it is not a Switchvox extension, you can enter host name, alternate host name (if available), Port, User ID, and Password.



# **Making Phones Visible to Switchvox**

If Switchvox cannot discover your phone (you cannot see your phone in the **Unknown Phones** list), you need to manually point the phone to the Switchvox server.

For Polycom phones: Reset the phone back to its factory defaults, then set the network configuration Server type to *HTTP* and the Server Address to 'SWITCHVOX\_IP\_ADDRESS/pc.' For example, if Switchvox has an IP address of '192.100.0.1' then set the server address to '192.100.0.1/pc'. To do this for Polycom models 320, 330, and 430, hold down the 1, 3, 5, and 7 keys simultaneously. For other models, hold down the 4, 6, 8, and \* keys until the phone prompts you for a password. Then press 4, 5, 6 on the keypad, then select **Exit**.

For snom phones: Reset the phone back to its factory defaults, then set the Setting URL to 'SWITCHVOX\_IP \_ADDRESS/pc/PHONE\_MAC\_ADDRESS.xml. For example, if Switchvox has an IP address of '192.100.0.1'and the phone has a MAC address of '00:11:11:AF:2A:EA', then set the Setting URL to '192.100.0.1/pc/00:11:11:AF:2A:EA.xml'. (See your phone's user manual for instructions on how to accomplish this.)

After those changes, make sure that the phone is rebooted. Rebooting can take as much as 10 minutes, and it is important that you let it finish completely. If the date and time are shown on your phone's display, that indicates that your phone has finished rebooting.

When your phone reboots and comes back online, it appears in the Unconfigured Phones list in Phone Setup.

**IMPORTANT:** To configure phones that are external to your network, Switchvox must be accessible outside of your network over port 80. Please contact your network administrator for information about how to do this.

# **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

# Only Accept Calls from the PBX

Check this box if your Switchvox-configured phones should only accept calls from this Switchvox.

# **Digit Map Options**

When your phone gets configuration files from Switchvox, then Switchvox provides digit map settings for your phone based on the extension lengths your system uses and typical dialing requirements.

## **Extension length**

This is the setting your digit map uses for dialing extensions on Switchvox. By default this is the length of the longest extension. You might change this if you have two Switchvoxes that are peered together and they use different extension lengths.

## Use 10 digit local dialing

Some providers require 10 digits to be entered, even for local dialing. If this is the case for your provider, you should check this box.

# **Custom Digit Map**

You can specify a custom digit map for Switchvox-configured phones. If you do this, Switchvox ignores all the other rules for creating a digit map, and pushes the custom digit map to your configured phones.

#### **Custom Digit Map Timeout**

You can specify a Timeout (in seconds) for each segment of your custom digit map. If there are more digit maps than timeout values, the default value of 3 is used. If there are more timeout values than digit maps, the extra timeout values are ignored.

# **Phone Discovery Options: DHCP Sniping**

Polycom phones with SIP Applications prior to 2.2.2 do not support DHCP INFORM. To configure these older phones, Switchvox must be more aggressive, usurping your network's DHCP server. Once the phone is upgraded, Switchvox returns to normal behavior. It's probably safe to leave this option enabled, but if your DHCP server reacts poorly when older phones are configured, disabling this option might help.

# **Custom NTP Server**

You can specify an Network Time Protocol Server to set the time on your Switchvox-configured phones.

# **Phone Feature Packs**

Phone Feature Packs let you easily configure your phones for use with Switchvox, and offer great tools for customizing your phones.

You need one Phone Feature Pack per SIP phone. Please see your reseller for instructions on purchasing and activating more Phone Feature Packs. If you purchased your phones with Switchvox, then each phone already has its own Phone Feature Pack, and is already configured for use with Switchvox. Phone Feature Packs are not necessary for any other type of extension.

A Switchvox Phone Feature Pack offers the following features:

Customized phone displays
 Customize some of the information displayed on your phone. See Phone Options.

Status Indicators

Set a line status indicator light on your phone. This can show whether a call is waiting in a parking lot, or whether you are logged into a queue. See Status Indicator Entries.

Distinctive Ringtones

Set the audio file of your choice to ring on incoming calls, depending on the caller. See Distinctive Ring Rules.

Auto-answer

Set an automatic answer in Intercom mode, depending on your caller. See Distinctive Ring Rules.

Combine Ringtones and Auto-answer

Combine a ringtone and auto-answer, depending on the caller. See Distinctive Ring Rules.

Phone Applications

Access many of the Switchvox features right from your phone.

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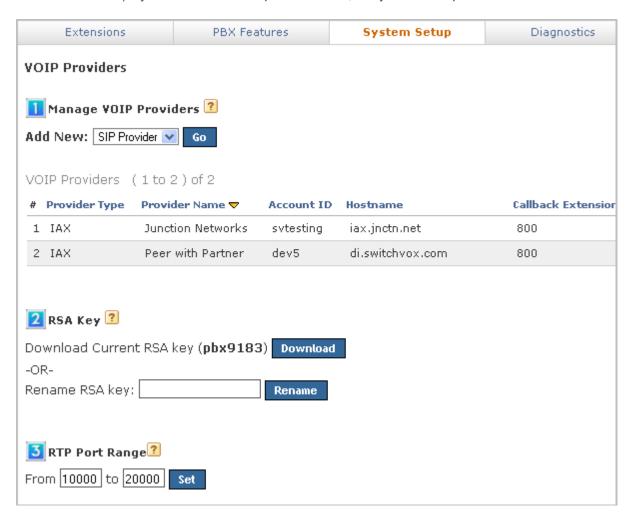
The Phone Features available on your phone will vary, depending on the type of phone you have.

- The **General Options** are available on snom and Polycom phones.
- The other features are only available on Polycom phones. If you have a newer Polycom phone with three or more lines (e.g., the VVX 1500 or SoundPoint 450, 550, 560,650, or 670) you should be able to take full advantage of your Phone Feature Pack. Other Polycom phones may not offer all of the features.

# **VOIP Providers**

This lets you manage your Voice Over Internet Protocol (VOIP) Providers. VOIP providers can be either commercial VOIP phone services, or other VOIP hardware such as a peered Switchvox.

There are two types of VOIP providers: SIP or IAX. (These are two common protocols used for placing phone calls over the internet.) If you are unsure which protocol to use, ask your service provider.

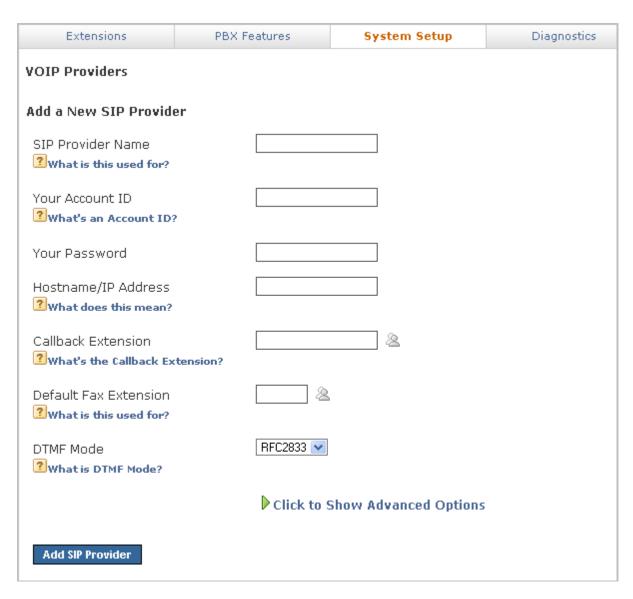


# Adding a SIP Provider

Select **SIP Provider** from the **Add New** dropdown, then click **Go**. The Add a New SIP Provider page is displayed.

Enter the information about the new provider, then click Add SIP Provider to create the new SIP provider.

After you have created a provider, you are prompted to go to the System Status page. This page displays your VOIP providers and their status. If Switchvox successfully contacts your provider, it is displayed as **Registered**. Now if you call the number your provider gave you, your call should ring the Callback Extension.



## **SIP Provider Name**

Create a name to refer to this provider. This is only used for referring to this Provider in other areas of the Admin Suite.

# **Your Account ID**

The account ID provided to you by your SIP provider. This may also be called Login, or Authentication ID.

#### **Your Password**

Your password for the account.

## Hostname/IP Address

The hostname or IP address for the SIP server, provided to you by your SIP provider. This may also be called Registrar Server.

## **Callback Extension**

The extension to ring when a voice call comes in over this provider or channel. Callback extensions can be any extension type (your receptionist's phone, an IVR, etc.).

#### **Default Fax Extension**

The extension to ring when a call comes in over this provider or channel, and is identified as a fax. (Switchvox only identifies T.38 faxes over SIP providers. A fax coming in over an IAX provider or a channel, or a fax using a different standard may not be identified as a fax.)

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

#### **DTMF Mode**

DTMF tones are the sounds emitted when you press buttons on your phone.

- SIP extensions: Refer to your phone's user manual for the DTMF mode that your phone uses.
- SIP providers: Ask your provider which DTMF mode it supports.

If you are unsure of which DTMF mode to select, use RFC2833 (the most common method).

# **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

# **Peer Settings**

The Peer Settings determine the relationship with this provider.

### **Host Type**

*Provider* is a host you must actively register to inform it of your location and settings. *Peer* is another SIP PBX, including another Switchvox.

## Host is a Switchvox system

If this VOIP Provider is a peered Switchvox, set this to Yes. This lets you use advanced functionality with the peered Switchvox.

#### Treat system's users like local users

If you want Switchvox to exchange presence and line-use information with this peered Switchvox, set this option to Yes.

## Jabber Hostname

You can specify a hostname for the peered Switchvox's Jabber server. To find this hostname, look in the peered server's Network Settings page.

This is important if you are peering Switchvoxes and want to use functions such as Presence or the Chat Panel.

# **Apply Incoming Call Rules to Provider**

This option lets you alter the routing of incoming calls.

Yes is the default setting. It uses the rules on the Incoming Calls section as well as the Callback Extension to decide which extension to ring when you receive a call over this provider.

Select *No* if this provider is a peer. This allows callers from this peer to directly dial any extension in Switchvox. Incoming calls will not follow the rules in the Incoming Calls section and they will not ring the Callback Extension.

#### **Outgoing Call Rules**

If you have selected *No* for the Apply Incoming Call Rules to Provider option, you can use this option to define how the remote peer can place outbound calls through this Switchvox. Use the check boxes to assign which outgoing rules apply to this peer. Check an *Allow* box to apply the rule to this peer's inbound calls. Check a *Deny* box to block all calls that match that rule, such as blocking all long-distance numbers. To bypass a rule, leave both options unchecked. See Outgoing Calls for more information about each outgoing rule.

# **Caller ID Settings**

The Caller ID settings determine how your Caller ID is set on outgoing calls.

### **Supports Changing Caller ID**

Changing your caller ID information may cause all outbound calls to fail, so be careful when using this setting. Be sure that your provider accepts a modified Caller ID, then enable this option and customize the Caller ID displayed on outbound calls.

#### Caller-ID method

Leave this set to *From Header* unless your provider explicitly tells you to use a different setting. The vast majority of providers require caller ID to be supplied in the From header.

#### Caller ID Name / Caller ID Number

Enter a name and phone number to display as your Caller ID when making outbound calls with this provider.

**WARNING:** Setting this field may cause all outbound calls to fail; be sure that your provider supports this feature. If your provider does not support this feature, clear the Caller ID Name and Caller ID Number fields.

# **Connection Settings**

The Connection settings determine how the connection is established with this provider.

#### **SIP Port**

This is the UDP port used to communicate with this SIP Provider. By default this is set to 5060, the standard port for SIP communication. This port can be changed if your provider or peer requires it.

## **SIP Expiry**

This controls the duration (in seconds) between SIP registration requests sent to your provider. Selecting a value other than 120 may cause problems with NAT routers. Do not change this unless explicitly instructed by your provider.

## **Proxy Host**

If your SIP provider says you need to use a SIP proxy, enter the proxy host IP here.

#### **Authentication User**

If your SIP provider says you need a different authentication name than your account name, enter that authentication name here.

#### **Always Trust this Provider**

Most providers require this option be set to Yes. It is a little less secure because it does not require authentication on incoming calls from this provider. If you set this to No and it does not work properly, set it back to Yes.

**Note:** This option lets the provider send you calls without authenticating. Most SIP providers do not authenticate, therefore setting this to No will likely cause problems. Leave this option set to Yes unless instructed otherwise.

# **Qualify Hosts**

Enabling this option sends a SIP Options packet to this SIP provider every 60 seconds. This keeps NAT tunnels open. Response latency for these requests are shown in the System Status page. Note: Do not enable this option if your provider does not respond to SIP Options packets.

#### Include user=phone in SIP

This option has no effect on the vast majority of SIP providers. Enable it if your provider explicitly requests it. This adds the string "user=phone" on the end of the SIP URI used in various messages.

#### **Use Local Address in From Header**

Leave this set to *No*. Changing it to Yes will violate section 8.1.1.3 of the SIP standard, RFC 3261. Only set this to Yes if your provider explicitly requests that you violate the SIP standard in this manner. If this is Yes, Switchvox uses its own local IP address in the From header of requests.

#### **SIP Provider Host List**

Some SIP providers have multiple servers on the internet that communicate with Switchvox. For example, you may register sip.example.com but receive calls from ny.example.com, la.example.com, and mia.example.com. Use this option to list multiple hosts (or IP addresses) so that Switchvox can route calls from them. If you do not know if there are multiple hosts, ask your provider. If there are multiple hosts, ask your provider to supply you with a list of their IP addresses.

# **Call Settings**

The Call settings determine how to manage calls with this provider.

#### **Provider Codecs**

Some providers support more than the normal codecs. These additional codecs may provide compression or other features that save bandwidth or lead to better call quality. However, you should check with your provider before selecting another codec, to be sure it is supported. If your Switchvox is a peer with another Switchvox, you can choose any audio codec.

For SIP providers, the video codecs H.263, H.263+, and H.264 are available. Video calling is not available over IAX. The following table describes the audio codecs. If you have Switchvox SMB and you entered a license key for the G729 Codec in **Machine Admin > Digium Addon Products**, you will also see G729 in the list of codecs.

**Table 1: Audio Codecs** 

Codec	Bandwidth Usage (kbits/s)	Sound Quality	CPU Usage
ULAW	85	Great	None
ALAW	85	Great	None
G722	85	Excellent	Low
G726	54	Good	Low
SPEEX	33	Great	High
GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

#### **Map Distinctive Rings**

Some providers let users set different ring tones based on the dialed number, but do not pass the dialed number information directly. You may have to experiment with this setting and the settings of your SIP provider to get the right phone number associated with the right ring type.

## **Enable Jitterbuffer**

Enable this option to combat network problems. A jitterbuffer induces a small delay in call latency to make up for lost or delayed voice packets. You may want to enable the jitterbuffer if you have bad network connectivity between Switchvox and the Provider.

The options are as follows:

Never: It's best to leave the jitterbuffer disabled if you are not having any audio quality problems.

**When Needed:** Intelligently enable the jitterbuffer with a dynamic window size when the call is terminated on Switchvox. Try this option first if you have audio quality issues.

**Always:** Always enable the jitterbuffer with a dynamic window size. This may be useful if you are using phones or other SIP devices that do not have their own jitterbuffer.

When Needed with Fixed Window: Intelligently enable the jitterbuffer with a fixed window size. This may be useful if jitter is sporadic, yet sudden. Or, if you need to mask problems with poorly configured routers that cannot implement QoS.

**Always with Fixed Window:** Always enable the jitterbuffer with a fixed window size. Select this if all else fails. Note that this just masks over the underlying issue. It's best to fix the jitter problem itself, instead of using this option.

#### **Allow Reunite**

We strongly recommend leaving this set to Never.

Enabling this option is incredibly risky, and will likely result in sporadic one-way audio, dropped calls, or not being able to place calls.

This lets Switchvox redirect the media stream of some calls, bypassing the PBX and passing directly between both participants. This may offer bandwidth and quality improvements in certain scenarios, but it's extremely risky.

**Never:** This is the safest setting that presents the least likelihood of trouble.

Always: The PBX directs media streams between participants when it is able to. Set this at your own risk.

**Non-NAT Connections Only:** The PBX redirects media streams only between participants who are not behind NAT.

**Always, Using the UPDATE method:** This is the same as the *Always* setting, but uses a different mechanism within the SIP protocol. Only set this if you have been explicitly instructed to do so.

**Non-NAT Connections Only, Using the UPDATE method:** The PBX redirects media streams only between participants who are not behind NAT, using a different mechanism within the SIP protocol.

**IMPORTANT:** This setting is just a recommendation to Switchvox that reinvite is possible. There are many scenarios, such as when a call is being monitored or recorded, where Switchvox will not be able to reinvite the call away.

## **Always Send Early Media**

Setting this option to Yes causes Switchvox to send RTP audio before a call is answered. Enabling this option may resolve an issue where audio is not sent by certain SIP providers when an inbound call is cascaded out of the same provider.

## Voicepulse Connect DID Workaround

This option is a workaround for SIP providers that send two INVITE messages for every phone call to handle SIP devices that handle DIDs and ones that do not. Only enable this option if you are sure your provider is sending two INVITEs for every call: one with the DID in the From header, and one with your account ID in the From header.

### **Fax Settings**

The Fax settings determine how faxes are handled with this provider.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

**IMPORTANT:** We recommend that you only send T.38 faxes over SIP. We do not recommend sending faxes over IAX connections. Switchvox cannot necessarily recognize a fax over an IAX connection.

#### **Error Correction Mode for G711 Faxes**

The default is to leave this feature enabled.

### Minimum transfer rate for fax transmissions

Minimum transfer rate used during fax rate negotiation. The default minimum transfer rate is 2400.

#### Maximum transfer rate for fax transmissions

Maximum transfer rate used during fax rate negotiation. The default maximum transfer rate is 14400.

## Redundant signal packets for T38 fax sessions

This option specifies the number of redundant signal packets contained in a T38 signal frame. The default value is 3.

## Redundant image packets for T38 fax sessions

This option specifies the number of redundant image packets contained in a T38 image frame. The default value is 1.

## Maximum expected T38 packet delay

The result of this specification is that all T38 packets are padded to prepare the fax stack for the specified delay. As long as the packet delay is less than or equal to the specified value, the fax stack will not realize that there is an actual packet delay because of the packet padding. The default value of 800 milliseconds should cover most use cases.

# Adding an IAX Provider

Select **IAX Provider** from the **Add New** dropdown, then click **Go**. The Add a New IAX Provider page is displayed.

Enter the information about the new provider, then click Add IAX Provider to create the new IAX provider.

After you have created a provider, you are prompted to go to the System Status page. This page displays your VOIP providers and their status. If Switchvox successfully contacts your provider, it is displayed as **Registered**. Now if you call the number your provider gave you, your call should ring the Callback Extension.

Extensions	PBX Features	System Setup	Diagnostics
VOIP Providers			
Add a New IAX Provid How do I setup IAX	er		
IAX Provider Name  ? What is this used for?			
IAX Provider Label  How do I find out my Label?	Provider		
Your Account ID  ? What's an Account ID	?		
Your Password			
Hostname/IP Address  What does this mean?			
Callback Extension  ? what's the Callback Ex	xtension?	2	
Incoming Authentication			
RSA Authentication Ke	y Use Existing or Upload No	Key: freeworlddialup S	Browse
Caller ID Name  ? What is Caller ID Nam	e?		
Caller ID Number  ? What is Caller ID Num	ber?		
	Click to	Show Advanced Options	5
Add IAX Provider			

#### **IAX Provider Name**

Any string to identify this VOIP provider. This is only used for referring to this VOIP Provider in other areas of the Admin Suite.

#### **IAX Provider Label**

Label this provider gives itself when placing calls to your system. This is usually the name inside of [] brackets in Asterisk configuration examples.

#### **Your Account ID**

Username provided by IAX Provider to authenticate your Switchvox PBX to the provider during IAX registration and outbound calls.

#### **Your Password**

Your password for the account.

#### Hostname/IP Address

Primary hostname or IP address of your provider's IAX server. This hostname will be used to send registration requests. It will also be the host that will send any inbound calls to your PBX.

If your provider has multiple servers for their IAX service, you can use the Primary, Secondary, and Tertiary Outbound Host settings to list them. Otherwise, the Hostname/IP Address field will be used for placing outbound calls as well as inbound calls and registration requests.

#### **Callback Extension**

The extension to ring when a voice call comes in over this provider or channel. Callback extensions can be any extension type (your receptionist's phone, an IVR, etc.).

#### **IAX Configuration Files**

Many IAX providers distribute their configuration instructions in the form of configuration files for the Asterisk PBX. The following information will enable you to translate between Asterisk configuration files and Switchvox's IAX Providers information.

Note: This section is not necessary if you do not need to convert settings from Asterisk configuration files.

Here is an example register line in the iax.conf configuration file:

iax.conf register statement:	register =>	username	:password	@iax.provider.com
IAX Provider fields		Your Account ID	Your Password	Hostname/IP Address

Here is a section in the iax.conf configuration file:

 iax.conf parameter
 IAX Provider fields

 [label]
 IAX Provider Label

 username=myuser
 Your Account ID

 Your Password or

host=iax.provider.com Incoming Password
Hostname/IP Address

type=peer not used

context=notused replaced by Callback Extension
auth=rsa Incoming Authentication Type
inkeys=keyname RSA Authentication Key

callerId="John Doe <5551212>" Caller ID Name and Caller ID Number

disallow=all

allow=ulaw Provider Codecs

allow=alaw

In the extensions conf file, here is how to extract the settings from a Dial command:

extensions.conf exten => \_ username :password @iax.provider.com /\${EXTEN})

Dial string: X.,1,Dial(IAX2/

Hostname/IP Address or

Primary Outbound

Host or Fields
Your Account ID
Your Password

Secondary
Outbound Host or

**Tertiary Outbound** 

Host

#### **Incoming Authentication Type**

Select one of these choices from the dropdown: MD5, plaintext, and RSA. MD5 and plain text use your password for authentication. It is preferable to use the MD5 method over the plain text method. RSA uses public and private encryption keys to authenticate the provider. If you select this option, enter the public RSA key in the RSA Authentication Key field.

#### **RSA Authentication Key**

Select from the dropdown of available keys, or from your computer hard drive.

# Caller ID Name / Caller ID Number

Enter a name and phone number to display as your Caller ID when making outbound calls with this provider.

**WARNING:** Setting this field may cause all outbound calls to fail; be sure that your provider supports this feature. If your provider does not support this feature, clear the Caller ID Name and Caller ID Number fields.

## **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

# **Peer Settings**

The Peer Settings determine the relationship with this provider.

# **Registration Type**

This setting controls registration between this Switchvox and the remote host.

Provider is the default setting where this Switchvox sends registration to the remote host.

Client is the setting where the remote host is the client and it sends registration requests to this Switchvox.

*Peer* is the setting for peered hosts that do not need to register. If you choose *Peer*, make sure that both hosts have static IP addresses.

## Host is a Switchvox system

If this VOIP Provider is a peered Switchvox, set this to Yes. This lets you use advanced functionality with the peered Switchvox.

## Treat system's users like local users

If you want Switchvox to exchange presence and line-use information with this peered Switchvox, set this option to Yes.

#### Jabber Hostname

You can specify a hostname for the peered Switchvox's Jabber server. To find this hostname, look in the peered server's Network Settings page.

This is important if you are peering Switchvoxes and want to use functions such as Presence or the Chat Panel.

## Apply Incoming Call Rules to Provider

This option lets you alter the routing of incoming calls.

Yes is the default setting. It uses the rules on the Incoming Calls section as well as the Callback Extension to decide which extension to ring when you receive a call over this provider.

Select *No* if this provider is a peer. This allows callers from this peer to directly dial any extension in Switchvox. Incoming calls will not follow the rules in the Incoming Calls section and they will not ring the Callback Extension.

## **Outgoing Call Rules**

If the Apply Incoming Call Rules to Provider option is *No*, use this section to define how the remote peer can place outbound calls through this PBX. Use the check boxes to assign which outgoing rules apply to this peer. Check the *Allow* box to apply the rule to this peer's inbound calls. Check *Deny* to block all calls that match that rule, such as blocking all long-distance numbers. To bypass a rule, leave both options unchecked. See Outgoing Calls for more information about each outgoing rule.

## **Connection Settings**

The Connection settings determine how the connection is established with this provider.

## Primary, Secondary, Tertiary Outbound Hosts

Some IAX providers use different hosts for registration, incoming, and outgoing calls. If your provider has multiple hosts for placing outbound calls, you can use the Primary, Secondary, and Tertiary Outbound Host fields to specify them. If these are left blank, the PBX will use the host specified in the Hostname/IP Address field for outgoing calls.

### **Incoming Password**

If this provider uses passwords for incoming call authentication and the password is different from your account password, specify it here. This condition is rare unless you are peering with a remote host and have explicitly configured the inbound and outbound passwords to be different.

## **Qualify Hosts**

Enabling this option will periodically send IAX POKE packets to the remote host and time their response. This latency measurement can then be seen in the System Status page.

## **Enable Jitterbuffer**

Enable this option to combat network problems. A jitterbuffer induces a small delay in call latency to make up for lost or delayed voice packets. You may want to enable the jitterbuffer if you have bad network connectivity

between Switchvox and the Provider.

## **Enable Trunking**

Trunking can be used to reduce the bandwidth used across multiple simultaneous calls; however, it must be enabled on both sides of the link. This is useful when you have two systems peered together over IAX with heavy voice traffic. If using commercial IAX services, check with the provider to see if it supports trunking. Enabling trunking on only one host will result in one-way audio on phone calls.

## **Use RSA Authentication for Outgoing**

Before setting this option, place your public RSA key on the remote host. You can retrieve this system's public RSA key in **System Setup> VOIP Providers**.

## Call Settings

The Call settings determine how to manage calls with this provider.

## **Provider Codecs**

Some providers support more than the normal codecs. These additional codecs may provide compression or other features that save bandwidth or lead to better call quality. However, you should check with your provider before selecting another codec, to be sure it is supported. If your Switchvox is a peer with another Switchvox, you can choose any audio codec.

For SIP providers, the video codecs H.263, H.263+, and H.264 are available. Video calling is not available over IAX. The following table describes the audio codecs. If you have Switchvox SMB and you entered a license key for the G729 Codec in **Machine Admin > Digium Addon Products**, you will also see G729 in the list of codecs.

Table 1: Audio Codecs
-----------------------

Codec	Bandwidth Usage (kbits/s)	Sound Quality	CPU Usage
ULAW	85	Great	None
ALAW	85	Great	None
G722	85	Excellent	Low
G726	54	Good	Low
SPEEX	33	Great	High
GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

## Fax Settings

The Fax settings determine how faxes are handled with this provider.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

**IMPORTANT:** We recommend that you only send T.38 faxes over SIP. We do not recommend sending faxes over IAX connections. Switchvox cannot necessarily recognize a fax over an IAX connection.

### **Error Correction Mode for G711 Faxes**

The default is to leave this feature enabled.

#### Minimum transfer rate for fax transmissions

Minimum transfer rate used during fax rate negotiation. The default minimum transfer rate is 2400.

### Maximum transfer rate for fax transmissions

Maximum transfer rate used during fax rate negotiation. The default maximum transfer rate is 14400.

## Adding a Peered PBX

You can set up two or more Switchvoxes so that you can make inter-Switchvox calls without employing a phone service provider. This is referred to as 'peering.' To peer with another Switchvox, you set up a new VOIP provider that is really the other Switchvox. The preferred protocol for peering is IAX, because there are fewer port-forwarding requirements than SIP.

**IMPORTANT:** If you are doing any video calling with Switchvox peers, you must use SIP. Video calling is not available over IAX.

The following instructions describe how to set up two offices, one in London and one in Boston. Both have their own Switchvox installation and broadband connection.

Assign extensions.

To make call routing easier, it's a good idea to make all London extensions start with one number, and Boston extensions start with a different number. In this example, we will make all London extensions 200-299, and all Boston extensions 400-499.

On each Switchvox, set up a VOIP provider that is the other Switchvox. Select System Setup > VOIP
 Providers, then add a new IAX or SIP provider. The settings are as follows. You need to click Show
 Advanced Options to access some of these settings.

In the new provider on the **London** Switchvox:

For SIP:

Provider Name: Boston PBX Your Account ID: pbx

For IAX:

Provider Name: Boston PBX
Provider Label: boston\_pbx
Your Account ID: london\_pbx

Your Password: pbxL

Hostname/IP Address: boston.com (or an IP address)

• Jabber Hostname: jabber.boston.com

In the new provider on the **Boston** Switchvox:

• For SIP:

Provider Name: London PBX Your Account ID: pbx

For IAX:

Provider Name: London PBX
Provider Label: london\_pbx
Your Account ID: boston\_pbx

Your Password: pbxB

Hostname/IP Address: Iondon.com (or an IP address)

· Jabber Hostname: jabber.london.com

In the new provider for both Switchvoxes, set the following options the same way:

- Callback Extension: Leave this blank.
- Default Fax Extension (SIP only): Leave this blank.
- DTMF Mode (SIP only): RFC2833
- Incoming Authentication Type (IAX only): MD5
- Caller ID Name and Number (IAX only): Leave these blank.
- · Host Type: Peer
- · Host is a Switchvox PBX: Yes
- · Treat system's users like local users: Yes
- · Apply Incoming Call Rules to Provider: No
- · Outgoing Call Rules: check the box for Allow Internal
- Qualify Hosts: Yes

Scroll down and click the Add Provider button to save your changes and add the new provider.

On each Switchvox, set up an Outgoing Call Rule to the other Switchvox. Select System Setup > Outgoing Calls, and click Add New Outgoing Rule.

In the new outgoing rule on the **London** Switchvox:

- Rule Name: Boston Switchvox
- · Is this rule final? Yes
- Pattern to match:

Number begins with the digits 4

The rest of the number must be between 2 and 2 digits in length. (Leave the remaining two fields blank.)

- Call through: Select the new VOIP Provider Boston Switchvox.
- Allow all existing extensions to use this rule? Yes

In the new outgoing rule on the **Boston** Switchvox:

- Rule Name: London Switchvox
- · Is this rule final? Yes
- Pattern to match:

Number begins with the digits 2

The rest of the number must be between 2 and 2 digits in length.

(Leave the remaining two fields blank.)

- Call through: Select the new VOIP Provider London Switchvox.
- Allow all existing extensions to use this rule? Yes

Click **Add Outgoing Rule**. A message is displayed that indicates the new outgoing rule was successfully created, and the new rule is included in the list of rules. Highlight the new rule and click the **Up** arrow until the new rule is a higher priority than the **Internal** rule.

4. If either Switchvox is behind a NAT router, you need to enable port forwarding on that router, and configure that Switchvox's Network Settings.

If the IP address of either Switchvox is in the following ranges, it is behind a NAT router:

192.168.0.0 - 192.168.255.255 10.0.0.0 - 10.255.255.255 172.16.0.0 - 172.32.255.255

The following ports need to be forwarded from your router to Switchvox:

For IAX traffic: UDP port 4569

For SIP traffic: UDP Ports 5060, and 10000-20000

To enable port forwarding, check with the documentation for your router. Documentation on port for-

warding for many common routers can be found at:

http://www.portforward.com.

To configure the Switchvox Network Settings, select **Machine Admin > Network Settings**, then click **Show Advanced Options**. Select Yes for **Allow Nat Port Forwarding**, and enter the external IP address of your router (click **Look Up External IP** if you do not know it). Click **Update Network Settings** to save your changes.

## Downloading and Renaming your RSA Key

RSA keys can be used for authenticating peers using IAX. This is not usually necessary when using commercial IAX providers, but it can be handy when peering multiple IAX-compatible PBXs together. Click **Download** to save the current RSA key. Then you can upload it to other phone systems when you create or modify an IAX Provider on that system.

You can rename this RSA key to something unique to this Switchvox PBX: Enter a new name in the **Rename RSA key** field, then click **Rename**. This can help you identify keys as you set up peered PBX systems. For example, you may name this key bostonoffice and another system's key londonoffice.

## **Setting an RTP Port Range**

These are the source ports used for establishing the RTP media pathway for SIP calls.

It is unlikely that you will need to change this setting. Only change it at your provider's request.

# **Outgoing Calls**

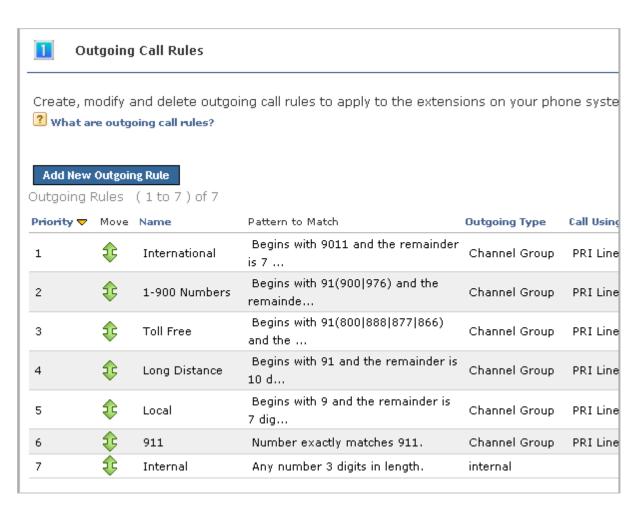
The Outgoing Calls tools let you handle all aspects of outgoing calls.

## **Outgoing Call Rules**

An outgoing call rule is a dialing pattern and associated routing information. For example, the **International** outgoing call rule is triggered when a user dials "9011" followed by 7 to 13 digits. Such calls are sent to the selected device (such as a provider or a Channel Group).

You can create multiple call rules that match the same pattern. In this case, if there is a pattern match, the rule that is highest in the priority list gets called first. If that rule fails, and it is not marked as a final rule, the next rule is called, and so on.

Switchvox is preconfigured with a set of outgoing call rules that serve most situations.



## **Creating a New Outgoing Rule**

To create a new call rule, click **Add New Outgoing Rule**.

Modify Outgoing Call Rule				
Rule Name:	Local			
Is this rule final?:  ? What is a final rule?	⊙Yes ○No			
Pattern to match:  What do these fields mean?	Number begins with the digits 91(760 619 858).  The rest of the number must be between 7 and 7 of Before connecting the call, trim 1 digits from the front, and then prepend the digits to the number.			
Call Through:	SIP Provider 💌			
SIP Provider:	Best Network 💌			
Save Changes				

#### **Rule Name**

Enter a unique name for this rule.

### Is this rule final?

If Yes (the default), when a call matches this rule but the route fails, then Switchvox does not look for additional matching rules and the call is not completed.

If *No*, when a call matches this rule but fails, Switchvox looks for another matching rule, and the call might be completed.

One reason that the routing might fail is if the extension is denied permission to use the outgoing rule.

### Pattern to match

Enter the call rule information in the boxes as follows:

#### Number begins with the digits $\Pi$ .

Enter any required precall numbers; for example, Switchvox normally requires dialing a 9 for an outside call. In this case, enter 9 in this box.

After the precall digit, enter the pattern you want to match. You can use any digit 0 through 9, pound (#), asterisk (\*), and the plus sign (+) to describe a pattern. If a call uses the pattern exactly as described, then the rule is applied.

You can also match multiple patterns in the same rule by using parenthesis and the pipe (|). To do this, put all of the possible patterns between parentheses and separate them with a pipe. The default rule **Toll Free** is a good example: 91(800|888|877|866). If a call begins with 91800, 91888, 91877, or 91866, then the rule is applied.

## The rest of the number must be between [] and [] digits in length.

Enter the length of a telephone number; for example, a local number must be exactly 7 digits. In this case, enter 7 in both boxes.

## Before connecting the call, trim [] digits from the front,

Enter the number of digits to trim when dialing; for example, enter 1 to remove the prepended 9 from the number sent to the provider.

## and then prepend the digits [] to the number.

Enter the digits to prepend to the telephone number; for example, enter 1 and the area code for the number.

### Call through

Where to route the call. The following options are available in the dropdown: SIP Provider, IAX Provider, Channel Group, Internal. If you select a Provider or a Channel Group, a dropdown is displayed that lets you select which provider or channel group to use.

## Allow all existing extensions to use this rule?

If Yes (the default), all existing extensions are set to Allow this rule, and the Default Extension Template is set to Allow this rule. If No, extensions and extension templates are neither Allowed or Denied this rule (the rule is blank). Note: If you have custom extension templates, you may want to set this rule to Allow or Deny.

Click **Add Outgoing Rule** to create the new rule. The Outgoing Calls page is redisplayed with a message indicating that the new rule was created.

## Modifying an Outgoing Rule

To modify an outgoing rule, click its **Modify** button. This displays the Outgoing Call Rule page showing the current definition for this rule. Make any changes that you want, then click **Save Changes**. Your changes are saved and the Outgoing Calls page is redisplayed with a message indicating that the rule was modified.

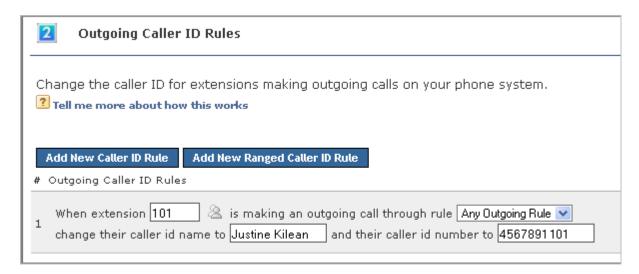
## **Deleting an Outgoing Rule**

To delete an outgoing rule, click its Delete button. Be sure that you do indeed want to delete the outgoing rule, then click **Yes, Delete Rule**. The outgoing rule is permanently deleted and the Outgoing Calls page is redisplayed with a message indicating that the rule was deleted.

## **Outgoing Caller ID Rules**

Outgoing Caller ID rules let you modify the outgoing caller ID name and number. Caller-ID rules can apply to a single extension, to any extension, or to a range of extensions.

If you want a rule to apply to any extension, enter any in the extension field.



If you have a ranged Caller-ID rule, you can derive that extension's correct caller ID by adding and prepending numbers to their extension. For example, if you have an extension block from 100 to 200, and a DID block of 800-555-1200 - 800-555-1300, you would use the following ranged rule:

When an extension is in the range from 100 to 200 and is making an outgoing call through rule Any Outgoing Rule, change their caller id to a number derived by adding 100 to their extension and prepending 8005551.

**Note:** Not all VOIP providers support this feature and some VOIP providers do not work properly with altered Caller IDs. Also, it is not possible to set Caller ID on regular analog phone lines.

## **Outgoing Call Diagnostics**

Outgoing Call Diagnostics can be used to determine which Outgoing Call Rule is being used and why.

If you click **Enable Outgoing Call Diagnostics Mode**, diagnostic information is read over the phone to all callers from all extensions. For example, If you were to Enable Outgoing Call Diagnostics Mode and then dial an internal extension (800), instead of connecting the call, the system would read back, "You have dialed, eight zero zero. This call will be sent to outgoing rule number seven. This rule will send the call to extension eight zero zero. Press one to attempt this call through rule number seven." As outgoing calls become more complex by trimming or appending digits, altering Caller ID, or matching multiple outgoing call rules, the diagnostic information provided will become more complex.

Listening to the diagnostic information and following the voice prompts provides insight into the path that calls are taking in the system to solve calling problems or unexpected call behavior taking place in the system. For example, the diagnostic may reveal that a call is using the wrong rule.

When using the Diagnostic mode, make sure the number you are dialing matches the pattern for the rule. If the pattern does not match, your outgoing call will not be connected using that rule. If the number you have dialed does match the pattern for the rule, make sure the extension you are placing the call from has permission to call through that rule (see Outgoing Call Rules).

# **Incoming Calls**

The Incoming Calls tools let you manage incoming call rules and routes.

## **Incoming Call Rules**

This section lets you create, modify, prioritize, and delete incoming call rules that apply to the extensions on Switchvox. Incoming call rules let you perform actions on calls based on who is calling you. These rules are system wide and happen before any of the extension call rules are triggered.



### **Block Number**

Lets you block callers from certain phone numbers or prefixes. For example, a rogue fax machine keeps calling your office with a caller ID number of 8005551234. You could add a Block Number incoming call rule that hangs up anytime this number calls you. You can block whole prefixes of numbers by choosing to block Prefix rather than Phone Number.

#### **Transfer Call**

Lets you transfer certain callers to a specific extension. For example, you could route all calls from a vendor whose phone number is 8005554321 to the extension for your purchasing department.

**Note:** All phone numbers used in an incoming call rule must be exactly as they appear in your caller ID. This number is usually the full 10-digit number, which includes the area code.

To create a new incoming call rule, select a type of rule from the **Create A New Incoming Call Rule** dropdown, then click **Add Rule**. This displays a new entry in the list (the new entry is highlighted).

## Type of number

Select Phone Number or Prefix from the dropdown.

## Number

Enter the numbers of the phone number or prefix that you want to affect.

#### Transfer to extension

If you created a Transfer rule, enter the extension the call should be transferred to.

### Action to perform

If you created a Block rule, select Play Busy Signal, Play Congestion, or Hangup from the dropdown.

## Times to apply this rule

Select Any Time, During Business Hours, Outside Business Hours, or Weekends Only from the dropdown. (For more information, see Time Frames.)

Click Save, and a message is displayed indicating that the new rule was successfully saved.

## **Modifying Incoming Call Rules**

To change an incoming call rule, modify the settings for the rule and click its **Save** button. The rule is saved and a message is displayed indicating the rule was successfully saved.

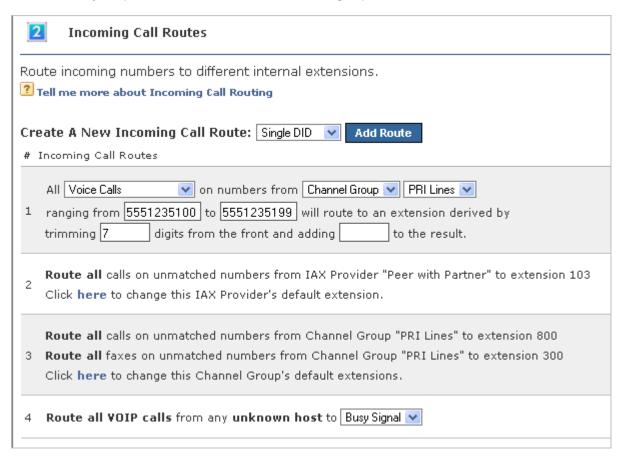
## **Deleting Incoming Call Rules**

To delete an incoming call rule, click its **Delete** button. The rule is deleted permanently and a message is displayed indicating the rule has been deleted.

## **Incoming Call Routes**

An Incoming Call Route maps numbers coming in on a particular Provider or Channel Group to a particular extension.

By default, Switchvox creates a default call route (the callback extension) for each interface as they are created (i.e., each provider has a rule). You can add new rules that allow for specialized call routing based on information sent by the provider or deduced from the channel group.



### Single DID

Lets you send a single phone number to a specific extension. For example:

Route number 8005551212 on numbers from Channel Group T1 to extension 802

### **Multiple DID**

Lets you send a range of phone numbers to an extension that is derived from the phone number itself. This is useful if each person has an outside number for directly dialing their extension. For example:

All Voice Calls on numbers from Channel Group T1 ranging from 8005551200 through 8005551300 will route to an extension derived by trimming 7 digits from the front and adding 0 to the result.

**Note:** Some SIP providers and most types of analog lines do not send the number dialed to the PBX. Ask your phone service provider if they support multiple incoming numbers or DIDs.

To create a new incoming call route, select the type of route from the **Create A New Incoming Call Route** dropdown, then click **Add Route**. This displays a new entry in the list (the new entry is highlighted).

## Type of Call

Select the type of calls to affect: Voice Calls, Faxes, or Voice Calls and Faxes.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

### From

Enter the source of the call, either a Provider or a Channel Group.

#### Number

If you created a Single DID route, enter the phone number to be routed.

## **Range of Numbers**

If you created a Multiple DID route, enter the first number in the range and the last number in the range.

## **Trimming**

If you created a Multiple DID route, enter the number of digits to trim from the front of the phone number.

## **Adding**

If you created a Multiple DID route, you can enter a value to add to the result or you can leave this blank.

## **Extension**

Enter the extension number to receive the call.

## **Modifying Incoming Call Routes**

To change an incoming call route, modify the settings for the route and click its **Save** button. The route is saved and a message is displayed indicating the route was successfully saved.

## **Deleting Incoming Call Routes**

To delete an incoming call route, click its **Delete** button. The route is permanently deleted and a message is displayed indicating the route has been deleted.

# **Voicemail Settings**

Select **System Setup > Voicemail Settings** to display the Voicemail Settings page. This lets you control any voicemail-related options.

## **Voicemail Notification Settings**

When voicemail is recorded for an extension, a notification email message is sent to the extension's email address. You can set a 'From' address for those email messages, and an SMTP server to use.

You may need to set this field to be a real email address before you are able to receive voicemail notification emails. Some strict mail servers will bounce emails appearing from the localhost.localdomain domain.

Customize the settings for voicemail notification emails			
Voicemail Notification Email "From" Address  ? What is this used for?	PBX@localhost.localdomain		
Outbound SMTP Server  ? Tell me more about this			
Outbound SMTP Username  ? About SMTP Auth			
Outbound SMTP Password  ? About SMTP Auth			
Save Email Settings			

#### Voicemail Notification Email "From" Address

Enter an email address. When you receive emails notifying you of new voicemail messages, they will come from this email address. You may need to set this field to be a real email address before you are able to receive voicemail notification emails. Some strict mail servers will bounce emails appearing from the domain localhost localdomain.

### **Outbound SMTP Server**

Enter the hostname of your SMTP server. Some ISPs or network admins require all outgoing email to pass through a specific SMTP server. If you are having trouble receiving voicemail notification emails, you may want to specify the hostname of your server here.

### **Outbound SMTP Username and Password**

Enter a username and password if your SMTP server requires it. It's probably safe to leave these blank, as many mail servers do not require them.

## **Running Voicemail Notification Diagnostics**

If you are having problems with voicemail notifications, enter an email address to send a test message, then click **Run Diagnostic Tool**. A diagnostic page is displayed that describes each step that was taken, and the result (passed or failed).

Voicemail Notification Email Diagnostics
Are users not receiving voicemail notification emails?
Email Address to Send a Test Email To
Run Diagnostic Tool

## **Voicemail Routing**

Voicemail Routing lets you handle several voicemail-related actions:

3 Voicemail Routing
During a user's voicemail greeting, callers may push the "*" key to log into the current v
During a user's voicemail greeting, callers may push the "0" key to exit and be forwarded to
If a user's voicemail box is full, callers will be forwarded to extension 🔃 😃
Save Settings

During a user's voicemail greeting, callers may push the "\*" key to log into the current voicemail box.

Check the box if you want this to be true. When the caller presses "\*," he or she is prompted for their password.

During a user's voicemail greeting, callers may push the "0" key to exit and be forwarded to extension [].

Enter an extension if you want this to be true. Click the people icon to select from a list of extensions. When the caller pushes "0," he or she has the option to accept and save their recording before being transferred to the extension.

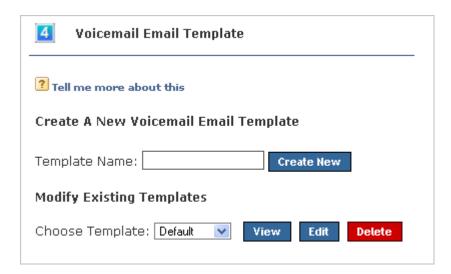
If a user's voicemail box is full, callers will be forwarded to extension [].

Enter an extension if you want this to be true. Click the people icon to select from a list of extensions.

## **Voicemail Email Templates**

Voicemail Email Templates let you customize voicemail notification emails. You can create multiple templates and tailor them to different types of email accounts. These templates are available to a user when he sets up his voicemail notifications (along with any templates he creates for himself).

You can use variables in your template that are substituted with real values when the email gets sent. For example, %VM\_NAME% in a template sends as the extension owner's first and last name. There is a full list of these variables on the pages where you create or edit a template.



## **Creating Voicemail Email Templates**

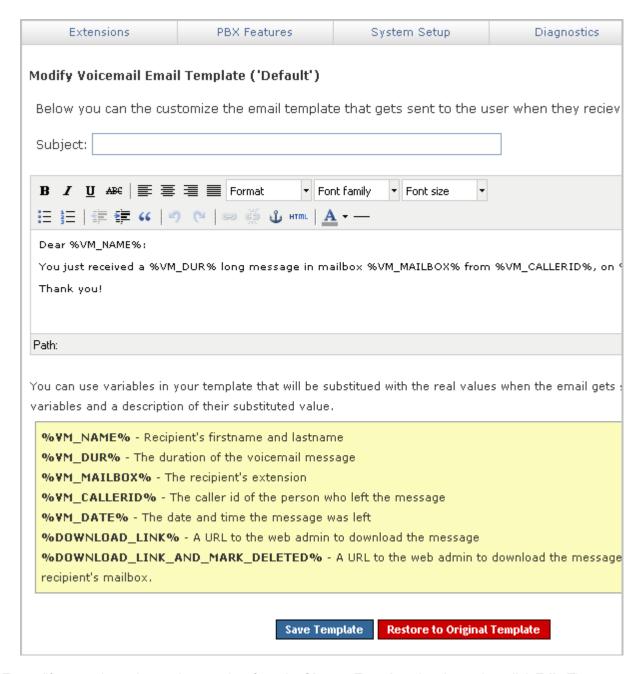
The Modify Template page is displayed with a sample template already entered. You can modify the subject and body (change it completely if you need to), using the list of possible variables at the bottom of the page. You can leave the subject blank, but you must put something in the body of the message.

To create a new template, enter a name in the Template Name field, then click Create New.

Click **Save Template** to save your changes and go back to the Voicemail Settings page. Click **Restore to Original Template** to restore this page to the original system default template.

## **Viewing & Modifying Voicemail Email Templates**

To see a 'preview' of a template, choose that template from the **Choose Template** dropdown, then click **View**. The template is displayed with sample values, in HTML and text-only.



To modify a template, choose that template from the **Choose Template** dropdown, then click **Edit**. The Modify template page is displayed. Click **Save Template** to save your changes and go back to the Voicemail Settings page. Click **Restore to Original Template** to restore this page to the original system default template.

## **Deleting Voicemail Email Templates**

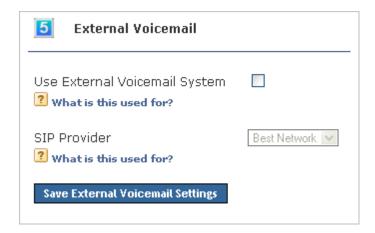
You can edit each of the system default templates, but you cannot delete them.

To delete one of your own custom templates, choose that template from the **Choose Template** dropdown, then click **Delete**. Be sure that you do indeed want to delete the template, then click **Yes, Delete Voicemail** 

**Email Template**. The template is deleted, and the Voicemail Settings page is displayed again. If any extension owners were using this template, those settings are changed to the 'Default' template.

## **External Voicemail**

If you want to use the voicemail system of an external provider, rather than this Switchvox's voicemail system, check this box and choose the SIP Provider whose voicemail system you want to use.



In Switchvox SMB, the SIP Provider can be a peered Switchvox (for more information, see Adding a Peered PBX). To use a peered Switchvox as the External Voicemail Provider:

- Be sure both Switchvoxes are running SMB version 4.0 or later.
- On the server Switchvox (the External Voicemail Provider):
- set up a new SIP provider that is the client Switchvox. On the new provider, show Advanced Options, select Yes for Host is a Switchvox PBX, and enter the **Jabber Hostname** for the client Switchvox.
- Be sure that each extension that exists on the client Switchvox also exists on the server Switchvox as a Virtual extension. For help managing this, see Creating & Updating Multiple Extensions.
- On the client Switchvox:
- Set up a SIP provider that is the server Switchvox.
- Check the box to Use External Voicemail System, and select the server Switchvox as the SIP provider.
- Set up an outgoing call rule so that calls to the voicemail-access extension (by default, this is 899) are sent to the server Switchvox. (For details, see Outgoing Call Rules.)
- Remember, when checking their voicemail users should go to the Switchvox User Suite on the server Switchvox.

## **Custom Voicemail Message Length**

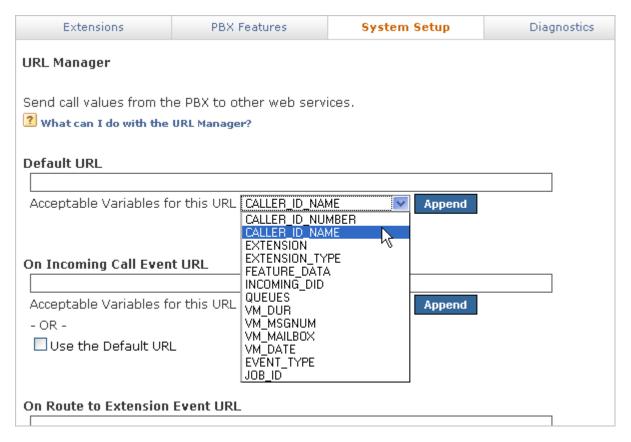
You can specify the number of seconds that it takes for Switchvox to 'give up' on a caller if they reach your voicemail but don't leave a message. For example: you specify 10 seconds, someone calls and reaches your voicemail, but they don't start talking, then Switchvox hangs up and does not save those 10 seconds of silence. Setting a number of seconds here should help reduce clutter in Mailboxes and on disk.

You can also set a number of seconds to let the caller speak, and after that the call is disconnected. This is useful if you are concerned about storage capacity.



# **URL Manager**

The URL Manager lets you set up Switchvox to interact with external Web applications. When certain events occur in Switchvox, it can submit information about the event to a Web application, and respond to the XML that is returned.



## **Default URL**

You can enter a Default URL so that you do not have to enter a full URL for each event. But, you can choose to use the default URL or not, so you still have flexibility on each event.

## **Events**

There are some Switchvox events that trigger an HTTP POST.

## On Incoming Call

Triggered at the beginning of an incoming call. This event applies to calls received from VOIP providers, any channel groups, as well as calls between extensions. It is common for requests of this type to return XML for routing the call or altering the Caller ID.

### On Route to Extension

Triggered when a call hits an extension. This event can occur multiple times for each call. For example, a call may hit an IVR, a queue, and then a phone extension. XML can be returned to route calls differently.

**WARNING:** Do not create routing loops. For example, if a call to extension 333 results in a request that sets transfer\_extension to 333, then Switchvox loops endlessly. Do not set transfer\_extension in the response to this event unless it is absolutely necessary.

#### On Call Answered

Triggered when a call is answered. This event applies to calls from external numbers, and from all Switchvox extensions.

#### On New Voicemail

Triggered when a voicemail message is left for an extension (including a message forwarded from another extension). This event requests the voicemail URL. **Note:** Setting XML values in the response to this request results in undefined behavior. Return empty XML for this request.

### On Checked Voicemail

Triggered when an extension checks its voicemail. This event could be used in conjunction with the On New Voicemail Event to track new voicemail messages and implement an escalation policy. Note: Setting XML values in the response to this request results in undefined behavior. Return empty XML for this request.

## On Agent Logged In

Triggered when a queue member logs into a queue.

### On Agent Logged Out

Triggered when a queue member logs out of a queue.

## On Call Hangup

Triggered when a call is hung up. This event can be used in conjunction with the JOB\_ID variable to record the completion of calls that invoked previous URL events. Note: Setting XML values in the response to this request results in undefined behavior. Return empty XML for this request.

## On Outgoing Call

Triggered when a call is created. This event applies to calls to external numbers and all Switchvox extensions.

## **Variables**

Switchvox events can submit various information about calls. To do this, you use variables in the URL. Not every variable is available in every event (check the Acceptable Variables dropdown on each event), so be prepared to handle an empty variable. Here is the complete list of all of the variables for all event types:

## CALLER\_ID\_NUMBER

Phone number from caller ID, including area code.

### CALLER\_ID\_NAME

Name from caller ID.

## **EXTENSION**

Internal extension associated with the event.

## **EXTENSION\_TYPE**

The type of the extension associated with the event. For the list of extension types, see Extension Types.

## **FEATURE DATA**

The data entered after a feature code. For example, in \*62204, you have an EXTENSION of 62, an EXTENSION TYPE of feature monitor, and a FEATURE DATA of 204.

## INCOMING\_DID

Phone number dialed by an external caller. May be empty for internal calls, or calls over certain analog lines that do not provide the DID.

### VM DUR

The duration of the voicemail message, in seconds.

## VM\_MSGNUM

The unique message number for this voicemail message.

## **VM MAILBOX**

Extension associated with the event's voicemail box.

## VM\_DATE

The date and time of the voicemail message.

#### OUFUES

A comma separated list of the queue account\_ids being logged in to or logged out of. In the case where the user is being logged out of all queues, the list will only contain the list of queues the user is a member of.

## **EVENT TYPE**

Event type that has been triggered. This is useful for the Default URL, which receives POSTs from any of the event types.

#### JOB\_ID

Unique ID of the call. *Note:* This string of digits is not a numerical value: 99.100 is not equal to 99.1. This variable is useful for correlating multiple events (and also certain IVR requests).

## XML Response

For some events, the remote web application can return XML that instructs Switchvox to perform operations on the call. Here is a sample of an XML response:

All of the tags inside the call\_info block are optional. The following XML tags can be used:

### callerid num

Set Caller ID Number to this value.

### callerid\_name

Set Caller ID Name to this value.

## transfer\_extension

Transfer call to the extension of this value. **WARNING:** It is possible to break Switchvox by creating routing loops.

## display\_url

Display this URL in the Switchboard as a clickable icon. For example: the remote web application might lookup the caller's phone number in your company's user database, then return the URL of the caller's user record.

## continue\_outgoing\_reporting

Continue reporting any subsequent outgoing calls for this JOB\_ID. For example, if a call is made to a queue, the event will continue when a queue-member's extension is dialed. The value of this tag must be **1** or **0**.

# **Diagnostics**

The Diagnostics tools give you different views of the call activity in Switchvox, and help you evaluate potential problems.

# **System Status**

This page shows you the overall health of Switchvox's connected devices including the VOIP Providers, SIP Phones, Hardware Devices, and Mailbox Disk Usage.

## **VOIP Providers**

This section lists the following information for each provider:

- Name of the provider as entered when the provider was created
- Host that Switchvox connects to when registering and sending calls
- o Account ID
- Extension that incoming calls are sent to by default
- State of the connection



The state of the connection should be **Registered** when the system is operating properly. If it is not, outgoing calls will fail and incoming calls cannot be received. Here are some of the most common reasons a VOIP provider is in a non-registered state, and a resolution:

- VOIP provider has just been entered and the registration has not finished yet. Wait a few minutes and then refresh the page.
- Username or password are incorrect. Check to make sure you have entered them correctly.
- Firewall is not allowing the communication between Switchvox and your provider to flow freely. When you use a SIP-aware firewall, keep in mind that it recognizes the data being sent back and forth. A normal router does not require any special configuration, but a SIP-aware firewall does. Contact the manufacturer of the firewall for details on setting up any necessary routing rules.

## **SIP Phones**

This section lists all the phone-extensions and information about their phones.

SIP Phor	nes		
Extensio	n マ Caller ID	User Agent	IP Address
101	Justine Kilean	PolycomSoundPointIP-SPIP_430-UA/2.1.0.2708	10.10.3.14
102	Jane Doeson	Aastra 9133i/1.4.1.2000 Brom Callotrl/1.5.1.0 MxSF/v3.2.6.26	10.10.3.21
103	John Smith	PolycomSoundPointIP-SPIP_300-UA/2.1.2.0078	10.10.3.22
104	Gilbert Brown	Grandstream GXP1200 1.1.6.16	10.10.3.69

## **Phone State**

The **State** column indicates the state of the extension's phone on Switchvox.

- Registered: this indicates that the phone is configured properly and is on the network.
- **Timeout:** this indicates that the phone was registered at some point, but the server no longer recognizes it and does not send calls to this phone anymore.
- Unreachable: this indicates that the phone has not yet registered with Switchvox.

If a phone is not **Registered** but it should be:

- Make sure that your phone is set to register every 120 seconds. (Some phones are by default set to only register once.) See your phone's user manual for more information about this setting.
- Make sure your phone is configured correctly. Your phone must know Switchvox's IP address, and it
  must have the same password as the "Phone Password" for the extension. For more information on
  configuring your phone, please see your phone's manual or call your phone's technical support.
- Make sure your phone is ON, the Ethernet cord is plugged into the phone and the correct Ethernet jack, and the phone is on the same LAN as Switchvox (not on a separate partitioned LAN subnet).
- Make sure there isn't a firewall between your phone and Switchvox that could be blocking your phone from accessing Switchvox.

## Diagnose a Problem

If a SIP phone is not in a Registered state, click its **Diagnose** button to run a diagnostic tool for that phone.

## **External Phones**

Phones are considered "External" if they are not on the same network as Switchvox. This generally means that they have to connect to Switchvox over the Internet, or they are located on a separately partitioned network. External phones cannot go through the full diagnostics because Switchvox does not have direct access to these phones.

## Finding the IP Address of your SIP Phone

To run the diagnostic tool, you need to know the IP address for your phone.

## Polycom Soundpoint IP 300/301/500/501/600/601

Each phone is set up to obtain its IP address via DHCP. You can find the IP address of a Polycom phone by pressing the **Menu** button on your phone. Then, on the LCD screen of your phone highlight "Status..." and press select and then highlight "Network..." and press select again, then highlight "TCP/IP Parameters..." and press select. Now find the line that starts with "IP:" and write down the IP Address.

## Sipura ATA

Connect a phone to the jack labeled "Phone 1" on the front of your ATA. Then on the phone dial "\*\*\*\*" you will then hear a voice say "Configuration Menu". Then dial "110#" and it will read back the ip address of your ATA.

## **Hardware Devices**

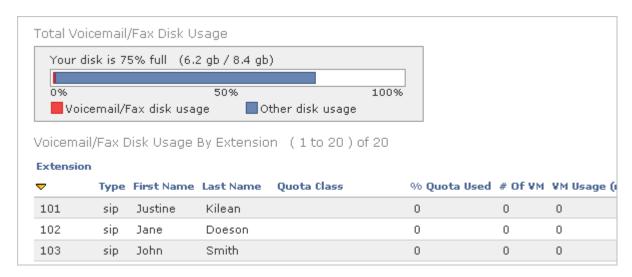
This section lists all the channels on the hardware devices installed for Switchvox, and an alarm state for each. The state of these channels, if properly connected, should be **No Alarm**.

If any other alarm state is shown (a RED Alarm or a Loss of Signal), check to make sure that the device is plugged into the correct line, and that you have the appropriate signaling type selected.



## Voicemail/Fax Disk Usage

This section shows an overview of the disk usage, and lists each extension and its disk usage.

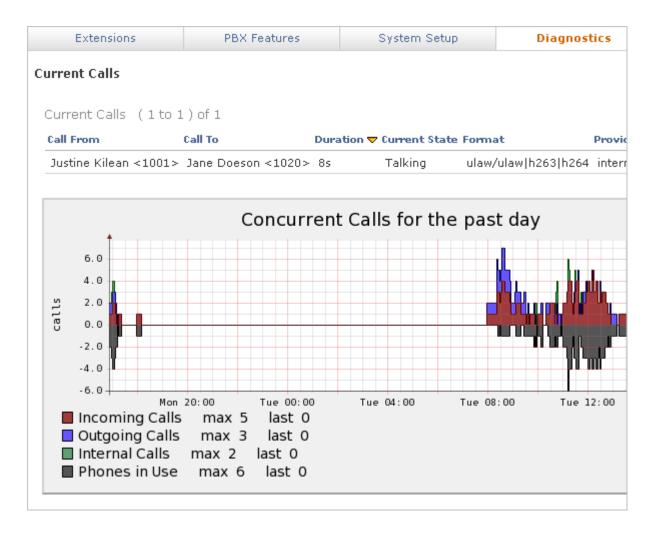


For information about disk-space quotas, see Voicemail/Fax Mailbox Quota (MB).

## **Current Calls**

This is a list of all current calls. It shows the person who called, the person who took the call, the duration of the call (in seconds), the current state of the call, and the provider.

To hang up on a call, click **Hangup**. Be sure that you do indeed want to hangup the call, then click **Yes, Hangup**.



## **Queue Status**

The Queue Status tool provides information for each call queue in Switchvox: Overall statistics, member status and statistics, and calls waiting in the queue.

## **Statistics**

Queue Statistics show you how a queue has been functioning.

## **Current Calls Waiting**

Calls waiting in the queue.

### **Completed Calls**

Calls answered by a queue member.

## **Abandoned Calls**

Calls in the queue, but hung up before being answered by a queue member.

### **Redirected Calls**

Call was redirected out of the call queue, or the caller pressed 0 key, or the call reached maximum timeout, or the call was sent the maximum number of times, or the call queue was at its maximum length.

## **Longest Queue Length**

Largest number of calls waiting in the queue at one time.

## **Average Entry Position**

Average position when calls first enter the queue.

## **Longest Wait Time for Completed Calls**

Longest time that a call was in the queue and was answered by a queue member.

## **Longest Wait Time for Abandoned Calls**

Longest time that a call was in the queue but hung up before being answered by a queue member.

## **Average Wait Time for Completed Calls**

Average time that a call was in the queue and was answered by a queue member.

## **Average Wait Time for Abandoned Calls**

Average time that a call was in the queue and hung up before being answered by a queue member.

## **Average Talk Time**

Average time that queue members spent talking with callers from the queue.

## **Member Status**

This shows each member of the queue, their extension and queue-status (logged in, logged out, or logged in but paused), who they are talking to, and the amount of time they spent on their last call. It also includes statistics: the number of calls they have taken today, and the average time they spent talking on those calls.

## **Current Calls Waiting**

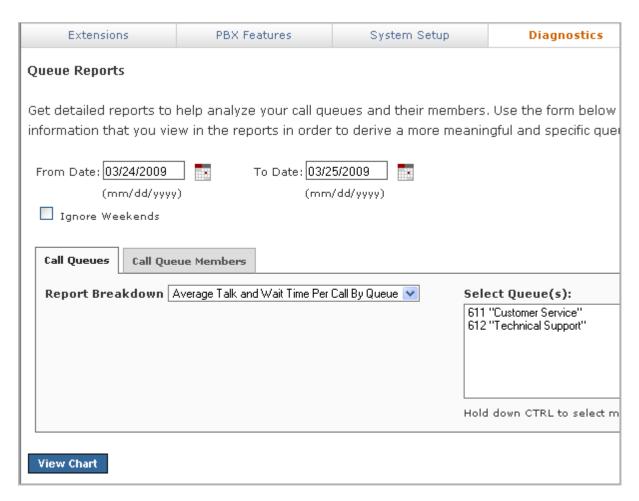
This shows the calls that are currently waiting in the queue, including the current position, caller ID, how long they have been waiting in the queue, and the position in which they entered the queue.

## **Past Statistics**

Click to view past statistics for yesterday, the past 2, 7, 10, or 30 days, or the past year. Past statistics don't include member status, or calls waiting.

# **Queue Reports**

This lets you generate a report about call activity for a queue (or a set of queues), or a queue member. Queue reports are generated for the date-range and criteria that you specify.



Check the 'Ignore Weekends' box if your queue is not active on weekends and calls on those days can be ignored.

To view the report, set up your criteria then click View Chart.

## **Call Queue Reports**

Queue reports can be broken down in different ways, based on talk and wait times, date, day of the week (e.g., Monday), or hour of the day.

Select the report breakdown, then select at least one queue to report on. To select multiple queues, hold down the CTRL key and click on the queues.

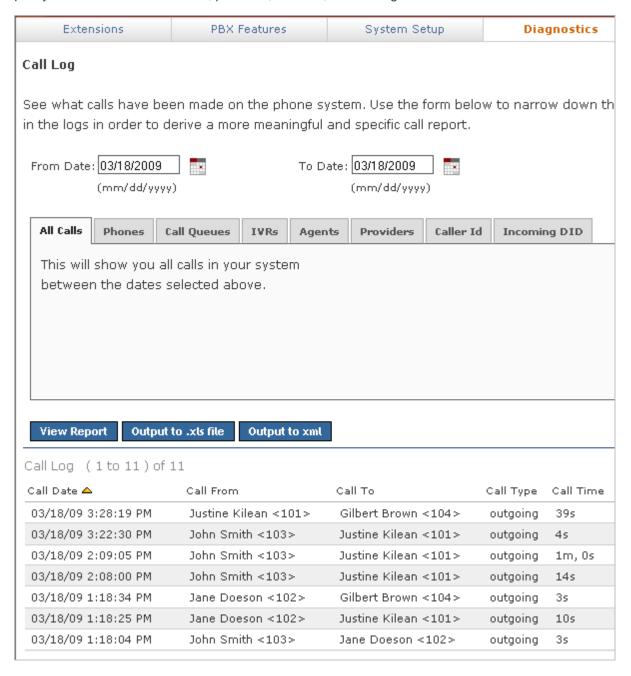
## **Call Queue Member Reports**

Queue Member reports can be broken down in different ways, based on talk time, member, date, and duration.

Select the report breakdown, then select at least one member to report on. To select multiple members, hold down the CTRL key and click on the members.

# **Call Log**

This lets you generate a log of call activity. Call logs are generated for the date-range and criteria that you specify. Criteria include extensions, providers, caller ID, or Incoming DID.



## **View Report**

Open the report in the browser window.

## Output to .xls file

Download a .xls file that you can open with an application such as Microsoft Excel.

# **Call Reporting**

This lets you generate a report about call activity. Call reports are generated for the date-range and criteria that you specify, and include the information that you specify.



Check the 'Ignore Weekends' box if your organization is not active on weekends and calls on those days can be ignored.

Call reports can be broken down in different ways, based on cumulative numbers, date, account, hour of the day and day of the week (e.g., Monday), extensions, providers, or the incoming DID.

You can select what fields of data you want to include in a report, depending on what calls you include in the report. The fields include total calls incoming and/or outgoing, talking time and call duration, and average times. Call duration means the entire call, including time spent in an IVR or waiting in a queue. Talk time means the time when the callers were actually talking to each other.

To generate the report, you can choose one of several viewing options.

## **View Report**

Open the report in the browser window.

### Output to .xls file

Download a .xls file that you can open with an application such as Microsoft Excel.

## Output to xml

Download an xml file.

## **Chart Report**

Open the report as a graphical chart in the browser window.

# **Scheduled Reports**

Scheduled Reports is similar to Call Reporting, but here you can set up a report to run at a certain time with a particular output, and the report is sent to you (via email) based on those criteria.

To create a new report, click **Create a new Scheduled Report**. Name the report, and then set up the schedule, parameters, and delivery option.

To edit an existing scheduled report, click **Modify** on the same line as the report.

The **Schedule Information** indicates the dates during which this report is active. Enter the date that you want to activate the report (the first date it should run), and the date that you want to deactivate it (the last date it should run). If you want the report to keep running indefinitely, indicate *No end date*.

The **Recurrence Pattern** indicates how often the report should run during the time it is active. You can control how often the report runs, from hourly to once a year. The start date of the report indicates the beginning day and time of the report, based on the recurrence pattern.

The Report Parameters let you define what fields of data to include in the report.

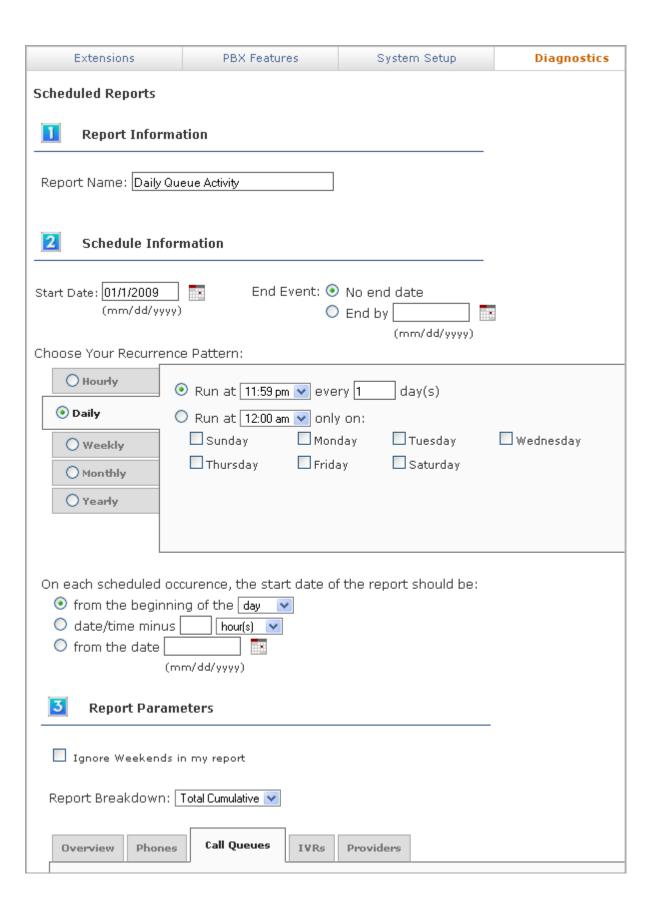
The **Delivery Options** let you define how you want to receive the report:

## **Output Type**

HTML, graphic chart, or XML.

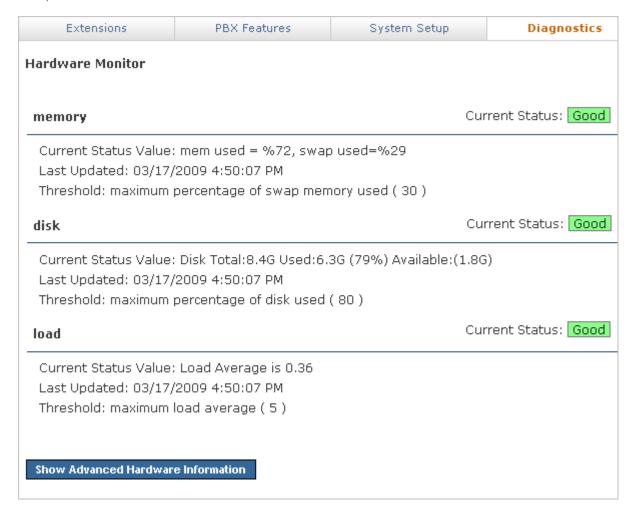
### **Delivery Method**

The delivery method available is Email. Enter your full email address.



## **Hardware Monitor**

This offers information about the health of the Switchvox server. As long as the Current Status for each item is *Good*, the hardware is fine.



If a field is over threshold, it needs immediate attention. The following are some suggestions for handling a problem:

#### memory

If **memory** is over threshold, one option is to turn the system off (wait a few moments), and then turn it back on. This normally frees up the memory usage. Another option is to install more RAM. You can also attempt to identify the process that is overloading the memory (call recording, concurrent calls, concurrent conference calls, etc.), and stop that process.

### disk

If **disk** is over threshold, check voicemail, fax, call recordings, and backups to find out what is taking up the disk space.

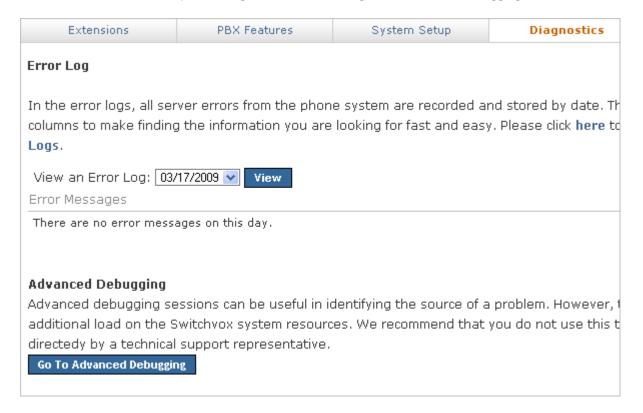
## load

If **load** is over threshold, check **Diagnostics > Current Calls** to see if the system is hitting a peak (spike) amount of concurrent calls.

# **Error Log**

This section provides tools for our customer service representatives to help you with any problems that you might have with Switchvox. You do not need to use these tools unless we ask you to.

The tools included here are simple error logs, advanced error logs, and advanced debugging sessions.



# **Advanced Debugging**

Advanced debugging sessions can be useful in identifying the source of a problem. However, these sessions place an additional load on the Switchvox system resources. We recommend that you do not use this tool unless you are being directed by a technical support representative.

These debugging sessions collect detailed PCAP, PRI, or ACLI information to help you see exactly what is happening in Switchvox.

# **TSR Report**

This section provides tools for our customer service representatives to help you with any problems that you might have with Switchvox. You do not need to use these tools unless we ask you to.

A TSR an be downloaded and sent to our technical support department, and Tech Support Access lets us access your machine to help diagnose a problem.

Extensions	PBX Features	System Setup	Diagnostics
------------	--------------	--------------	-------------

## TSR Report



## Download a Technical Service Report

To download a TSR (Technical Service Request) Report please click the "Download TSR Report problems with your PBX you can send a TSR Report to customer service so they can better dimight have.

## Download TSR Report



## **Enable Tech Support Access**

Please do not enable tech support access unless requested by a tech support representative Note: After tech support access is enabled you will also need port 22 forwarded on your fire

Disable Tech Support Access

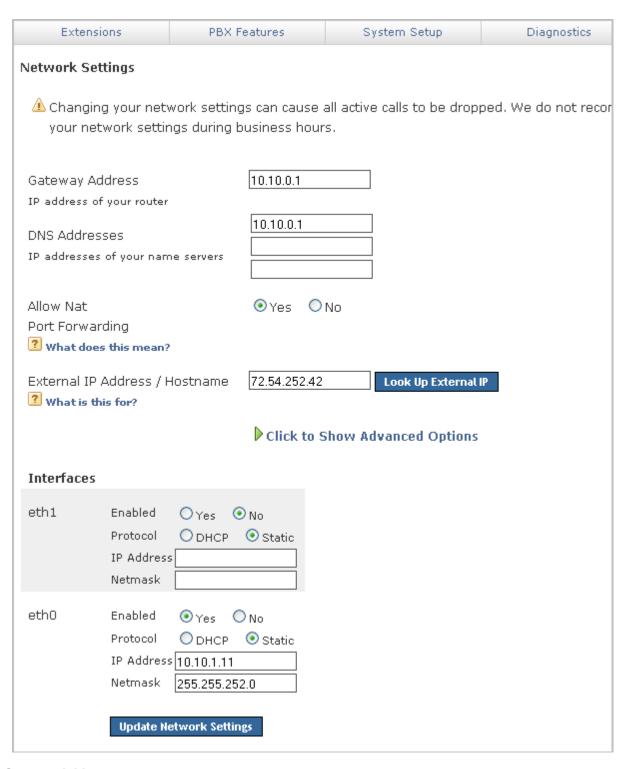
# **Machine Admin**

The Machine Admin tools let you manage your server's network settings, make and restore backups, apply updates, set the system clock, and more.

# **Network Settings**

This lets you configure the networking information for Switchvox. These settings are similar to settings you would set on any other computer on your network.

Enter the information, then click **Update Network Settings**.



## **Gateway Address**

This is the IP address of the machine to which Switchvox sends outbound traffic. It is typically the address of your router.

#### **DNS Addresses**

You can enter up to three DNS addresses. Switchvox uses these addresses to translate any host/domain names into IP addresses.

### Allow NAT Port Forwarding

Select Yes to allow NAT Port Forwarding to Switchvox. This option is useful if you need to handle calls going to and coming from an external network, and Switchvox is behind a router that performs NAT.

On your router, you must forward the following ports to your Switchvox server. Also, in **Machine Admin > Access Control**, you must have a rule that allows the appropriate traffic for an external network. See Access Control for more information.

Protocol	TCP/UDP	Port(s)	Description
IAX	UDP	4569	Signaling port needed for phones outside of your network.
SIP	UDP	5060	Signaling port needed for phones outside of your network.
SIP	UDP	10000-10500	RTP audio ports needed for phones outside of your network.
	TCP	80	HTTP port for remote web voicemail access
	TCP	443	HTTPS port for remote web admin access
	TCP	5222 & 843	Ports for using the Switchboard remotely
	TCP	5269	Port for remote XMPP (Jabber/chat) access

### **External IP Address**

Your external IP is the public IP Address of your router. If you are not sure what your public IP is, click **Look Up External IP** to automatically find it.

Note: SIP phones that are outside of your network must use this external IP for registration.

You cannot use this feature if your ISP does not give you a static external IP address. Check with your ISP to make sure that the public IP you enter here will not change. If you do not have a static public IP, you will only be able to use phones from within your local network.

#### **Interfaces**

The IP and Netmask address. You set these addresses when you installed the Switchvox software, and you do not normally need to change this information after installation.

### **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

#### Hostname

You can set a hostname for Switchvox (e.g., pbx.example.com) if you have a DNS setting. This may alleviate problems with delivering emails through particularly strict SMTP servers.

**IMPORTANT:** This option also sets the web server's SSL key. This may help prevent warnings from your browser when logging into Switchvox. In most cases it is best to leave this option unchanged.

#### Jabber Hostname

You can set a hostname for the Jabber server in Switchvox (e.g., jabber.example.com) if you have a DNS setting.

This is important if you are peering Switchvoxes and want to use functions such as Presence or the Chat Panel.

#### IP ToS

This lets you set the ToS or DSCP field in VoIP packets sent by Switchvox. This field can be used by firewalls and switches to distinguish specific types of traffic to apply QoS rules, such as favoring all voice traffic for better quality. Setting the ToS field isn't a requirement for prioritizing VOIP traffic on your router, it's just one way to identify VOIP traffic. If you're not explicitly prioritizing this field in your network equipment, changing this option will have no effect on your VOIP quality.

For more information see the following resources:

RFC 2474 - "Definition of the Differentiated Services Field (DS field) in the IPv4 and IPv6 Headers", Nichols, K., et al, December 1998.

IANA Assignments, DSCP registry, Differentiated Services Field Codepoints: http://www.iana.org/assignments/dscp-registry

#### **Custom SSL Certificate**

Use this field to add a Custom SSL Certificate. This option can be used to prevent your browser from popping up a security warning when you access the Admin Suite. After you have purchased an HTTPS certificate from an online certificate authority, it can be uploaded into these sections in order to allow Switchvox to use it for encrypting web traffic.

Another option for this section is if your organization creates its own web certificates for encrypting various intranet traffic.

If you are unsure about HTTPS certificates, it is safe to leave this section blank.

#### **Additional Local Networks**

Use this section to list any additional networks that are not separated from Switchvox by a NATing router.

This option is only needed if:

- Switchvox is behind a router that performs NAT.
- · You wish to use soft phones from outside your network.
- There is more than one network (such as a remote office VPN) connected to the local net.

Networks can be listed in either CIDR (e.g., 192.168.0.0/24) or netmask (e.g., 192.168.0.0/255.255.255) notation.

### **SNMP Settings**

SNMP (Simple Network Management Protocol) is used to monitor devices on an IP network. If you are not already using SNMP, then you can safely ignore these settings.

If you are already using SNMP, you can monitor the OIDs that the Switchvox agent provides.

**IMPORTANT:** Be sure to allow SNMP traffic in Switchvox Access Control. (By default, SNMP traffic is not allowed).

### SNMP v1/v2c Community Name

Enter the Community Name for the SNMP community to which Switchvox should belong.

#### SNMP v3 User Name

Enter the SNMP User Name, if you are using SNMP version 3.

### **SNMP v3 Password**

Enter the SNMP Password, if you are using SNMP version 3.

### **Switchvox OIDs**

The Switchvox prefix is .1.3.6.1.4.1.22736.10

1	Switchvox	Information
	OWILCITYON	miomiation

.1.3.6.1.4.1.22736.10.1.1	software version
.1.3.6.1.4.1.22736.10.1.2	max concurrent calls
.1.3.6.1.4.1.22736.10.1.3	used feature packs
.1.3.6.1.4.1.22736.10.1.4	max feature packs
.1.3.6.1.4.1.22736.10.1.5	used extensions
.1.3.6.1.4.1.22736.10.1.6	max extensions

### .2 VOIP Providers

.1.3.6.1.4.1.22736.10.2.1	number of VOIP providers
.1.3.6.1.4.1.22736.10.2.2	number of state 'Ok' VOIP providers
.1.3.6.1.4.1.22736.10.2.3.1.1	first VOIP provider id
.1.3.6.1.4.1.22736.10.2.3.1.2	first VOIP provider type
.1.3.6.1.4.1.22736.10.2.3.1.3	first VOIP provider name
.1.3.6.1.4.1.22736.10.2.3.1.4	first VOIP provider host
.1.3.6.1.4.1.22736.10.2.3.1.5	first VOIP provider account id
.1.3.6.1.4.1.22736.10.2.3.1.6	first VOIP provider callback extension
.1.3.6.1.4.1.22736.10.2.3.1.7	first VOIP provider latency
.1.3.6.1.4.1.22736.10.2.3.1.8	first VOIP provider state
.1.3.6.1.4.1.22736.10.2.3.[2-xxx].[1-8]	details for each additional VOIP provider

### .3 Hardware Status

.1.3.6.1.4.1.22736.10.3.1.1	memory used
.1.3.6.1.4.1.22736.10.3.1.2	memory total
.1.3.6.1.4.1.22736.10.3.1.3	swap used
.1.3.6.1.4.1.22736.10.3.1.4	swap total
.1.3.6.1.4.1.22736.10.3.1.5	last update (memory/swap)
.1.3.6.1.4.1.22736.10.3.2.1	disk used
.1.3.6.1.4.1.22736.10.3.2.2	disk total
.1.3.6.1.4.1.22736.10.3.2.3	last update (disk)
.1.3.6.1.4.1.22736.10.3.3.1	load

### .4 Phone Status

.1.3.6.1.4.1.22736.10.4.1	number of SIP phones
.1.3.6.1.4.1.22736.10.4.2	number of reachable SIP phones
.1.3.6.1.4.1.22736.10.4.3.1.1	first phone id
.1.3.6.1.4.1.22736.10.4.3.1.2	first phone extension
.1.3.6.1.4.1.22736.10.4.3.1.3	first phone caller id
.1.3.6.1.4.1.22736.10.4.3.1.4	first phone user agent
.1.3.6.1.4.1.22736.10.4.3.1.5	first phone ip address

.1.3.6.1.4.1.22736.10.4.3.1.6 .1.3.6.1.4.1.22736.10.4.3.[2-xxx].[1-6]	first phone state details for each additional phone
.5 Current Calls	
.1.3.6.1.4.1.22736.10.5.1 .1.3.6.1.4.1.22736.10.5.2 .1.3.6.1.4.1.22736.10.5.3 .1.3.6.1.4.1.22736.10.5.4 .1.3.6.1.4.1.22736.10.5.5 .1.3.6.1.4.1.22736.10.5.6 .1.3.6.1.4.1.22736.10.5.7 .1.3.6.1.4.1.22736.10.5.8 .1.3.6.1.4.1.22736.10.5.9 .1.3.6.1.4.1.22736.10.5.10 .1.3.6.1.4.1.22736.10.5.11 .1.3.6.1.4.1.22736.10.5.11	total current calls number of calls in state 'agent_login' number of calls in state 'agent_logout' number of calls in state 'checking_voicemail' number of calls in state 'conference' number of calls in state 'directory' number of calls in state 'faxing' number of calls in state 'intercom' number of calls in state 'ivr' number of calls in state 'leaving_voicemail' number of calls in state 'monitoring' number of calls in state 'parked'
.1.3.6.1.4.1.22736.10.5.13 .1.3.6.1.4.1.22736.10.5.14 .1.3.6.1.4.1.22736.10.5.15 .1.3.6.1.4.1.22736.10.5.16 .1.3.6.1.4.1.22736.10.5.17	number of calls in state 'queued' number of calls in state 'receiving_fax' number of calls in state 'ringing' number of calls in state 'talking' number of calls in state 'unknown'
.1.3.6.1.4.1.22736.10.6.1 .1.3.6.1.4.1.22736.10.6.2 .1.3.6.1.4.1.22736.10.6.3.1.1 .1.3.6.1.4.1.22736.10.6.3.1.2 .1.3.6.1.4.1.22736.10.6.3.1.3 .1.3.6.1.4.1.22736.10.6.3.1.4 .1.3.6.1.4.1.22736.10.6.3.1.5 .1.3.6.1.4.1.22736.10.6.3.1.6 .1.3.6.1.4.1.22736.10.6.3.1.6 .1.3.6.1.4.1.22736.10.6.3.1.6	number of devices number of state 'Ok' devices first device id first device name first device channel first device channel group first device signalling first device state details for each additional device
.1.3.6.1.4.1.22736.10.7.1 .1.3.6.1.4.1.22736.10.7.2	expire date days left

### **Other OIDs**

The Switchvox agent provides some additional OIDs that are not specific to the Switchvox application.

.1.3.6.1.4.1.795.14.1.100.5.0	Adaptech RAID card: overall status
.iso.org.dod.internet.mgmt.mib-2.system	Linux system OIDs. You can use any OID under this.
.iso.org.dod.internet.private.enterprises.ucdavis	Linux system OIDs. You can use any OID under this.

# **Access Control**

This lets you allow network access to Switchvox services based on IP address and Netmask.

The default action is to deny access, so if you don't allow a service for a network, then the network is denied access to that service.

Extensions	PBX Features	Syste	m Set	tup			Diag	nostic	)S
ccess Control									
How do I use this page?									
access Control Rules (	1 to 2 ) of 2			ı		í			1
Rule Name ▼	Network	Web Admin Portal	Web User Portal	Admin API	User API	Printing	IMAP	XMPP Server (Jabber)	WALTER CHANGE AND ADDRESS OF
All Networks	0.0.0.0/0								
Local Network	10.10.0.0/22	V	¥	V	V	V	V	V	-
Rule Name:									
Network:									
Allowed Services:	☐ Web Admin Portal								
	Web User Portal								
	Admin API								
	User API Printing								
	☐ IMAP								
	MPP Server (Jabber	)							
	XMPP Client (Jabber)								
	SIP								
	□ IAX								
	□ NTP								
Add	d Rule Cancel								

### **Access Control Rules**

The Local Network rule by default allows all traffic for all services. The All Networks rule allows traffic for various services depending on your Switchvox configuration. You can modify the services allowed for the Local Network and All Networks rules, but you cannot change the name or the network.

### **Creating an Access Control Rule**

To create a new rule, click **Add an Access Control Rule**. Enter a unique **Rule Name**, and the **Network** that the rule applies to (the IP address and Netmask). For each service that you want to allow this Network to access, check that service's box.

#### **Web Admin Portal**

The Switchvox Admin Tool Suite.

#### Web User Portal

The Switchvox User Tool Suite (for extension owners).

### **Admin API**

API calls to Admin tools.

### **User API**

API calls to User tools.

#### **Printing**

Printing to Switchvox, including printing files to be faxed.

#### **IMAP**

The Switchvox IMAP server for voicemail and faxes.

### XMPP Server (Jabber)

XMPP chat server traffic.

### XMPP Client (Jabber)

XMPP chat client traffic.

### SIP

SIP traffic.

#### IAX

IAX traffic.

#### NTP

NTP traffic (Network Time Protocol).

### SNMP

SNMP traffic. If you want to monitor Switchvox using SNMP, make sure this service is checked. For information about the OIDs that Switchvox publishes, see SNMP Settings.

### Modifying an Access Control Rule

To modify a rule, click the **Pencil** icon next to the rule. The rule becomes active, and you can change the name, network, and services. Click the **Checkmark** icon to save your changes.

### **Deleting an Access Control Rule**

To delete a rule, click the **X** icon next to the rule. Be sure that you do indeed want to delete this rule, then click **Yes, Delete Rule**. The rule is permanently deleted.

### **Managing Physical Access to the Server**

If you have an appliance with an LCD Panel, you can disable other access to the appliance. You can also clear the password on the LCD Panel, in case you have forgotten it.

### Disable server configuration from a local keyboard

Check this box if you do not want anyone to be able to connect a keyboard and monitor to the Switchvox server and make changes to the server configuration.

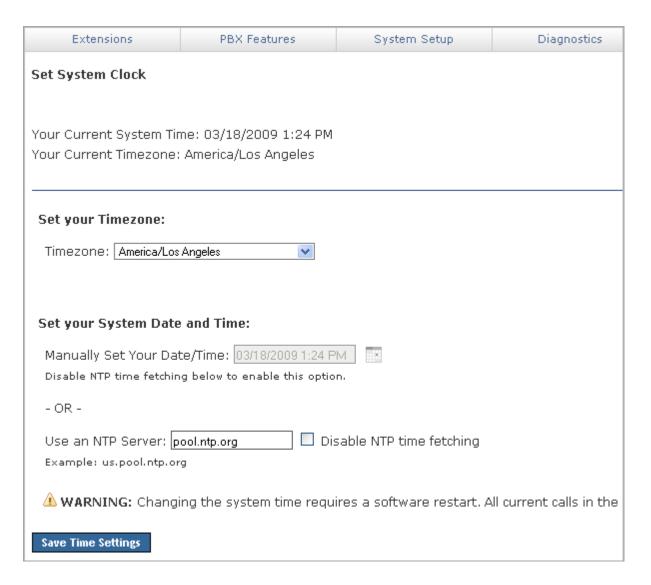
This means that the LCD Panel is the only way to access Switchvox's Basic Server Functions. You can set a password to protect the use of the LCD Panel.

### **Clear LCD Password**

Click this button to clear the password for the LCD Panel. This means that anyone with access to the Switchvox server can use the LCD Panel to configure Switchvox's Basic Server Functions, without entering a password.

# **Set System Clock**

This shows you the current time zone and time setting. By default, Switchvox uses NTP to retrieve its time settings. Unless you have a particular reason to change this, leave your clock set to the default NTP settings. The time zone can be changed if the machine is moved from one location to another.



# **Backups**

Switchvox lets you create and download a backup file that can contain all of the data and audio files for Switchvox. This file may be restored back into Switchvox at any time to return Switchvox to the same state it was in when the backup file was created.

Always make a backup before making changes to Switchvox and before restoring from backup.

**IMPORTANT:** The 'best practice' for backups is to schedule a daily automatic backup that stores the backup file on an FTP server.

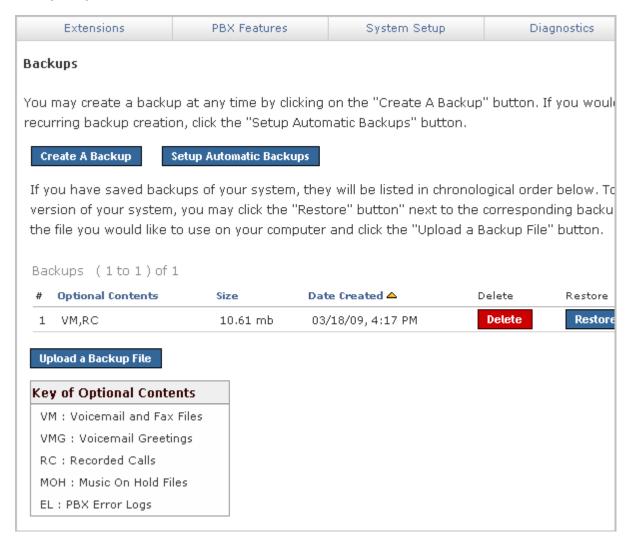
# **Creating a Backup File**

To create a backup file, click Create A Backup. This displays the Backups page.

Click the check boxes next to any of the optional components that you want to include, then click **Create Backup**.

**WARNING:** The backup process may take several minutes depending on the size of the backup file. Do not click any links or buttons during the backup processing.

After the backup is created, the Backup page is redisplayed with the new backup file in the Backups table. Click on **Download** to save the file. Be sure to put it in a location from which it can be easily accessed to restore your system.



# **Creating Automatic Backups**

It is good practice to schedule daily automatic backups. To do this, click **Setup Automatic Backups** to display the Backup schedule page.

When you have entered the settings that you want, click **Save Settings** to save this automatic backup and go back to the main Backups page.

To disable automatic backups, clear all of the FTP information, then click **Save Settings**. This redisplays the main Backup page.

### **Day and Time**

Set the day of the week and time to run the backup process.

### **Number of Copies**

When this number is exceeded, the oldest backup is deleted from the FTP host.

#### **FTP Information**

Set the hostname of your FTP server, the username and password that you use to access the server, and the path to the directory where you want to store the backup files. Click **Test FTP Settings** to test this information and be sure that your backup files can be properly transferred.

### **Optional Components**

Click each 'Optional' box for the files and logs that you want to include in the backup.

### Restoring a Backup

If the backup file you want to restore is listed in the table of Backups, you can restore from the list. If not, you need to upload the backup file and then restore.

**WARNING**: When you restore a backup file the following things happen:

- Your current Switchvox configuration is overwritten with the contents of this backup file.
- All current calls will be dropped.
- All agents will be logged out of queues.
- All data created since the backup was created will be lost, including extensions and voicemail.

So it's important to be absolutely sure that you want to restore your backup file.

### **Restoring From the List**

In the list of backups, find the backup file that you want to restore and click its **Restore** button.

Be sure that you do indeed want to restore this backup, then click Yes, Restore Backup.

### Restoring From an Uploaded Backup File

If the backup file that you want to restore from is not in the list, then you need to upload it. To do this, click **Upload a Backup File**. That page lets you browse your machine to find the correct backup file. When you have selected the file, click **Upload and Restore Backup**.

Be sure that you do indeed want to restore this backup file, then click Yes, Restore Backup.

# **Deleting a Backup File**

To delete a backup file from the list, click on its **Delete** button. Be sure that you do indeed want to permanently delete the backup file, then click **Yes, Delete Backup**. That backup file is permanently deleted from the system is removed from the list. The Backup page is displayed with a message indicating that the file has been deleted.

# **Updates**

This shows you the following information about Switchvox:

- Current Switchvox software version number
- Maximum number of extensions and concurrent calls
- Number of Phone Feature Packs (all, and unused)
- Any Switchvox software updates that are available for you to apply

Extensions	PBX Features	Sy	stem Setup	Diagnostics
Updates				
?What is this for?				
Current Version:	13810	Release Not	es	
Maximum User Extension	s: 9999	(currently usi	ng 20)	
Maximum Concurrent Cal	ls: 99			
Phone Configuration Toke	ens: 150	(currently usi	ng 0)	
Subscription Expiration:	Nov 07 2020			
📤 <b>WARNING:</b> Updati	ng your system will r	esult in an a	utomatic softwar	e restart of your pho
phone connections will	be dropped and any	agents logg	ed in will be logg	ed out of their respe
Update D	escription			Apply
Upgrade to 13872 U	pgrade the system to r	elease 13872	more info	Apply This Update
Obsolete Updates				
Update	Description			Apply

## **Phone Feature Packs**

Phone Feature Packs let you easily configure your phones for use with Switchvox, and offer great tools for customizing your phones.

You need one Phone Feature Pack per SIP phone. Please see your reseller for instructions on purchasing and activating more Phone Feature Packs. If you purchased your phones with Switchvox, then each phone already has its own Phone Feature Pack, and is already configured for use with Switchvox. Phone Feature Packs are not necessary for any other type of extension.

A Switchvox Phone Feature Pack offers the following features:

- Customized phone displays
   Customize some of the information displayed on your phone. See Phone Options.
- Status Indicators

Set a line status indicator light on your phone. This can show whether a call is waiting in a parking lot, or whether you are logged into a queue. See Status Indicator Entries.

- Distinctive Ringtones
   Set the audio file of your choice to ring on incoming calls, depending on the caller. See Distinctive Ring Rules.
- Auto-answer
   Set an automatic answer in Intercom mode, depending on your caller. See Distinctive Ring Rules.

Combine Ringtones and Auto-answer

Combine a ringtone and auto-answer, depending on the caller. See Distinctive Ring Rules.

Phone Applications

Access many of the Switchvox features right from your phone.

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The Phone Features available on your phone will vary, depending on the type of phone you have.

- The **General Options** are available on snom and Polycom phones.
- The other features are only available on Polycom phones. If you have a newer Polycom phone with three or more lines (e.g., the VVX 1500 or SoundPoint 450, 550, 560,650, or 670) you should be able to take full advantage of your Phone Feature Pack. Other Polycom phones may not offer all of the features.

## **Subscription Expiration**

If your Switchvox subscription is about to expire, then you will see alerts and options to renew your subscription. If you renew your subscription those alerts will go away. Or, you can turn the alerts off at any time.

## Switchvox Software Updates

If the **Updates** page says that no updates are available for your system, then you already have the newest version of the software.

If there is an update listed on this page, it is listed with a **more info...** link and a button that lets you apply the update. You can click on **more info** to see the release notes for the update (the release notes tell you what changes are included in this new update). When you are ready to update your system, click **Apply This Update**.

Be sure to make a backup before you apply the update (see Backups), and test Switchvox's basic functionality after the update is complete.

# **Product Registration**

This is used to register Switchvox. Usually registration is completed when the system is first installed. If this is the case, this page displays the first few characters of your Registration Code, which you will need if you call Switchvox Technical Support.

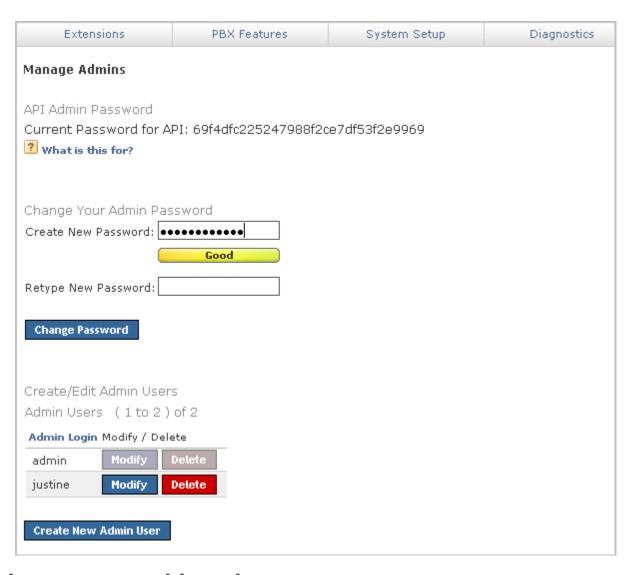
If you have not yet registered Switchvox, please do so.

# **Digium Addon Products**

This lets you register and enable Digium addon products. Select a product and enter your License Key, then click **Submit**. If you install a Fax license, you will be prompted to download and install the related fax software.

# **Manage Admins**

This lets you change the main admin password (User Name 'admin'), and manage additional admin accounts.



# Language and Locale

The Switchvox web-based tool suite is available in several languages:

- English for United States
- English for United Kingdom
- Spanish for Spain
- Spanish for Mexico
- Italian

To set your preferred language:

- 1. Select a language from the dropdown.
- 2. Select a locale for that language.

If you choose to use a language other than English for US, there are some important issues to be aware of:

Dates

All other languages and locales handle dates with the format of day, then month, then year (as opposed to mm/dd/yyyy).

Sound Packs

There are Sound Packs available for each Language preference. Once you select a language, Switchvox serves you the sounds from that language's soundpack. For example, if you switch to the Italian language, and then you call the voicemail access extension, the prompts will be in Italian. Soundpacks must be installed by the Switchvox administrator (see the Admin Suite Digium Addon Products).

Creating Extensions

By default, new extensions are created as English for US. If you want new extensions to default to your preferred language, use Extension Templates.

### The Main Admin Password

To change the password for the main admin account, enter a new password into the **Create New Password** field, and then retype it in the **Retype New Password** field (for verification).

The password is case-sensitive, and it can contain special characters. The strength of your new password is indicated, to help you create a good password. When you are happy with your new password, click **Change Password**.

Each time you change the main admin password, the Call API password is changed and displayed here. If you change your password, be sure to put this new API password in place of the old password wherever you use the Call API with admin authorization.

## **Creating & Editing Admin Users**

You can create multiple 'sub-admin' users, each with their own permissions. This way, you can assign Switchvox administrative tasks to various people, and limit their access to just the tasks they need to perform.

Click **Create New Admin User** to display the **New Admin** page. Enter a new Login name for the user, then enter and re-enter a new password. The password is case-sensitive, and it can contain special characters. The strength of your new password is indicated, to help you create a good password.

Leave the box checked for **Force password change at next login**, unless you really want your new sub-admin user to use the password that you created. If the sub-admin creates a weak password, Switchvox notifies you with an alert.

You can set this sub-admin user's access for each of the main sections in Switchvox. You can block the sub-admin user from using a section at all, or let him or her see the information in that section, or make changes (which allows full access for that section).

Click **Create Admin**, and the **Manage Admins** page is redisplayed with the new sub-admin user included in the Admin Users list.

To modify a sub-admin user, click **Modify** for that user. You can change the password (leave the password fields blank if you do not want to change the password), and all of the access settings. If you want the sub-admin user to change his or her password, make sure the box is checked for **Force password change at next login**. Click **Update Admin** to save your changes and go back to the **Manage Admins** page.

Extensions	PBX Features	System 9	Setup	Diagnost		
Manage Admins						
Login Information for Admin Login: justine						
Password	Too Short					
	100 SHOPE					
Retype Password						
Leave blank if you do not	wish to change the passwo	rd for this admii	7.			
ŕ						
Force password change	ge at next login.					
Access Level						
Section Name		Access Lev	el			
Extensions	Onone	O read only	Oread/write			
Manage Extensions		• read only	O read/write			
Extension Templates	Onone	• read only	O read/write			
Extension Settings	Onone	• read only	O read/write			
Extension Groups	Onone	• read only	O read/write			
Extension Permissions	o none	• read only	O read/write			
PBX Features	Onone	O read only	O read/write			
Music On Hold		oread only	O read/write			
IVR Editor		• read only	O read/write			
Sound Manager		oread only	O read/write			
Time Frames		⊙ read only	O read/write			
Call Recording	Onone	oread only	O read/write			
Switchboard Plugins	Onone	oread only	O read/write			
S	^	^	<u> </u>			
System Setup	Onone	,	O read/write			
Hardware Setup		• read only	O read/write			
Channel Admin	◯ none	<ul><li>read only</li></ul>	oread/write			

#### **API Admin Password**

The Call Creation API has been deprecated. Please use the Extend API (for details, go to the web site developers.digium.com).

The Call Creation API offered a way for external applications to originate phone calls via the PBX. The API required authentication, and used a modified version of the admin password.

# **System Reload**

We recommend that you do not use these options unless instructed to do so by a Technical Support representative.

If you decide to use these tools to reload, restart, reboot, or shut down Switchvox, be prepared:

- Make a backup file (see Creating a Backup File).
- $\circ~$  Inform your Switchvox users that calls will be dropped and members will be logged out of their queues.

# The User Tool Suite

# What is New in version 4.5?

Version 4.5 offers great new features for your phone, additional profile information with a picture, and a language setting.

### Phone Feature Packs: Ringtones, Tools, and Display Options

The following features come with a Phone Feature Pack:

- Upload Distinctive Ringtones to Switchvox
- Use Ringtones and other options to identify different types of incoming calls
- Define several items on your phone's display
- o Access Switchvox tools right from your phone

Directory, Voicemail Mailbox, Parking Lots, Phonebooks, and Record Call.

For details, see Phone Features.

### **Profiles**



An Extension Profile includes some or all of the following:

- Image
- First and Last Name
- o Title
- Location
- Extension Number

Profiles can be shown in the Switchboard and on a phone that uses a Phone Feature Pack. For details, see Your Phone Feature Pack.

### Language Setting

You can select a language preference for the User Suite, which will also set your Sound preference. For details, see Language Setting.



# **Getting Started**

Welcome to Switchvox, the world's most powerful, yet easy to configure IP-PBX. This document describes how to use the Web-based Switchvox User Tool Suite.

Throughout this document, we refer to the User Suite menu options as a path, using '>'. Instead of saying "select the Modify Account option from the Settings menu in the navigation bar, we say "select **Settings > Modify Account**."

**IMPORTANT:** You may find some features described in this document that are not available to you. This is determined by the version of Switchvox you are using, and what features your Switchvox administrator has given you permission to use. Please contact your administrator if you have any questions.

### **About the User Suite**

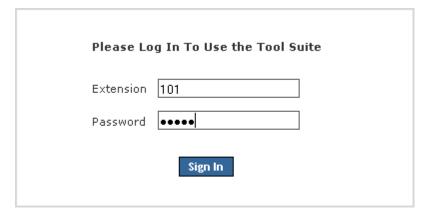
Each user has a Switchvox account that is his or her phone extension, and a password for that account. Ask your Switchvox administrator for your extension number and password, and for the URL of your Switchvox User Tool Suite (the User Suite).

# **Logging Into the User Suite**

On a computer that is on the same network as Switchvox, open a Web browser and enter the URL of the User Suite:

https://YourServerIPaddress/main

The Login page is displayed.



Enter your extension and password (provided by your administrator). Once you are logged in, you may have to change your password. If you do, the strength of your new password is indicated, to help you create a good password. The next time you go to the User Suite, log in using your new password.

To log out of the User Suite, click **Logout** (located in the upper right-hand corner).

## **User Suite Navigation**

The User Suite navigation bar contains the following dropdown menus:

Settings

Set up your account, specify how you want to handle calls, manage your Phonebooks, and set up your own conference room.

### Voicemail / Fax

Manage incoming and outgoing voicemail and faxes, set up voicemail greetings, and customize email notifications for voicemail.

### Call History

See logs of all calls to and from your extension and run usage reports.

# **Getting Help**

Click any help link to open a pop-up window with information about a particular tool. Help links are available as question-mark icons and as text links.



If a pop-up window does not display, set your Web browser to allow pop-ups from the domain of your Switchvox server.

# **Settings**

The Settings area is where you set up your account, specify how you want to handle calls, manage your Phonebooks, and set up your own conference room.

# **Modify Account**

This section lists basic information about your Switchvox account.

Your administrator may have given you permission to change this information. If you do make changes, click **Modify Extension** to save those changes. The page is refreshed, and a message indicates that your changes were successfully saved.

If you are changing your password, the strength of your new password is indicated as you type, to help you create a good password. The next time you go to the User Suite, log in using your new password.

### **Your Profile**



An Extension Profile includes some or all of the following:

- Image
- First and Last Name
- o Title
- Location
- Extension Number

Profiles can be shown in the Switchboard and on a phone that uses a Phone Feature Pack. For details, see Your Phone Feature Pack.

### **Profile Picture**

To add a picture:

- 1. Click Add an image.
- 2. Click **Browse**, and then find the image file on your computer. Images must be at least 75x100 pixels, and must be in jpg format.
- 3. Click **Upload**.

  The image is uploaded and displayed with the crop area that will be used to make a 75x100 image. You

can move the crop area around to choose the best portion of the picture. The Preview on the right shows you what the picture will look like once it is cropped.

- 4. Click Save Image.
- 5. Click Save Extension Settings.

To upload a different picture, re-crop the existing picture, or delete the existing picture:

- 1. Click View Image.
- 2. Click ReCrop, or Delete.

If you want to upload a different image, **Delete** this image now, and then click **Add an image** to start again.

- 3. Confirm your changes.
- 4. Click Save Extension Settings.

# **Language Setting**

The Switchvox web-based tool suite is available in several languages:

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- English for United Kingdom
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- Spanish for Mexico
- Italian

To set your preferred language:

- 1. Select a language from the dropdown.
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If you choose to use a language other than English for US, there are some important issues to be aware of:

Dates

All other languages and locales handle dates with the format of day, then month, then year (as opposed to mm/dd/yyyy).

Sound Packs

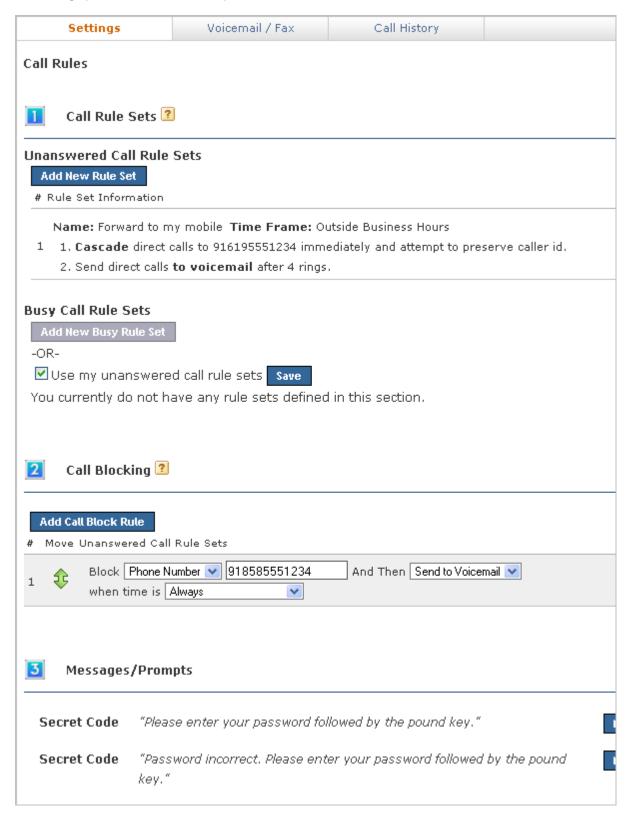
There are Sound Packs available for each Language preference. Once you select a language, Switchvox serves you the sounds from that language's soundpack. For example, if you switch to the Italian language, and then you call the voicemail access extension, the prompts will be in Italian. Soundpacks must be installed by the Switchvox administrator (see the Admin Suite Digium Addon Products).

# **Call Rules**

Call Rules control how your calls are handled. Here are some examples:

- o When I'm on the phone, go straight to voicemail
- o If I don't answer, 'cascade' calls to my mobile phone then my home phone

- Forward all my calls to another extension
- o On weekends and evenings, forward all my calls to another extension
- o Hang up on calls from a certain phone number



### **Call Rule Sets**

**Unanswered Call Rule Sets** specify how a call is handled after a certain number of rings. **Busy Call Rule Sets** specify how a call is handled when your line is busy. You can set up different rules for those situations, or use the same rules.

When a call is unanswered or your line is busy, Switchvox evaluates each of your Call Rules in the order listed. The first set is evaluated in order, then the second set, and so on. That means that the 'During Business Hours' rule that forwards a call to your mobile phone must come before the 'Anytime' rule that sends a call straight to your voicemail.

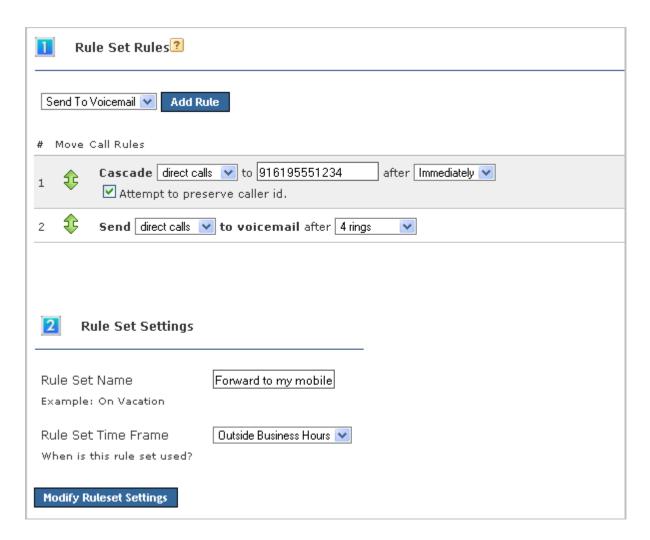
It's a good idea to test your rules by calling your extension from another phone.

### Creating a New Call Rule Set

To create a new Call Rule Set under **Unanswered** or **Busy**, click the appropriate **New Rule Set** button. To create a new Busy Call Rule Set, you must first un-check the box **Use my unanswered call rule sets**. That makes the **Add New Busy Rule Set** button active (it turns blue), so you can click it and create a new rule set.

When the settings are correct, click on Save Ruleset Settings to create the new rule set and add its rules.

- 1. Select the type of rule to add from the dropdown
- 2. Click Add Rule.
- 3. Specify whether the rule applies to direct calls, queue calls, or all calls.
- 4. Depending on the type of rule you are adding, enter the appropriate extensions or phone numbers.
- 5. If you are adding a Secret Code rule, enter 1 to 10 digits.
- 6. If you are creating an Unanswered rule, specify the number of times to ring your phone before following this rule.



### **Rule Set Name**

Enter a unique name for this rule set.

### **Rule Set Time Frame**

Select a time frame from the dropdown. Switchvox evaluates the current date and time against this time frame, and if they match then the rule is followed. If they do not match, Switchvox moves on to the next rule. See Time Frames for more information.

### Send to Voicemail

This rule sends a call to your voicemail so the caller can leave a message.

#### **Call Forward**

This rule forwards a call to another Switchvox extension. The extension can be another phone, a call queue, or any other type of extension. Once a call is forwarded to the extension, the call rules for that extension are followed.

### Call Cascade

This rule rings another extension or an external phone number (i.e., your mobile phone), and if the call isn't answered then it is passed on to the next rule in the set. This action depends on your Outgoing Call Provider, but if your provider supports it, you can stack several rules that try and reach you at several phone numbers. **Note:** If you are cascading a call to an external phone number, be sure to add a '9' or other necessary digits that you normally use to dial out.

#### **Secret Code**

This rule prompts the caller to enter a secret code, then passes the caller to the next rule in the set. If the caller cannot enter the correct secret code after the specified number of tries, then you can specify what to do with the call.

### Ring All

This rule simultaneously rings up to 3 extensions or external phone numbers. *Note:* If you are ringing an external phone number, be sure to add a '9' or other necessary digits that you normally use to dial out.

### **Managing Call Rule Sets**

If you have multiple Rule Sets, you can drag and drop them to prioritize them appropriately, or you can delete them and replace them with new rules. Your rules need to be put in order because they are executed from top to bottom.

You can have different rule sets for different times, and activate or deactivate them as you see fit. In order to activate a rule set, click on the Activate button to the right of the rule, and fill in the time you want the rule to occur. The active rule set will be highlighted yellow and will display how long it will be active on the right side. This process is the same for Unanswered or Busy Call Rule Sets.

## **Call Blocking**

Switchvox lets you specify phone numbers that you do not want to accept calls from. You can block specific phone numbers, or any numbers that begin with the same prefix, and you can block them at all times or during specified time frames. You can also choose how Switchvox handles the blocked calls.

Click **Add Call Block Rule** and a new blank rule section appears. Select *Phone Number* or *Prefix* from the dropdown, then enter the number. Be careful entering a prefix; if you put in the wrong prefix you might end up blocking more calls than you expected. Select the action Switchvox should take with the blocked call, then select a Time Frame if you want to control when these calls are blocked. Click **Save** to save your changes to the Call Block Rule.

## Messages/Prompts

If you use the *Secret Code* feature in a Call Rule, the default sound prompts are provided here. You can change the prompts by recording your own sound over your phone, or uploading a new sound file.

To add or change your own sound, click the prompt's **New** button. To play the sound, click the prompt's **Play** button. To remove a sound that you created, and go back to the default sound, click the prompt's **Use Default** button.

# **Time Frames**

Switchvox can operate differently based on the date, day, and time.

A time frame is considered valid when the current date and time match any one of a Time Frame's rules. Not all of the rules have to match.

## **Creating a New Time Frame**

To create a new time frame, enter a unique name and click **Create New**. Enter any appropriate Start and End Dates, Start and End Days, and Start and End times, then click **Add**.

Continue adding rules until you have completed the Time Frame definition.

The following are sample time frame definitions:

Start Date	End Date	Start Day	End Day	Start Time	End Time
-	-	Monday	Monday	-	-
-	-	Wednesday	Wednesday	-	-

Mondays 12:01AM to 12:00PM, and Wednesdays 12:01AM to 12:00PM

Start Date	End Date	Start Day	End Day	Start Time	End Time
-	-	-	-	3:00 AM	6:00 AM

### **Early Morning**

Start Date	End Date	Start Day	End Day	Start Time	End Time
07/01/2009	07/07/2009	-	-	-	-

### The First Week in July 2009

Start Date	End Date	Start Day	End Day	Start Time	End Time
01/21/2009	-	-	-	-	-

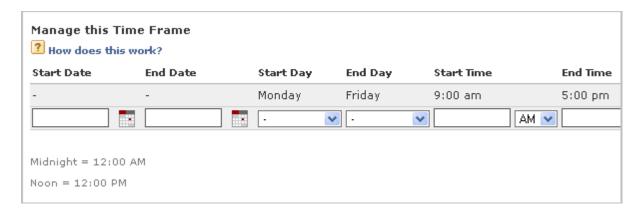
After January 21, 2009

Start Date	End Date	Start Day	End Day	Start Time	End Time
02/01/2009	02/28/2009	Monday	Friday	6:00 AM	7:00 AM

Weekdays in February, 2009 between 6:00 and 7:00 in the morning

# **Editing a Time Frame**

To change a time frame, click its **Edit** button. This displays the Manage This Time Frame page. Create a new rule, or multiple rules, and delete any rules that you do not want to include in this time frame. (You cannot edit a rule.)



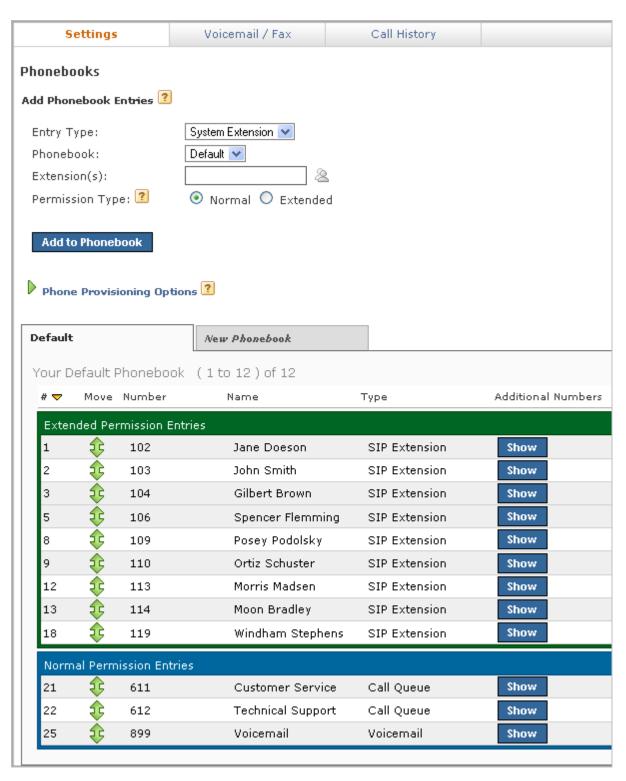
# **Deleting a Time Frame**

To delete a time frame, click its **Delete** button. Be sure that you do indeed want to delete this time frame, then click **Yes, Delete Time Frame**. The Time Frames main page is deleted with a message that the time frame was deleted.

Before you delete a Time Frame, be sure you do not have any IVR Menus or Call Rules that depend on it.

# **Phonebooks**

Your Switchvox Phonebooks let you organize your contacts so that you can easily access them from your Switchboard. Phonebooks can include your fellow Switchvox users, and external contacts.



# **Adding Phonebook Entries**

There are different types of Phonebook entries, depending on whether the entry is known within Switchvox, or is an external phone number.

### **System Extension Entries**

System Extension Entries are other Switchvox extensions. You can add user-type extensions, and other types of extensions such as queues or IVRs.

#### **Phonebook**

Select the Phonebook this entry belongs in. To create a new Phonebook, use the **New Phonebook** tab.

### Extension(s)

Enter at least one extension here. To enter multiple extensions, put a comma (,) after each extension. Click the people icon to see a directory of your Switchvox extensions. Hold down *Ctrl* to select many extensions, or hold down *Shift* to select a block of extensions.

### **Permission Type**

*Normal:* Shows the user's Presence, and whether or not he or she is on the phone. Also includes a dropdown that lets you open a chat, call additional numbers, or Intercom the person.

*Extended:* Shows the Normal features, plus the caller ID name and number of the user's active calls. It also includes options to monitor, record, whisper, barge into, or pick up the user's calls.

Only phone-type extensions can be Extended Permission entries. If you try to add any other type of extension (e.g., a Call Queue), it is added as a Normal entry.

You can change an entry's permission type at any time.

If you cannot see or use an option in an Extended Entry, your Switchvox administrator has not given you permission to do so.

### **External Number Entries**

External Number Entries are for names and phone numbers other than Switchvox extensions. Once you add an external entry, you will also be able to put Additional Numbers into that entry.

Make sure to enter external phone numbers as you would dial them from your phone. For example, if you must dial 9 to get an external line, this entry must begin with a 9.

### **Phonebook**

Select the Phonebook this entry belongs in. To create a new Phonebook, use the **New Phonebook** tab.

### Number

Enter the phone number.

#### Name

Enter a name.

#### Jabber ID

Enter this person's Jabber ID to display his Jabber presence on your Switchboard in his Phonebook entry.

The Jabber ID has a user ID (the extension) and a Jabber Hostname. For example, **101@jabber.peeredswitchvox.com**.

### Status Indicator Entries

These entries are only available for Polycom and snom phones that Switchvox has configured in the **Phone Setup** tool. If you don't see this type of entry, check with your Switchvox administrator.

A Status Indicator entry controls a line status indicator light on your phone.

#### **Phonebook**

Select the Phonebook this entry belongs in. To create a new Phonebook, use the **New Phonebook** tab.

**IMPORTANT:** This must be the same Phonebook as the one in the Phone Features Options for this entry to affect your phone. Also, it must be one of the first entries in the Phonebook if you want it to display without having to scroll through the entries on your phone.

#### **Entry Name**

Enter a name for this entry. This isn't required, as you don't see the entry in your Switchboard.

### **Monitor Type**

*Parking Lot:* This lights up the line status indicator on your phone when a caller is waiting in the specified **Parking Lot Extension**.

Queue Login Status: This lights up the line status indicator on your phone when you are logged into the specified queue. Enter the **Agent Login Extension** that you use to log into this queue, and the queue that you want to monitor.

### **Switchvox-Peer Entries**

If your Switchvox is peered with another Switchvox, you can include the peered Switchvox extensions in your Phonebook.

For peered extensions, use the **External Number** entry type (because this extension is not on your Switchvox).

#### **Phonebook**

Select the Phonebook this entry belongs in. To create a new Phonebook, use the New Phonebook tab.

### Number

Enter the extension number from the peered Switchvox, just as you would dial it on your phone. Peered extensions might start with a different number than yours, or have more digits.

#### Name

Enter a name for this entry.

#### Jabber ID

Enter this person's Jabber ID to display his Jabber presence on your Switchboard in his Phonebook entry.

The Jabber ID has a user ID (the extension) and a Jabber Hostname. For example, **101@jabber.peeredswitchvox.com**.

# **Managing Phonebooks**

To create a new Phonebook, click the **New Phonebook** tab. Enter a unique name for this Phonebook, and click **Create New Phonebook**. This Phonebook is created and is now available in the Phonebook dropdown when you create a new entry.

To move an entry back and forth as a Normal or Extended entry, drag the entry's green arrow icon.

To delete an entry, click its **Delete** button. Be sure that you do indeed want to delete this entry, then click **Yes, Delete**. The Phonebook is refreshed, and this entry is gone.

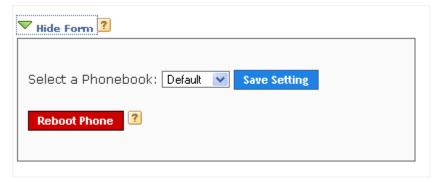
To delete a Phonebook, click its **X** icon. Be sure that you do indeed want to delete the Phonebook, then click **Yes, Delete**.

### **Phone Features Options**

The **Phone Features Options** let you specify a Phonebook to push out to your phone. Your phone must have a Phone Feature Pack, otherwise these settings are not available. For details, see Your Phone Feature Pack.

When you make changes to a Phonebook, or you want to change the Phonebook that is already on your phone, click **Reboot Phone** to send the correct Phonebook out to your phone. Rebooting your phone takes a few minutes, and you will not be able to make or receive calls while it is rebooting.

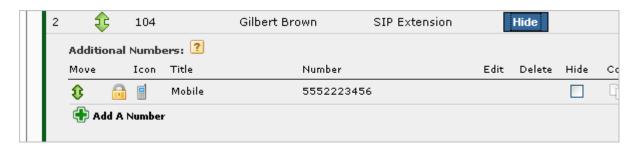
**Note:** If the phone is not on the same network as Switchvox, the phone might not be rebooted. In that case, you must reboot the phone manually.



### **Additional Numbers**

Each Phonebook entry is based on one main extension or phone number, but it can have also have additional numbers associated with it. This way, you can have one Phonebook entry for a person, and have all of their contact numbers included in that entry. The Additional Numbers are available via a dropdown (the blue plus sign) in the Switchboard Phonebook entry.

To see and modify Additional Numbers for an entry, click the entry's **Show** button.



Additional Numbers can be entered by an extension owner, or you can enter them in your Phonebook entry.

Additional Numbers entered by an extension owner are available when you add the extension to your Phonebook. These numbers appear as locked and cannot be edited, but you can choose to hide them in the Switchboard.

Additional numbers that you add to your Phonebook entry can be edited and deleted. You can also copy any Additional Numbers (including the owner's numbers), and then edit or delete the copy.

### **Additional Info**

Your account and profile include Additional information that is used to help others in your organization find you when you are away from your desk. For example, you can add your mobile phone number, your home phone number, or even another extension that you use frequently. Numbers added here are shown in your coworkers' Phonebook entries.



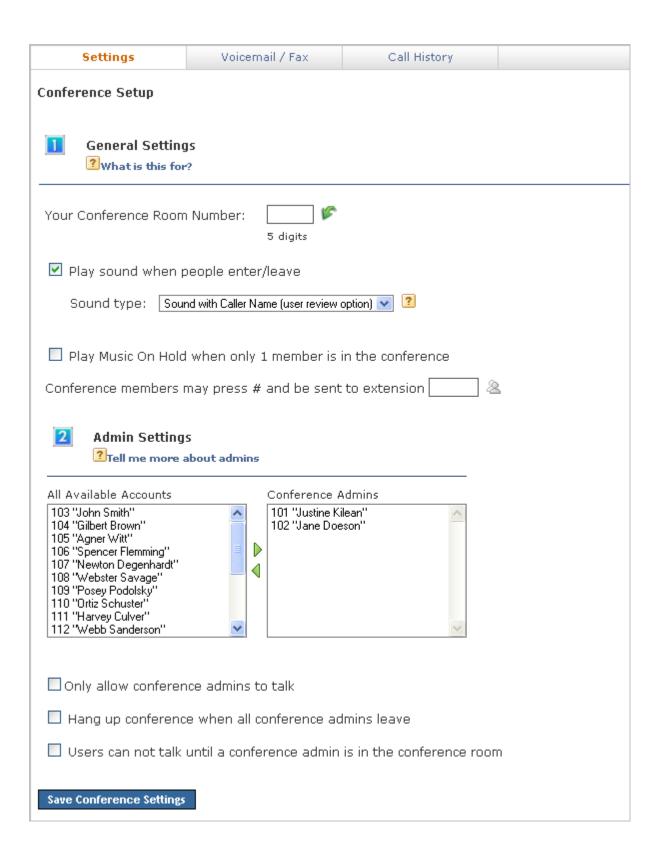
To create a new number, click **Add A Number**. You can set an icon, title, and a phone number or extension. The icon and title are displayed in your co-workers' Phonebook entries. To change the icon for an entry, click the icon.

**IMPORTANT:** Make sure to enter the number as you would dial it from your phone. For example, begin with a 9 if that's how you would normally dial that number.

Click the pencil icon to change a number, or the red X to delete a number. Drag the green arrows to change the order that your numbers are shown in.

# **Conference Setup**

This section lets you create your own conference room. Ask your Switchvox Administrator for the main conference room extension number. When a caller dials that number, he or she can access your conference room by dialing your conference room number.



## **General Settings**

You can change the behavior of your conference room using several options.

#### **Your Conference Room Number**

Enter a unique 5-digit number for your conference room. This is the number that you give to callers so they can access your conference room. Click the green arrow to have the system pick a random, unused room number for you.

### Play sound when people enter/leave

Check this box and select a sound to play to all members of the conference room when a new caller enters the room. There are three options for the sound type that is played when a caller enters and exits your conference room:

Only Sound: Only a sound is played, no information about the caller.

Sound with Caller Name: The caller is prompted to record his or her name. The recorded name is then played along with a sound.

Sound with Caller Name (user review option): This is the same as Sound with Caller Name, but it gives the caller an opportunity to listen to the recorded name, then accept or re-record it before joining the conference room.

### Play Music On Hold when only 1 member is in the conference room

Check the box if you want this to be true.

### Conference members may press # and be sent to extension []

Enter an extension here, so that when member hit the # key, they are transferred to that extension.

# **Admin Settings**

Admins are special members of a conference room who have additional privileges. Use the lists in this section to identify which extensions are treated as admins of your conference room, and set the options for your admins.

### Only allow conference admins to talk

Check this box to make all non-admin callers listen-only.

### Hang up conference when all conference admins leave

Check this box to ensure that your conference room is closed down after the conference is over.

#### Users cannot talk until a conference admin is in the conference room

Check this box to keep callers mute until an admin is in the room. Callers can join the room, but they cannot speak to each other.

### **Phone Features**

Use the Phone Features page to customize the way your phone works. Your phone must have a Phone Feature Pack, otherwise these settings are not available. For details, see Your Phone Feature Pack.



**IMPORTANT**: If you make changes to any of these settings, you must reboot your phone for the changes to take affect.

## **Phone Options**

Phone Options let you set up the behavior of your phone. Your phone must have a Phone Feature Pack, otherwise these settings are not available. For details, see Your Phone Feature Pack. Phone Setup

### **General Options**

General options can be set for snom phones and Polycom phones.

#### Line Label

This is the set of letters and/or numbers that are shown on the line for your extension. By default, this is your extension number.

### Auto-answer Switchboard initiated calls

This sets your phone to automatically answer Switchboard-initiated calls. If you click on a Switchboard phonebook entry, normally your phone rings and you have to pick it up, then Switchvox dials the number for that entry. If this box is checked, your phone rings and then automatically goes to Intercom mode and answers the Switchboard, and you hear the phone ringing for the call to your phonebook entry.

#### Hide missed calls

This turns off the **missed calls** notification. You can turn this back on but you will have to restart your phone, so any calls you have missed in the meantime will be ignored, and you will not have a missed call notification after the phone is restarted.

### **Polycom Options**

Polycom Options are for Polycom phones only.

#### Line keys per registration

This sets the number of buttoned-lines on your phone that are set to this extension. For example, if you have a 3-line phone, you can set all 3 lines to your extension. That way, you can have an active call and 2 calls on hold, and use the buttons to manage those calls.

Any line on the phone that is not used for your own extension will be automatically filled with your Phonebook entries. For details, see Phone Features Options.

### Do not show caller profile on phone during incoming/outgoing calls

This turns off the Switchvox Profile feature. For details on profiles, see Your Profile.

### **Distinctive Ringtones**

Switchvox can store and play distinctive ringtones that can be used on your Polycom phone. Your phone must have a Phone Feature Pack, otherwise these settings are not available. For details, see Your Phone Feature Pack.

### Add a Ringtone

You can add as many ringtones to Switchvox as you want. However, each phone has two important limitations for supporting ringtones that do not come standard on the phone: Maximum disk-use of 300 kilobytes for all ringtones Maximum of 8 ringtones



## To add a ringtone:

- 1. Click **Browse**, and find the file on your computer.
- 2. In the Name field, enter a name for this ringtone.
- 3. Click Add Ringtone.

The Distinctive Ringtones list is refreshed, with your new ringtone listed.

## **Distinctive Ringtones List**

This list shows the ringtones that you have uploaded to Switchvox, and the ringtones that come with a Polycom phone. (If you have added ringtones directly to your phone, those are not included in this list.) Depending on the permissions you have on each ringtone, you can do the following:

Play the ringtone.

The audio file is downloaded to your computer, then played using your default audio player for .WAV files.

**Edit** the ringtone.

You can upload a new file, or change the Name.

X Delete the ringtone.

You must verify that you want the ringtone deleted. If you delete a ringtone, every Ring-Rule that uses that ringtone is also deleted.

# **Distinctive Ring Rules**

These rules control what happens when you get an incoming call on your Polycom phone. Your phone must have a Phone Feature Pack, otherwise these settings are not available. For details, see Your Phone Feature Pack.

Ring Rules define conditions for callers or kinds of calls, and then define what the phone does when those calls come in. Here are the actions the phone can take if a call meets your conditions:

- Ring with the specified ringtone
- o Auto-answer in Intercom mode
- o Ring and auto-answer
- Mute the ring (your line still blinks)

To add a new Ring Rule:

- 1. Enter a Rule Name.
- 2. Select the action your phone should take (Ring Type).



- 3. Select a Ringtone, if you have chosen a Ring action.
- 4. Click Add Ringtone Rule.

The new rule is created, with a link to **Modify Rule Conditions**.

5. Click Modify Rule Conditions.

The Rule Condition window is displayed.

- 6. Select a Call Type and set the Conditions
  - Extension

If the incoming call is from any one of these extensions, the condition is met. This can be one or many extensions.

Caller ID Number Range

If the incoming caller ID number is equal to one of these numbers, or it falls between them, the condition is met.

Caller ID Number Pattern

If the incoming caller ID number matches this number pattern, the condition is met.

Use the asterisk (\*) to indicate that "any number matches." For example, **619555**\* matches every call with area code 619 and prefix 555.

Caller ID Name Pattern

If the incoming caller ID name matches this text pattern, the condition is met.

Use the asterisk (\*) to indicate that "any letter matches." For example, \*Smith matches every call from a name that ends with Smith.

Ring Hint

If the incoming call has this Ring Hint associated with it, the condition is met.

Ring Hints are created and managed by your Switchvox administrator. They can indicate almost anything, so check with your administrator before using this condition.

· Queue Calls

If the incoming call comes from any queue, the condition is met.

· Direct Calls

If the incoming call was made directly to your extension (or to a DID that is directly routed to your extension), the condition is met.

Internal Calls

If the incoming call originated in Switchvox, the condition is met.

External Calls

If the incoming call originated outside of Switchvox, the condition is met.

When you have created the conditions that you want, click Close.

## 7. Restart the phone.

Once you have set up Ring Rules and restarted your phone, your ringtone disk space usage is displayed. This is specific to your phone, and can help you decide which ringtones to use for your Ring Rules. You can add as many ringtones to Switchvox as you want. However, each phone has two important limitations for supporting ringtones that do not come standard on the phone: Maximum disk-use of 300 kilobytes for all ringtones Maximum of 8 ringtones

To change existing Ring Rules or Conditions:



Click the Edit button to change the details of a rule, or of one condition in a rule.



Click the Save button to save the changes that you made.



Click the delete button to delete a Ring Rule, or to delete one condition in a rule.

# Your Phone Feature Pack

If your phone does not have a Phone Feature Pack, please see your Switchvox administrator.

A Switchvox Phone Feature Pack offers the following features:

Customized phone displays

Customize some of the information displayed on your phone. See Phone Options.

Status Indicators

Set a line status indicator light on your phone. This can show whether a call is waiting in a parking lot, or whether you are logged into a queue. See Status Indicator Entries.

Distinctive Ringtones

Set the audio file of your choice to ring on incoming calls, depending on the caller. See Distinctive Ring Rules.

Auto-answer

Set an automatic answer in Intercom mode, depending on your caller. See Distinctive Ring Rules.

Combine Ringtones and Auto-answer

Combine a ringtone and auto-answer, depending on the caller. See Distinctive Ring Rules.

Phone Applications

Access many of the Switchvox features right from your phone.

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The Phone Features available on your phone will vary, depending on the type of phone you have.

- The **General Options** are available on snom and Polycom phones.
- The other features are only available on Polycom phones. If you have a newer Polycom phone with three or more lines (e.g., the VVX 1500 or SoundPoint 450, 550, 560,650, or 670) you should be able to take full advantage of your Phone Feature Pack. Other Polycom phones may not offer all of the features.

# **Phone Applications**

To use the Switchvox Applications on your Polycom phone:

- Press the **Applications** key on your phone.
   If your phone does not have that key, press the **Services** key.
- 2. If necessary, log into **Applications** by entering your password.

You will need to log into the Applications area if you have logged out, or if your phone has been logged into a different extension.

Press the alpha/numeric soft key until it is set to numeric, then enter your password (the same numeric password that you use to log into the Switchvox User Suite and your Voicemail). When your password is correct, press the **Right Arrow** key to highlight **Login**, then press the **Select** key (this is probably a Checkmark on your phone).



- Select the Application that you want to use.
   Press the Arrow keys to move up and down in the list of Applications, and press the Select key for the Application that you want to use.
- You can log out of Applications at any time. Use the **Arrow** keys to navigate to **Log Out**, then press **Select**.

**Note:** Depending on your phone, navigating with the arrow keys might work differently. Also, when you first press the **Applications** key on your phone, the display will show the last page that was available. For example, you might see the profile of your last incoming caller.

The following Switchvox Applications are available:



**Directory Access** 

Access the Switchvox Directory right from your phone.



Parking Lots

View and pick up parked calls.



Phonebooks

Access your Switchvox Phonebooks from your phone.



Record Call

Record your active call. (Call recordings are found in your Mailbox).



Voicemail

View and listen to your Switchvox voicemail.



Log Out

Log out of these Switchvox Applications.

Within each Application, you can use your phone's soft keys.



## Home

Return to the list of applications.

#### **Exit**

Exit Switchvox Applications and return to the default screen for your phone.

### Refresh

Refresh the information on the screen. For example, you might press Refresh in Parking Lots, to see if a call was just parked.

## **Back**

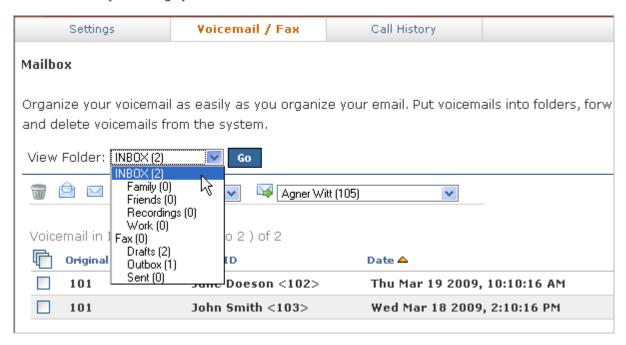
Go back to the previous screen that you used. For example, if you navigated through your Phonebooks to a Phonebook entry, Back would return you to the list of Phonebook entries.

# Voicemail / Fax

This area lets you set up your voicemail and fax options, and manage incoming and outgoing voicemail and faxes.

# **Mailbox**

Your Mailbox lets you manage your voicemail and faxes in Switchvox.



You can access your Mailbox here in the User Suite, or from your desktop email application. See Putting the Mailbox on Your Desktop for more information about using your desktop email application.

## Voicemail

Your voicemail comes into the INBOX folder. Each message displays the original mailbox the message was left in (the message might have been forwarded to you by a co-worker), the caller ID of the caller (if it was available), the date and time the message was left, and the duration of the message.

To play or download a message, click its **Play** button. This marks the message as **Read** (it is no longer shown as bold, and the message-waiting indicator light on your phone would go off if this was your only unread message).

To delete messages, or mark them as *read* or *unread*, check the box for each message that you want to affect. Then, click the appropriate icon (at the top of the list). If you delete messages, you are prompted to verify the action. Be sure that you do indeed want to delete the messages, then click **Yes, Delete**.

To move messages to a different folder, check the box for each message that you want to move, select the folder where the messages belong, then click **Move To Folder**.

To forward messages to another Switchvox user, check the box for each message that you want to forward, select the extension to forward them to, then click **Forward To**.

To sort by column in the message list, click the column name.

**Note:** You can also manage your voicemail from your phone by dialing the voicemail access extension (default is 899). Follow the prompts to give your password and check your messages. Or, your phone may have a **Messages** button that dials the voicemail access extension for you.

## **Faxes**

Your faxes come into the Fax folder. Your outgoing faxes are stored in the folders Drafts, Outbox, and Sent. The Drafts folder includes the files that you have printed to your Switchvox fax printer. The Outbox folder includes faxes that Switchvox is currently trying to send (or was unable to send). The Sent folder includes faxes that Switchvox has successfully sent.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

To send a fax, check its box (you can only send one fax at a time) then click the fax machine icon (at the top of the list). You are prompted to enter a fax number. Enter the number for the fax machine you want to send this fax to, then click **Send Fax**.

IMPORTANT: When you enter the fax number, be sure to prefix any digits that you normally use to dial out.

Your fax is in the *Outbox* folder while it is being sent, and moved to the *Sent* folder when it has been successfully sent. If Switchvox cannot reach the other fax number to send your fax successfully, it remains in the *Outbox* folder.

**Note:** Switchvox sends fax notification email messages. These notification messages show the status of your fax. A *Pending* fax is still being sent (a fax may be Pending for a little while if Switchvox has to try multiple times), a *Sent* fax has been sent successfully, and a *Failed* fax was never received by the fax machine at the number you indicated. For details on setting up fax notifications, see Fax Options.

To preview page 1 of a fax, click **Preview**.

To view or download an entire fax, click **Download**. This marks the fax as **Read** (it is no longer shown as bold).

To delete faxes, or mark them as *read* or *unread*, check the box for each fax that you want to affect. Then, click the appropriate icon (at the top of the list). If you delete faxes, you are prompted to verify the action. Be sure that you do indeed want to delete the faxes, then click **Yes, Delete**.

To move faxes to a different folder, check the box for each fax that you want to move, select the folder where the faxes belong, then click **Move To Folder**.

To forward faxes to another Switchvox user, check the box for each fax that you want to forward, select the extension to forward them to, then click **Forward To**.

To sort by column in the message list, click the column name.

For more information about faxing, see Creating & Sending a Fax.

# Putting the Mailbox on Your Desktop

Your Switchvox Mailbox uses IMAP, a protocol that lets your desktop email application access remote messages as if they were stored on your computer. In this case, it means that you can access your voicemail and faxes in your Switchvox Mailbox using your regular email application (e.g., Microsoft Outlook or Mozilla Thunderbird).

Your Switchvox Mailbox does not offer outgoing mail service, so if you want to forward a message or a fax, you need to use your regular email account.

To set up your Switchvox IMAP Mailbox in your desktop email application, you'll need the following information:

- The IP address or domain name of your Switchvox mail server. This is the same IP or name that you use for the Switchvox User Suite.
- Your 'email address,' which in this case is your extension number, the @ sign, then your IP address or domain name. For example: 101@192.168.0.100, or 101@pbx.mybusiness.com.
- o Your 'user name', which in this case is your extension number.
- Your password, which is the same password that you use to log into the Switchvox User Suite and to collect your voicemail.

**IMPORTANT:** If you are having trouble using your Switchvox IMAP account in your email application, check with your Switchvox administrator. It's possible that you need to use a different port number than the default (143), or that you need to set specific security options.

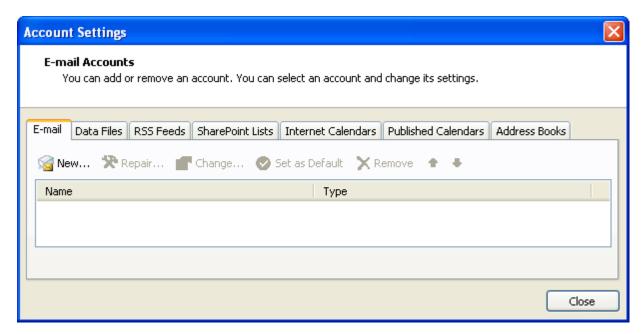
In your email application, start the process to create a new Internet email account. When you are prompted to enter the type of incoming mail server for the account, choose IMAP. Enter the rest of the information as prompted.

## Sample IMAP Setup

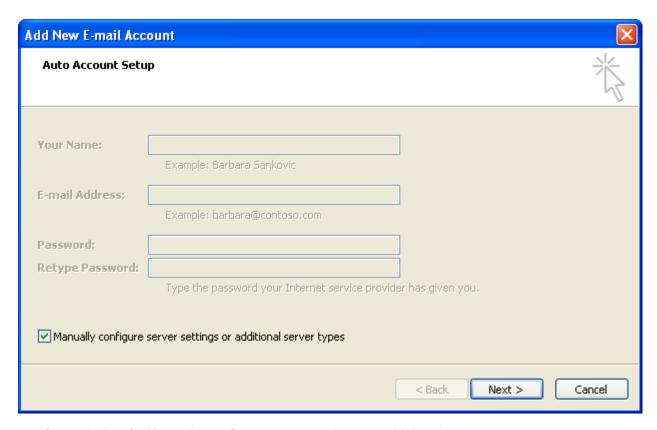
The following gives an example of creating an IMAP account in Microsoft Office Outlook 2007, running on the operating system Windows XP Professional.

1. Open Microsoft Outlook, and select **Tools > Account Settings**.

The *E-mail Accounts* window is displayed.

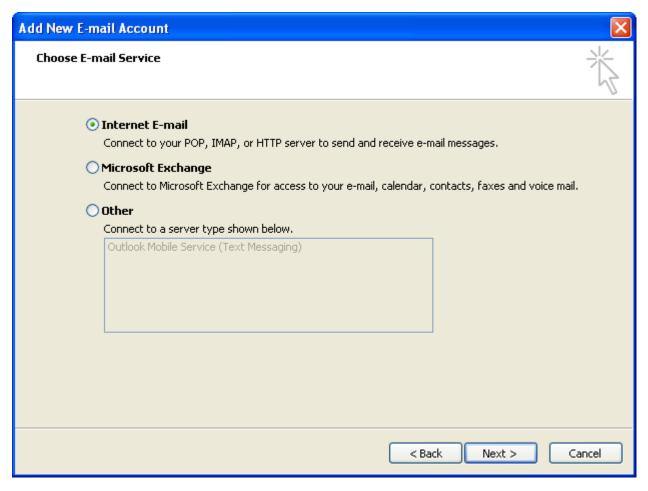


- 2. Click **New** to create a new e-mail account.
- 3. The Auto Account Setup window is displayed.



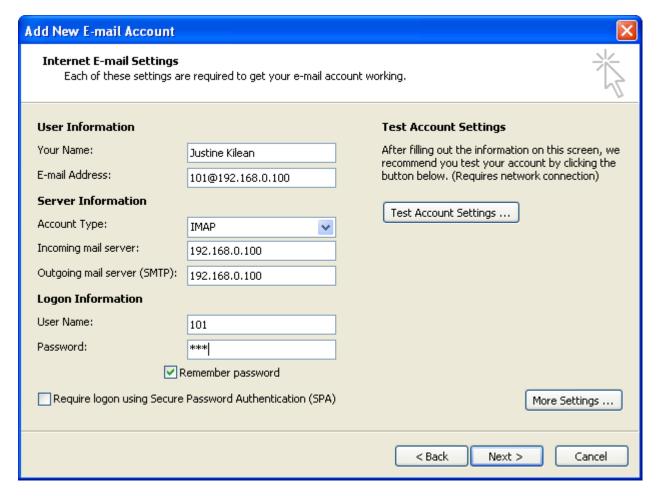
- 4. Check the box for Manually configure server settings or additional server types.
- 5. Click Next.

The Choose E-mail Service window is displayed.



- 6. Select Internet E-mail.
- 7. Click Next.

The Internet E-mail Settings window is displayed.



- 8. Enter your name and your email address in the **User Information** section.
- 9. Select the Account Type IMAP, and enter the IP address or domain name of your Switchvox mail server in the **Server Information** section.
- 10Enter your user name and password in the Logon Information section.

Check the box **Remember password** if you don't want to have to enter your password each time you access your email.

11.Click Next.

You're finished adding this account.

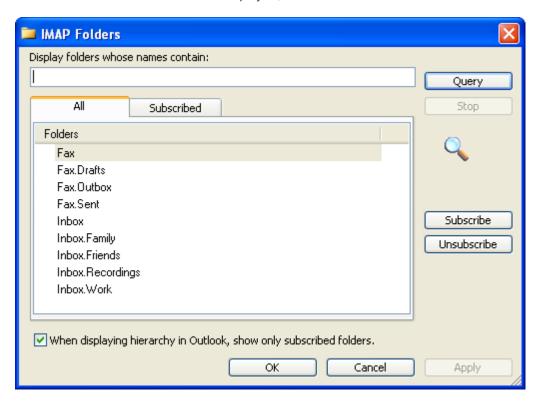
## Subscribing to Folders

When you have your Switchvox IMAP account set up in your desktop email application, you can manage the same folders that are in the Switchvox User Suite Mailbox (**Voicemail / Fax > Mailbox**).

You can 'subscribe' to all of the folders, or just the ones you want to use. If you 'unsubscribe' to a folder, it still exists in your IMAP account, it just isn't shown in your email application. You can always subscribe to it again later.

You can also create folders using your desktop application, and those folders are accessible in your User Suite Mailbox.

To subscribe to your IMAP folders in Microsoft Outlook, highlight your Switchvox IMAP Inbox and select **Tools > IMAP Folders...** The *IMAP Folders* window is displayed, shown here:



In Mozilla Thunderbird, highlight the account and select **File > Subscribe**. In either case, a list of folders is displayed that lets you subscribe and unsubscribe.

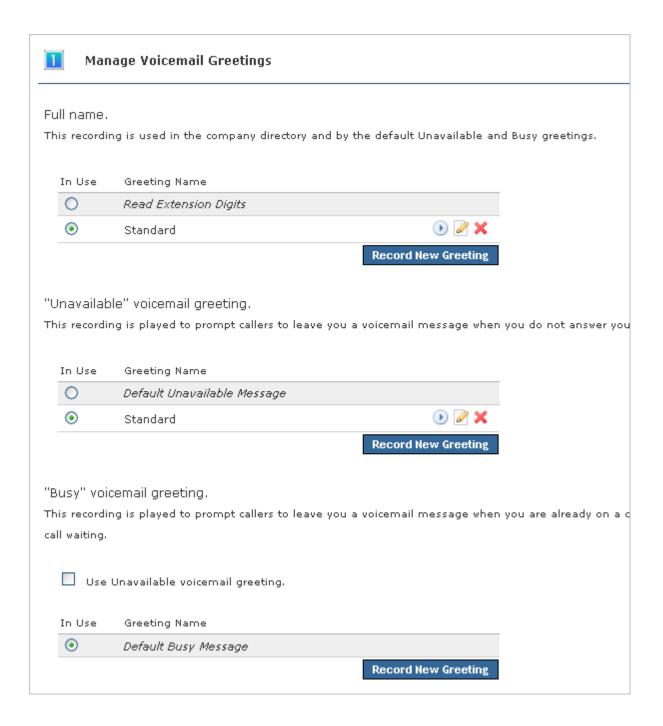
# **Voicemail Options**

Your Voicemail Options include setting up your voicemail greetings and defining how you want to receive notifications of a new voicemail message.

# **Managing Greetings**

Manage Voicemail Greetings lets you set up all of your greetings that might be played to a caller when you don't answer the phone. You can upload a sound file from your computer, or record a greeting using your phone.

The default greeting, in every situation, is for Switchvox to read your extension digits and then let the caller record a voicemail message. You can record or upload your name, and multiple custom messages for when you don't answer the phone, or you are already on a call and don't answer the phone. If you don't want to distinguish between those situations, make sure you have created an Unavailable Voicemail greeting, then check the box under the "Busy" voicemail greeting, **Use Unavailable voicemail greeting**.



# **Setting Notifications**

Voicemail notification options let you customize how you want to receive a notification when you get a voicemail message. You can set up multiple email addresses, each with an email template and instructions for attaching the voicemail as a WAV file.

**Note:** Notification email messages do not synchronize with your Switchvox Mailbox. That means that if you listen to a voicemail WAV file attached to a notification message, Switchvox does not know that you have 'read' the voicemail.



To add a new notification option, click **Add New Notification Email**. Enter your Email address, select a template, and select Yes or No to indicate whether or not to attach the voicemail WAV file to the email message.

To edit a notification option, click its **Edit** button.

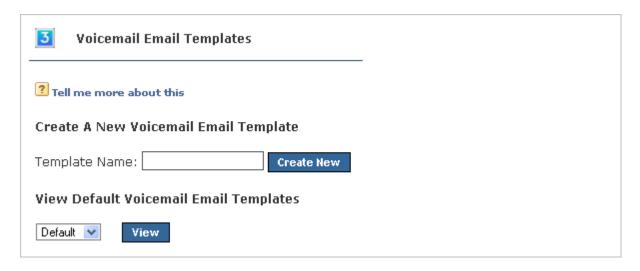
To delete a notification option, click its **Delete** button. You cannot delete the **local copy** option (that is the copy in your Switchvox IMAP mailbox), but you can change the template. This is the template that you will see if you subscribe to your Switchvox IMAP mailbox using your desktop email application.

## Delete local copy of voicemail

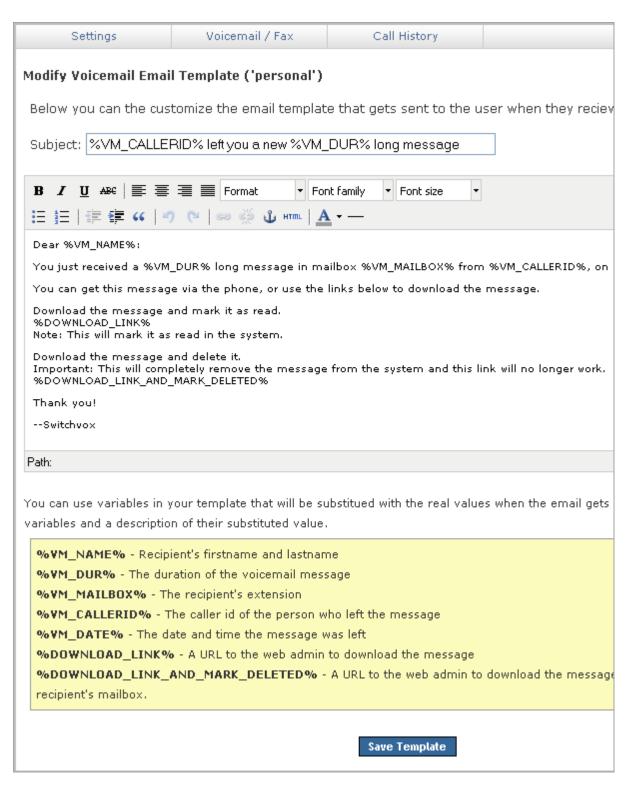
To delete the Switchvox copy of the voicemail once its sent to your email address(es), check the box **Delete local copy of voicemail**. This option lets you handle your voicemail entirely through your own email account.

# **Managing Notification Templates**

Voicemail Email Templates let you customize your voicemail notification email messages. You can create multiple templates and tailor them to different types of email accounts. These custom templates are available (along with default templates) when you create or edit a Voicemail Notification Option.



You can use variables in your template that are substituted with real values when the email gets sent. For example, %VM\_CALLERID% in a template prints as the caller's name and phone number. There is a full list of these variables on the pages where you create or edit a template.



# **Automatically Forwarding Voicemail**

You can automatically forward all of this extension's voicemail messages to other extensions in Switchvox. This is helpful for generic extensions such as 'Accounting,' or 'Shipping,' or for queue extensions such as

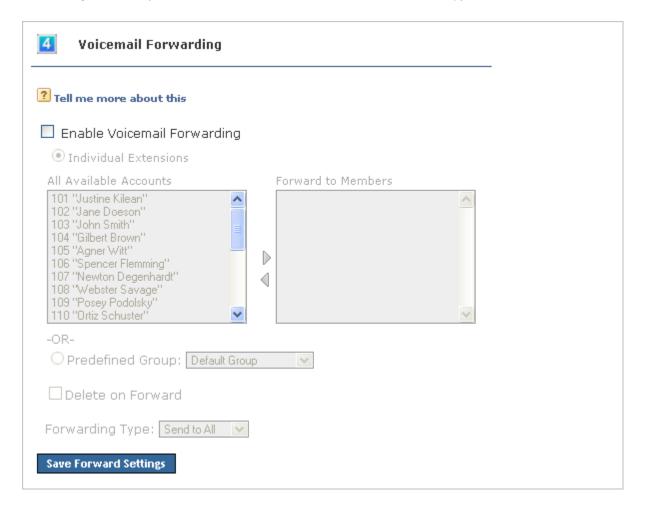
'Customer Service.' Callers can leave messages on one extension, then all of those messages are forwarded to the appropriate people. This is also useful internally, if you want to send a message to everyone in a group.

To define a specific set of extensions to receive the forwarded voicemail, select **Individual Extensions**, and move the appropriate extensions into the box on the right.

To specify an Extension Group to receive the forwarded voicemail, select **Predefined Group**, then choose an Extension Group from the dropdown.

To delete each message from this extension's Mailbox after the message is forwarded, check the box **Delete** on Forward.

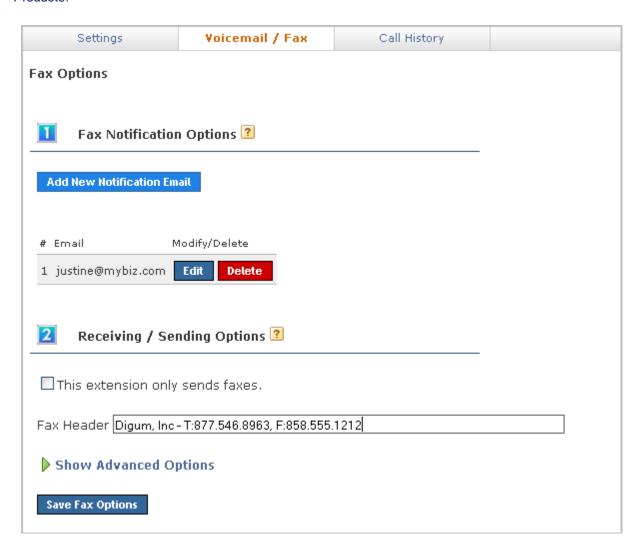
To send forwarded messages to all of the recipients, choose *Send to All* from the **Forwarding Type** dropdown. To send forwarded messages to one recipient at a time, choose *Round Robin*. With *Round Robin*, Switchvox forwards each message to the next recipient, moving through the list of recipients in the same order as they are listed (in the extensions box above, or in the Extension Group).



# **Fax Options**

Fax Options let you set up your faxing environment. You can enter multiple email addresses to receive notifications of a new incoming fax, create a fax header, and define this extension as only sending faxes.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.



# **Fax Notification Options**

Add, Edit, or Delete the email addresses that you want to receive notifications of a new incoming fax.

This is useful if you want to be notified at both your work and personal email addresses, or if you want your administrative assistant to be notified.

# This extension only sends faxes

If this extension is really a fax machine and won't make any voice calls, check the box **This extension only sends faxes**.

Switchvox handles all outgoing activity as a fax:

- o Print a fax-file
- o Put it in the Fax. Outbox folder in that extensions's Mailbox
- Send the fax

## **Fax Header**

Enter a Fax Header to be sent with each fax that you send. For example:

```
Digium, Inc. | 877.546.8963 | Fax: 858.555.1212
```

This is the information that is printed along the very top of your fax pages (not to be confused with a cover page).

## **Advanced Options**

It is unusual to change the Advanced Options, so they are hidden. If you find that you need to change an Advanced Option, click **Show Advanced Options**.

#### **Local Station ID**

Text string that identifies the sender identification to the remote side of the fax transmission. Default is to leave blank.

## Disable Error Correction Mode (ECM) for G711 fax sessions

Disable error correction mode (ECM). The default is to leave this feature enabled.

#### Minimum transfer rate for fax transmissions

Minimum transfer rate used during fax rate negotiation. The default minimum transfer rate is 2400.

### Maximum transfer rate for fax transmissions

Maximum transfer rate used during fax rate negotiation. The default maximum transfer rate is 14400.

## Number of redundant signal packets for T38 fax sessions

T38 Error Correction Code (ECC) Signal Redundancy. This option specifies the number of redundant signal packets contained in a T38 signal frame. The default value is 3.

### Number of redundant image packets for T38 fax sessions

T38 Error Correction Code (ECC) Data Redundancy. This option specifies the number of redundant image packets contained in a T38 image frame. The default value is 1.

## Number of maximum expected T38 packet delay

Maximum expected T38 packet delay in milliseconds. The result of this specification is all T38 packets are padded to prepare the fax stack for the specified delay. As long as the packet delay is less than or equal to the specified value, the fax stack will not realize that there is an actual packet delay because of the packet padding. The default value of 800 milliseconds should cover most use cases.

# Creating & Sending a Fax

Sending a fax from your desktop computer using Switchvox is a two-step process:

- 1. Open a file in any application and print it to the Switchvox fax printer.
  - This stores a copy of the file in your Switchvox Mailbox Fax Drafts folder.
- 2. Go to the Switchvox User Suite **Voicemail / Fax > Mailbox**, select the *Fax Drafts* folder, and click the fax machine icon (at the top of the list) to send that file to a fax machine. For details, see Faxes.

You must install a fax license and the fax software to make faxing available. For details, see Digium Addon Products.

## **Setting up a Fax Printer**

To make faxes for Switchvox, you need to first set up a generic printer that you can use from any application. You only have to do this once.

Depending on your desktop environment there are many ways to set up a printer, but there is some specific information that you need to set up a Switchvox Fax Printer:

o For Windows-based operating systems, the URL is

```
http://YourDomain:631/printers/faxprinter
```

Where YourDomain is the domain that you normally use to go to your Switchvox User Suite. This may be an IP address or a domain name.

o For Linux- or Macintosh-based operating systems, the URL is

```
http://ext:password@YourDomain:631/printers/faxprinter
```

Where <code>ext</code> is your extension, <code>password</code> is the password that you use to log into the Switchvox User Suite or to collect your voicemail, and <code>YourDomain</code> is the domain that you normally use to go to your Switchvox User Suite. This may be an IP address or a domain name.

For any operating system, use a generic postscript printer driver.

Our suggestion for Windows-based operating systems is the driver from IBM: *IBM Network Printer 17 PS*. For Linux- and Macintosh-based operating systems you should choose a generic postscript printer driver.

## Sample Printer Setup

The following gives an example of setting up a Switchvox fax printer for a computer running a Windows XP Professional operating system.

1. From the Start menu, go to Printers and Faxes.

The Printers and Faxes window is displayed.



## 2. Click Add a Printer.

The Add Printer Wizard is displayed.



## 3. Click Next.

The Local or Network Printer windows is displayed.



4. Select A Network Printer...

## 5. Click Next.

The Specify a Printer window is displayed.

Add Printer Wizard	
Specify a Printer  If you don't know the name or address of the printer, you can search for a printer that meets your needs.	
What printer do you want to connect to?  Browse for a printer	
O Blowse I	or a printer
Connect to this printer (or to browse for a printer, select this option and click Next):	
Name:	
	Example: \\server\printer
<ul> <li>Connect to a printer on the Internet or on a home or office network:</li> </ul>	
URL:	http://pbx.mybiz.com:631/printers/faxpr
Example: http://server/printers/myprinter/.printer	
	< Back Next > Cancel

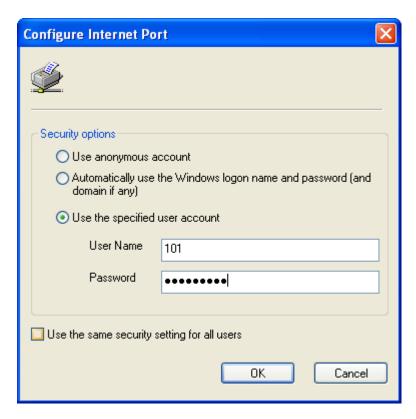
6. Select Connect to a printer on the Internet or on a home or office network, and enter the following URL:

http://YourDomain:631/printers/faxprinter

Where YourDomain is the domain that you normally use to go to your Switchvox User Suite. This may be an IP address or a domain name.

## 7. Click Next.

The Configure Internet Port window is displayed.



- 8. Select *Use the specified user account*, and enter your extension number and your password. This is the same password that you use to log into the Switchvox User Suite or to collect your voicemail.
- 9. Click OK.

The manufacturer and model window is displayed.



10 Select *IBM* from the Manufacturer list on the left, then select *IBM Network Printer 17 PS* from the Printers list on the right.

We suggest this printer driver because it is a widely available, standard printer driver. If you are using a different operating system, choose a generic postscript printer driver.

The Printer Wizard copies the appropriate files for you and sets up your new printer.

Wait until the Default Printer page is displayed, then go on.

11 Select Yes or No, depending on whether you want this to be your default printer. (You probably want to select No).

The final page of the Printer Wizard is displayed.

12.Click Finish.

In the Printers and Faxes window, you can see the new printer faxprinter on http://YourDomain:631

## Sending a Fax

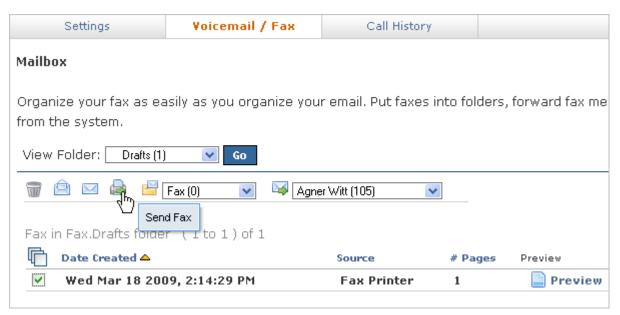
Now that you have your fax printer set up, you can create a fax from any application.

- 1. Select the Print option from the application that you are using to view this document. (Or any open file that is handy.) In the Print dialog box, select the new 'faxprinter' as the printer to use, and print the file.
- 2. In your Switchvox User Suite, go to **Voicemail / Fax > Mailbox**. Select *Fax:Drafts* from the **View Folder** dropdown.

Your *Fax.Drafts* folder is opened, and your drafts are listed. The test page that you printed is in the list, and you can preview the first page, or download a PDF of the entire fax.

Check the box for your fax, then click the fax machine icon (at the top of the list). You are prompted to enter a fax number. Enter the number for the fax machine you want to send this fax to, then click **Send Fax**.

Your fax is in the *Outbox* folder while it is being sent, and moved to the *Sent* folder when it has been successfully sent.



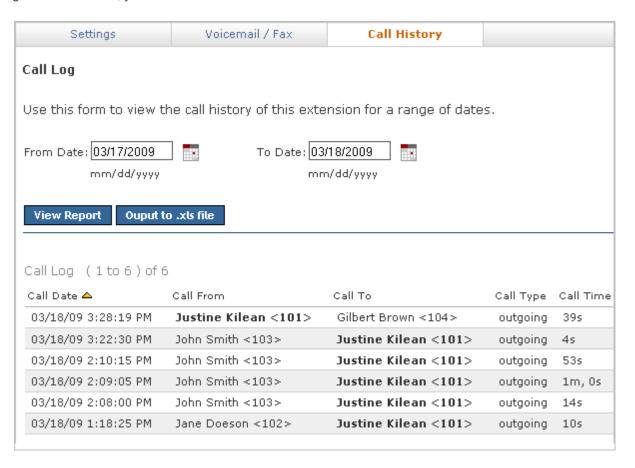
# **Call History**

Under the Call History tab are two sections, Call Log and Call Reporting. These allow you to track your phone use.

# **Call Log**

The Call Log is a simple list of the calls that have been made to or by your extension. You can select a date range by setting the **From** and **To** dates. Click **View Report** to see log in the User Suite, or click **Output to** .xls file to download an xls file.

By default, a list of all incoming and outgoing calls is displayed. The list includes the call date, who the call was from and to, the type of call (incoming or outgoing), the call time, and the talk time. If you are viewing the log in the User Suite, you can mouse over the Call Details to see more details about the call.

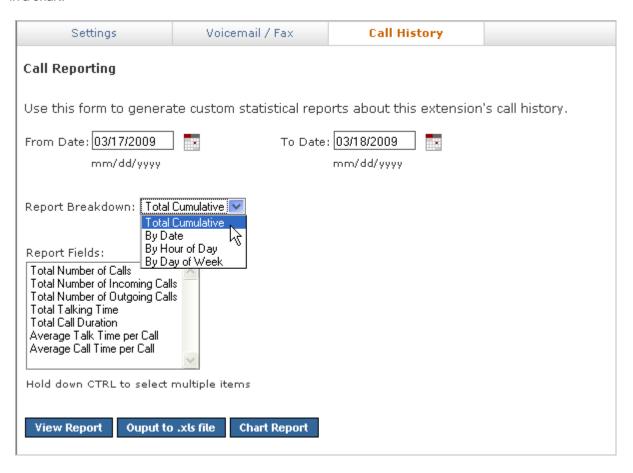


# **Call Reporting**

This lets you generate a report about call activity. Call reports are generated for the date-range and criteria that you specify, and include the information that you specify.

In the Report fields section you select one or more of the fields you would like included in the report or graph. After you have chosen the desired fields you can choose how you want the report broken down. By default the reports are broken down by date such that a data point is displayed for each day between the From and To date.

You can instead opt to have the report broken down by the Hour of Day, the Day of Week or just one data point for the whole range by selecting Total Cumulative. When you have finished specifying your criteria, you can opt to have the report printed to the screen, sent to your desktop as an .xls file or view it in graphical form in a chart.



# The Switchboard

The Switchboard is a graphical display of what's happening on Switchvox. It shows your own calls, your coworkers' calls, call-queue activity, and your parking lot. You can drag and drop to transfer calls, and one click lets you make a call, pick up calls, record or monitor calls, log in and out of call queues, and more.

In addition to all of the Switchvox interaction, the Switchboard is also 'Web Aware,' which means that you can integrate other web applications with your call activity. Built-in panels bring SugarCRM, Salesforce, and Google Maps to the Switchboard, and you can build custom panels that use the URL of your choice.

To open your Switchboard, click Launch Switchboard (in the upper right corner of your User Suite).

In the upper right of the Switchboard, you can see the number of voicemail messages that you have, new and old. There is also an **Options** menu that lets you control your Switchboard.

Your Switchboard lets you have up to 6 active lines on your extension (in the Current Calls panel), regardless of your phone's capabilities.

# **About the Options Menu**

The **Options** menu lets you control what is on your Switchboard, and how it is laid out. It also let you set your status for your fellow Switchvox users.

# **Save Layout**

**Save Layout** lets you save your Switchboard layout. The next time you open the Switchboard, the window is the same size, and the same panels are open in the same place.

## **Panels**

**Panels** let you open and close all of your Switchboard panels. Each of your Phonebooks is available as a separate panel.

Your Switchvox administrator must give you permission to use many of the features available in the Switchboard Panels. If you see a feature described here that is not available to you, please ask your administrator.

## My Presence

**My Presence** lets you set your Switchboard presence so that your co-workers can see your status and decide how to communicate with you.

The Away and Extended Away options let you include a comment so that you can let people know when you might be back.

# **Panels**

The Switchboard panels let you see into the activity on Switchvox and manage your calls.

## **Current Calls Panel**

The Current Calls panel displays all of the active calls on your extension, and gives you many options for handling those calls. You can have as many as six current calls in the Switchboard, even if your handset does not support that many.

## Send Call

If you choose not to answer a call, click **Send** to send the call to your Call Rules. The default rule is to send your calls to voicemail, so most likely **Send** means the caller is sent to your voicemail.

## Hold & Resume

When you are on a call, one click puts that call on Hold, or Resumes the call.

Note: Resume is not available until you put the call on Hold.

#### **Take Another Call**

You can put your active call on Hold, and then answer another incoming call. You can have as many as six current calls in the Switchboard, even if your handset does not support that many.

## Transfer: Assisted & Blind

To transfer a call to someone in your Phonebook, just click the call and drag it from Current Calls to the Phonebook entry. You can transfer a call at any time, even if you have the call on hold, or the other extension is on an active call.

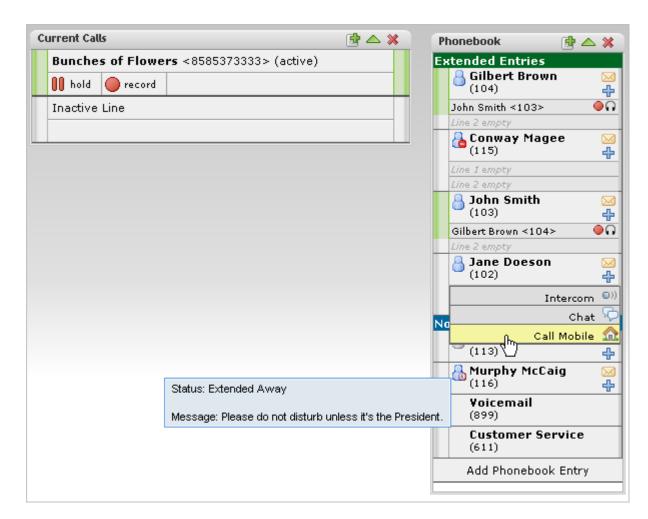
In an assisted transfer, you would put the caller on hold, call your co-worker to let him know you'd like to transfer a caller, then complete the transfer. In a blind transfer, you would just drag the call to the Phonebook entry.

## **Transfer to Voicemail**

To transfer a call to your coworker's voicemail, just click the call and drag it from Current Calls to the Phonebook entry's envelope icon. This is handy if you know your co-worker isn't able to take the call.

#### Record

When you are on a call, you can click to start and stop a recording of your call. The recording is sent to your voicemail Mailbox, where you can save or forward it. Don't forget, it's up to you to let the other party know that you're recording the call.



# **Directory Panel**

The Directory panel offers you access to all of the extensions in Switchvox. You can click and drag a call to an entry in the Directory Panel to transfer the call to that extension.

# **Parking Lot Panel**

The Parking Lot panel lets you 'Park' a call so that the call can be picked up at a different extension. To Park a call, just click and drag the call to the Parking Lot panel. Then, you or any of your coworkers can pick up a phone and dial the Parking Lot extension to be connected to the caller.

An extension must have permission to pick up a parked call.

## **Profile Panel**

The Profile Panel shows you the profile of the extension-owner you are talking to. Profile information can be entered by the Switchvox administrator, or by the extension owner.



An Extension Profile includes some or all of the following:

- Image
- First and Last Name
- o Title
- Location
- o Extension Number

Profiles can be shown in the Switchboard and on a phone that uses a Phone Feature Pack. For details, see Your Phone Feature Pack.

This panel also offers several one-click options to reach this person:

Chat

Start a chat with this extension.

Call

Call this extension. The Switchboard rings your phone, and then when you answer it rings this extension. (You may be able to set up your phone so that it automatically answer calls from the Switchboard; see Phone Options.)

Voicemail

Call the voicemail for this extension. The Switchboard rings your phone, and then when you answer it rings this extension's voicemail box so that you can leave a message. (You may be able to set up your phone so that it automatically answer calls from the Switchboard; see Phone Options.)

## **Phonebook Panels**

Each Phonebook panel displays the entries in that Phonebook. You can click on any entry to dial that number. Some extensions can display as Extended Entries, to help you communicate more effectively with that person.

#### **Normal Entries**

Normal Phonebook entries show your coworkers' extensions or your frequently dialed external phone numbers. Each extension is shown with its Presence information, and you can click on any of these to dial them, like a speed dial button.

Also, Normal entries provide a 'plus' icon that includes options to dial that person's Additional Numbers, chat with him, or Intercom to him.

You must have permission to use Chat and Intercom. Chat may not be available if for some reason you can't chat with that person.

### **Extended Entries**

Extended Phonebook entries have all of the normal extension options, but they also let you do the following with that extension's calls:

- · Pick up an incoming call
- · See who the caller is
- · Record the call
- Monitor the call (the callers do not hear you)
- Barge into the call (both callers hear you)
- · Whisper on the call (only your co-worker hears you)

You must have permission to use each of these options.

#### **Presence**

All Phonebook entries display the person's **Presence**, which helps you and your coworkers be selective about what calls to transfer to each other. An *Away* or *Extended Away* Presence can include comments.

#### Intercom

All Phonebook entries can include the **Intercom** option, which lets you talk to a coworker through the speaker on his phone. With Intercom, your recipient doesn't even need to pick up the handset to answer your call.

You must have permission to use this.

### **Pickup**

Extended Phonebook entries can include the **Pickup** option (the green 'up' arrow). One click and your coworker's incoming call rings your extension also.

You must have permission to use this.

#### Record

Extended Phonebook entries can include the **Record** option. One click records the call, and sends the recording to your voicemail Mailbox.

You must have permission to use this.

### **Monitor**

Extended Phonebook entries can include the **Monitor** option. If you monitor a call, the callers cannot hear you.

You must have permission to use this.

#### Barge

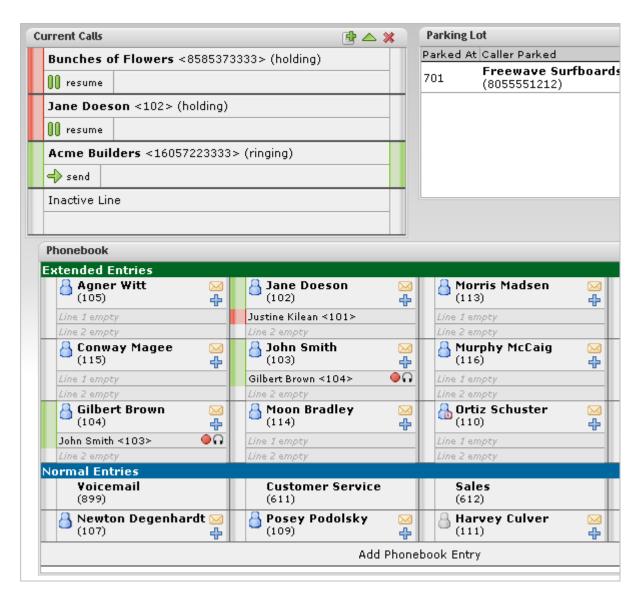
Extended Phonebook entries can include the **Barge** option. If you barge into a call, both of the callers can hear you.

You must have permission to use this.

#### Whisper

Extended Phonebook entries can include the **Whisper** option. If you whisper into a call, the caller cannot hear only (only the extension owner can hear you).

You must have permission to use this.



## **Chat Panel**

The Chat panel lets you chat with your Switchvox co-workers. The main page of the Chat Panel offers a list of people you can double-click to start a chat. Or, enter a Jabber ID and click **Start Chat**.

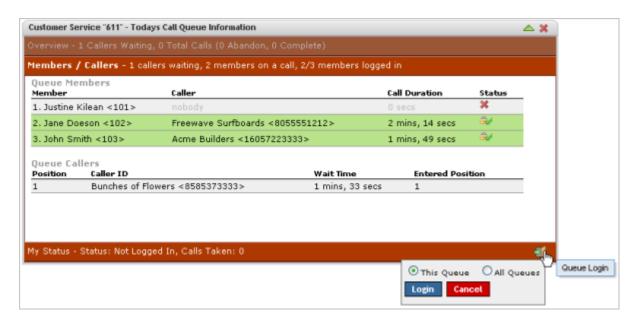
You must have permission to use this.

## **Queue Panels**

Each Call Queue panel in your Switchboard contains views into that queue's activity and statistics for the day.

For queue members, you can log into or out of each queue with one click. Or, you can pause your status and add a comment, letting your co-workers know why you're unavailable and when you'll be back.

You must have permission to use this.



## **CRM Panels**

These time-saving panels integrate with your CRM application. When a call rings your phone, all of the information for this customer is displayed immediately, before you even answer the phone!

You must have permission to use this.

# **Google Maps Panel**

The Google Maps panel shows you where your caller is on a map based on area code and prefix. A popup window lets you scroll through previous callers' locations.

You must have permission to use this.

## **Custom Panels**

Your Switchvox administrator may have set up custom panels that integrate with Web applications. Ask your administrator for more information.

# Popup URLs

Your Switchvox administrator can write external applications that set a display URL for each incoming call. If this value is set, then the URL button lights up and clicking it opens a new window with the specified URL. If you want that window to automatically open when a call comes in, click the green plus sign at the top-right of the panel.