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COMMUNICATIONS



Switchvox

Administrator Manual

5.5



Switchvox Administration Reference Manual

Switchvox SMB version 5.5

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Setup

This section describes the tools under the Setup category. These tools will enable you to manage your extensions and phones, set up call routing, and give your administrators the appropriate permissions.

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Extensions

The Extensions tools give you a means to create and manage Switchvox extensions. They include the following topics:

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Manage (Extensions)

When you select Manage, by default, it displays a list of All Extensions in your PBX system, 50 per page, with their extension number, type, and name of the person assigned to the extension.

You can filter this list of extensions by selecting the desired Extension Type from the dropdown. Options are to display only phones, IVRs, call queues, or virtual extensions.

View Extensions

To filter the list of extensions, select an extension type from the View dropdown. The list then displays only extensions of the selected type.

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

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

Password Warnings

There may be warning icons displayed in the Extension column next to any extension if the system has evaluated its passwords as weak. Weak passwords can make your Switchvox vulnerable to intrusion.

By default, Switchvox requires strong passwords be used for extensions and phone-registrations. These weak passwords may have been created prior to a strong password requirement, or you may have turned that requirement off. For details about requiring strong passwords, see [“Settings” page 30](#).

For details about setting an extension password, see [“Numeric Password.” page 9](#). For details about setting a phone-registration password, see [“Phone Password” page 13](#).

Log In

Click an extension’s **Log In** icon to open its Switchvox Extension Tool Suite. This lets you access that extensions' Extension Suite without knowing or entering its password. The extension-owner can be logged in at the same time, so be careful when either of you are making changes.

Create an Extension

The **Create Extension** button defines the new extension type and template, then displays an appropriate Profile Information page to complete the extension definition. After completing the profile information, **Save Extension** saves everything and redisplay the Manage Extensions page, which now lists the new extension.

NOTE: If you have not created your own template, leave the template field as Default.

Extension Types

There are many different types of extensions in Switchvox.

The phone and virtual extension types (SIP, Analog, and Virtual) have most of the same settings, and they have a corresponding Extension Suite account.

Other types of extensions do not have a user name and password for a Extension Suite account.

See the following sections for specific information for the following extension types:

- [“Phone and Virtual Extensions” page 8](#)
- [“IVR Extensions” page 15](#)
- [“Simple Conference Room Extensions” page 19](#)
- [“Meet-me Conference Center Extensions” page 20](#)
- [“Group Pickup Extensions” page 24](#)

- [“Intercom Extension” page 21](#)
- [“Directory Extensions” page 22](#)
- [“Voicemail Access Extension” page 23](#)
- [“Call Parking Extensions” page 23](#)
- [“Dial Tone Extension” page 24](#)
- [“Call Queue Extensions” page 15](#)
- [“Agent Log In Extensions” page 20](#)
- [“Agent Log Out Extensions” page 21](#)
- [“Feature Code Extensions” page 24](#)

Phone and Virtual Extensions

The settings are mostly the same for SIP phones, Analog phones, and Virtual extensions. These extensions have a corresponding Extension Suite account (/main):

- **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. This is an appropriate extension-type for a fax machine or an intercom.
NOTE: We strongly recommend you do not use this option for your phone extensions. Use instead analog telephone adapters, or a channel bank. If you choose to use Analog Phone extensions with phones physically plugged into Switchvox, those phones will not be able to take full advantage of the Switchvox features.
- **Virtual Extension.** An extension without a phone or a Switchboard. This type of extension behaves like a SIP Phone extension except 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.

Profile Information

Extension. A unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

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 Groups” page 37](#)).

First Name. 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.

Last Name. 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.

Email Address. The email address of the owner of this extension. This is used to send voicemail-notification messages.

Location. The physical location of the owner of this extension. This is used in the extension Profile.

Title. The title of the owner of this extension. This is used in the extension Profile.

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

NOTE: All languages and locales except English for United States handle dates with the format of dd/mm/yyyy (as opposed to mm/dd/yyyy).

By default, new extensions are created as English for US, with English as the Sound Prompt Language. If you want new extensions to default to your preferred language, use [“Templates” page 29](#).

Numeric Password. A Numeric Password for this extension. The password is numeric because it will be entered from the numeric keypad on a telephone.

The extension owner uses this password to log into their account in the User Tool Suite. The strength of a 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.

You can require that the extension-owner enter a strong password. For details, see [page 30](#).

Profile Picture

You can upload and modify a picture to include in your profile information.

- Click **Upload Picture**.
- Click inside the text box to open a dialog box and find the image file on your computer. Images must be at least 75x100 pixels, and must be in jpg format.
- 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 after it is cropped.
- Click **Upload Picture**.

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

- Click **View Picture**.
- Click **ReCrop**, or **Delete**. If you want to upload a different image, Delete this image now, and then click Upload Picture to start again.
- Confirm your changes. Check the appropriate boxes if you want the extension owner (user) to be able to change his or her Location, Title, or Picture.

Login Actions

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 that person to create a new password before continuing to use the software. (See also [page 30](#).)

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 logs 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 you need to change an Advanced Option, click Show Advanced Options.

General Settings

Voicemail/Fax Mailbox Quota (MB). This value is the maximum number of megabytes that can be used for voicemail messages and faxes. It can be set for extensions and extension groups.

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

Sound Prompt Language. If there are one or more sound packs available in different languages, those languages are displayed in this dropdown. Select the language that this extension hears (e.g., when calling the voicemail access extension, the prompts are in Italian).

To obtain sound packs for different languages, see www.switchvox.com/product_addons.

For instructions on using sound packs, see [“About Sound Packs” page 94](#).

Minimalist Mode. This mode presents the Web Suite interface with reduced graphics. If Minimalist Mode is set to Yes, the following apply:

- The background graphic and larger graphics on buttons and tabs are removed.
- Some text, especially in section headers, is smaller.
- Animations are removed.

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.

Phone Settings

Digium Phone Settings. These settings enable you to control everything about your Digium phone. To assign a Digium Phone to an extension, see [“Digium Phones” page 31](#).

Volume levels for the ringer, speaker, handset, and headset are controlled from the volume bar on the phone itself.

NOTE: The settings are also available in Extension Templates and Bulk Modify, for your convenience.

- **Line Label** is the label displayed on the phone's Line Key for this extension. You can enter a specific label, or use variables to describe what the label should be. Variables are useful in Extension Templates so you do not have to set a unique label for each extension.
- **Msgs Button Action** is the action that happens when the Msgs button is pressed. Use **Open Visual Voicemail** to use the phone Voicemail application to navigate via the phone display, or use **Dial Voicemail Extension** to call the voicemail-system extension (default extension number 899) and navigate via sound prompts.
- **Idle Screen** is the image that displays on the phone's idle screen.
- **Display Missed Calls Notification.** If this is YES, a Missed Calls notification is displayed in the top-left corner of the phone display. If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Brightness and Contrast** control the phone display. If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Backlight Dimming.** If this is YES, the phone's backlight will dim to the specified level after the phone is not in use for the specified number of seconds. If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Enable Line Keys for Contacts.** If YES, the phone's unregistered Line Keys are used as Rapid Dial Keys.
- **Default Ringtone** is the ringtone used for incoming calls (unless the extension-owner has set up one or more Ring Rules). If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Reset Volume Every Call.** If YES, the call volume will be reset to the default after each call. The volume can be changed during the call, using the volume bar. If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Headset Answer.** If YES, when the user presses the Answer softkey during an incoming call, the call is answered on the headset, not the speaker. This setting should not be confused with the Auto-answer Switchboard initiated calls feature under the Common Settings tab. If this is changed on the phone itself using Preferences, then you will no longer be able to change it here in the web-based suite.
- **Advanced Options**
 - **Accept Local Calls.** If set to This PBX's Networks, then the phone accepts calls from any of the networks described in Phone Networks. If this is set to Any Host, then the phone accepts calls from any host.

- Local SIP Port** is the phone's port on which SIP traffic is transmitted. You should not need to change this, but if you do you must be sure that the SIP traffic to and from the phone is routed correctly.
- Override Dial Plan
- Phone-admin Password
- Reset Phone-registration Password on Reassignment from Phone
- Audio Codecs

Common Phone Settings

These 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.

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 Phone Settings

These can be set 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 the Extension-Owner's User Guide, [*"Phone Features" page 12*](#).

Show caller profile on phone. These options determine whether or not the Switchvox Profile feature is used on the phone.

Advanced 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).

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.

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
G729	30	Good	Medium
SPEEX	33	Great	High
GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

Common Settings

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.

Phone Password. Enter your phone password and verify it. (See also [“Settings” page 30.](#))

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

This password may be different than the password used to administer the SIP phone from the phone itself. Also, this password does not change when this extension's voicemail password changes.

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.

Permissions

These permissions specify which features the extension is allowed to use.

Change Profile Information

These settings let you control which parts of a profile the extension-owner can change.

Create Call Rules. These settings let you control which Call Rule actions the extension-owner can create.

Use Voicemail Features. These settings let you control which voicemail features the extension-owner can use.

Use PBX Features. These settings let you control which general PBX features the extension-owner can use.

Use Digium Phone Apps. These settings let you control which Digium Phone Apps are on the extension's Digium Phone.

Change Digium Phone Settings. These settings let you control which of the Digium Phone Settings the extension-owner can change.

Use Distinctive Ringtones. These settings let you control which Ringtones features the extension-owner can use.

Use Calling Features. These settings let you control whether the extension-owner can use these calling features.

Use Phone Feature Pack Phone Features. This lets you control whether the extension-owner can use the PBX Applications on a Polycom phone.

Outgoing Call Rules

Specify how this extension can interact with your [“Outgoing Call Rules” page 53](#).

- To allow this extension to use a rule, put the rule in the collection as Allow.
- To deny this extension the use of a rule, put the rule in the collection as Deny.
- To ignore an Outgoing Call Rule entirely, do not include the Rule in this collection.

Extension Groups

This collector box lets you indicate this extension's membership in Extension Groups.

To put an Extension Group into the Extension Groups collection:

- Type a name into the text field. As you type, Switchvox offers suggestions of matching groups.
- Click the Find icon to display the groups available in Switchvox.

To delete an item from the collection, select it and click the Delete icon.

To select multiple items in the collection:

- Control-click to select two or more items that aren't together in order.
- Click an item, and then Shift-click another item to select both items and all items between them.

IVR Extensions

This is an extension that presents a menu to the caller, or performs an action. (It is also known as ‘Auto-Attendant’ in some PBX systems.)

You must have an IVR to create an IVR extension.

For information about setting up IVRs, see [“Interactive Voice Response \(IVR\) Editor” page 60](#).

The default extension 800 is set with the IVR menu Example IVR. You can customize that IVR, or create your own.

IVR Extension

Enter a unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

Route to IVR Menu

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.)

Call Queue Extensions

A Call Queue extension lets an incoming call ring a group of extensions so that any one of those extensions can pick up the call.

Queue Setup

The Queue 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 [“Settings” page 30](#).

Queue Name

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

Ringling 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.

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.

This collector box lets you find and collect extensions or groups that belong in this queue.

To put an extension or group into the collection:

- As you type a name or extension number into the text field, Switchvox offers suggestions of matching extensions.
- Find icon displays the extensions available in Switchvox. You can sort this view by the column headers. If you are working with phone-type extensions, you can expand the window to show an extension's profile.
- To mark an extension or group as Permanent or Login, select them and click the appropriate icon.

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. If you have an extension group in the queue, the group's order is used. Any non-phone extensions in a group are ignored.

To put items in the right order, select the items you want to move, then click the Up, Top, Down, or Bottom button.

To select multiple items in the collection:

- Control-click to select two or more items that aren't together in order.
- Click an item, and then Shift-click another item to select both items and all items between them.

To delete an item from the collection, select it and click the Delete icon.

Queue Member Settings

These settings determine how queue members behave in the queue.

Seconds to Ring Each Member. This is 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. This is 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 in which no one is logged in, there is potential for system performance issues because the system keeps trying to find someone to answer the queue call. Unless you have a specific reason for doing so, leave this value at the default of 5.

Seconds for Wrap Up. This is 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. This is the selected 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” page 93](#).

Acknowledge Call. Yes indicates members must press 1 to accept the call.

If the member hangs up without pressing 1, then the call goes to the next appropriate queue member.

Auto Log Off After Missed Calls. 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 or she is automatically logged out of this queue. Calls will not ring this member until he or she logs in again.
- If you do not want members to be logged out automatically, set this to 0 (zero).

Receive calls while on a call. Yes indicates that 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.

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

Queue Callers Hear. Indicates whether callers should hear ringing or Music On Hold while they are waiting in the queue.

If you select Ringing, then Music On Hold and Announcements options are not available.

Music On Hold Group. Indicates the Music On Hold Group you want to play for callers waiting in this queue.

Announce Position in Queue

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

Announce Estimated Hold Time in Position Announcements. If yes, 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. This is the 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” page 93](#).

The following selections are available for announcements:

- **Language:** Plays 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 queue, select that language.
- **You are next:** The call is now first in line and will be answered by the next available representative.
- **Caller Number:** This announces to the caller what numeric place he/she has in the answer queue.
- **Calls waiting:** Waiting to speak to a representative.
- **Hold time:** Announces to the caller the current estimated hold time.
- **Minutes:** Minutes
- **Thank you:** Issues this message: Thank you for your patience

In Queue Call Routing

The In Queue Call Routing settings determine any alternative call routing.

Values entered in the boxes define queue call routing.

- 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.

Switchboard Panel Permissions

- Sets the permission level for extensions to display this queue in the Queue Panel on their Switchboard.
- Adds extensions or extension groups with one of the following permission levels:
My Status: Lets the extension open this queue panel and display his call information for today.
Overview: Lets the extension open this queue panel and display 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 display his call information for today, an overview of all the queue information for today, and the members and callers in the queue.

Simple Conference Room Extensions

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

Conference Room Extension. This is the unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

Play Sound on Entry/Exit

Yes indicates you want a sound to play as callers enter and exit the conference.

Play Music On Hold for One Person. Yes indicates Music On Hold will be played when there is only one person in the conference.

Out of Conference. This is the extension number to transfer calls to when a caller presses # in a conference.

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.

Meet Me Conference Center Extension

This is the unique number for the extension. For information about the extension-number length, see [“Settings” page 30](#).

Number of Invalid Attempts

This is the number of attempts to allow before transferring the caller to the specified extension.

Route Caller to

This is the extension number to transfer calls to if the caller has entered too many invalid conference room numbers.

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

This is the unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

Do not require password to login

YES indicates when people call this extension, they are logged in without having to enter a password.

Login/Logout toggle

Yes indicates members can call this extension to be logged in or out. If already logged in, callers are logged out. If 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

YES indicates members must 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

YES indicates 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?

Queues to Affect

Affect All Queues: YES indicates all of the queues in the system are affected.

Otherwise, only the indicated specific queues into the collection are affected.

If you don't want to affect call queues with this Agent Log In, do not select any queues.

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 [“Settings” page 30](#).

Queues to Affect

Affect All Queues: YES indicates that all of the queues in the system are affected.

Otherwise, enter specific queues into the collection.

If you don't want to affect call queues with this Agent Log In, do not select any queues.

Do not require password to log out

YES indicates this is true.. When people call this extension, they are logged out without having to enter a password.

Affect virtual extension login/logout status

YES indicates if the log-out action affects virtual extensions. For example, if a virtual extension is logged into a phone and into a queue, it indicates if the log-out action applies to the virtual extension, or just the queue.

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 /Paging Settings

Intercom Extension

Enter a unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

Alert-Info header

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

Use a value of Intercom or SVAutoPickup. If you use Intercom, a ringtone will play before the phone auto-answers 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).

Choose Sound to Play***Play beep/sound first***

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.

Folder

The Folder (within Sound Manager) that contains the sound that you want to play.

Sound to Play

The sound that you want to play before you start talking. The Description gives you the script of the sound.

Intercom Users and Receivers

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.

Authorized Extensions

Extensions or Extension Groups that can dial this extension.

Recipients

Extensions or Extension Groups that are intercom'd or paged when this extension is dialed.

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 includes the Default Extension Group (which includes 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 [“Settings” page 30](#).

0 Forwards to Extension

The extension the caller is forwarded when he or she presses 0 from the Directory. If this is blank and the caller presses 0, Switchvox leaves the caller within the Directory.

Search the Directory by members

Select from First Name or Last Name.

Directory Members

Choose the type of members that you want to add, and then add items to the collection.

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.

Parking Lot Start and End

The range of 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 [“Settings” page 30](#).

Seconds Before Calls are Returned

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.

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 [“Settings” page 30](#).

Do not require password as long as caller is calling from their own extension

YES indicates that this is 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 [“Settings” page 30](#).

Authorized Extensions

Choose a type of extension to add, and then put groups or extensions into the collection.

Extensions to Pickup

Choose a type of extension to add, and then put groups or extensions into the collection.

Dial Tone Extensions

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

Dial Tone Extension

A unique number for this extension. For information about the extension-number length, see [“Settings” page 30](#).

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.

Choose a type of extension to add, and then put groups or extensions into the collections.

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.

Choose a type of extension to add, and then put groups or extensions into the collections.

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.

Choose a type of extension to add, and then put groups or extensions into the collections.

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:

*33200*111*918885551212. If the password is not required, just leave it out:

*33200*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.

***Bulk-Modify
Extensions***

The Bulk-Modify Extension tool lets you search for and select a group of extensions, then modify specific settings at the same time.

Type some search criteria. The system displays extensions meeting your criteria. You may select individually, or you may search for a group that will be displayed in a Select Extension box.

1. Select the extensions you want to modify, and they are added to the Extensions collection box.
2. Select the type of setting that you want to modify, then select a setting within that type.

3. Specify the value.
4. Click Add or Cancel.
5. Review the extensions and the changes that will be made to those extensions.
6. Click Save Modifications.

Import Extensions

The Import Extensions tool lets you create and update multiple SIP or Virtual extensions from data in an imported CSV file.

The Import Extensions File

The 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, use just the comma with no spaces or text. If you want to make it blank, put a space in that column.

When your Extension File is ready, go to **Setup > Extensions > Manage** and click **Import Extensions** (see [“Import Options” page 27](#)).

File Information

This table describes the data for each type of import:

	ext	fname	lname	email	title	location	password	phone_ password	mac_ address
SIP: New	X	X	X	X	A	A	X	X	A
SIP: Update	X	A	A	A	A	A	A	A	A
Virtual: New	X	X	X	X	A	A	X	NA	NA
Virtual: Update	X	A	A	A	A	A	A	NA	NA

X = Required ::A = Accepted :: NA = Not Used

File Examples

SIP Extension File

```
ext,fname,lname,email,password,phone_password,mac_address
100,Joe,Smith,jsmith@home.com,100,s3CurEp@S$,0019159C0646
101,Jane,Doe,jdoe@home.com,101,c@nT6ueSS,0013174C5532
102,Dan,Jackson,djackson@home.com,102,g00DP@$$,
```

Virtual Extension File with location and other information

```
fname,lname,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
```

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
```

Import Options

The Import Extensions options determine how to manage this import.

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

Extension File. Browse your machine to find your Extension File.

For details about creating this file, see [“Import Extensions” page 26](#).

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.

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, 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 your extension numbers are the same length as specified in **Setup > Extensions > Settings**.

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

Groups

Extension Group tools enable you to group extensions together so you can apply changes and features to many extensions at a time.

The All Extensions group includes every extension in Switchvox. If you add an extension, it is automatically included as a Group Member. This group cannot be modified.

Manage Extension Groups

Create Extension Group adds a new extension group.

The **Modify** icon for the group lets you select the group you want to modify and display the Modify Group page.

The **Delete** icon for the group selects the group you want to delete and displays the Delete Group page. Click Yes, Delete Extension Group to remove the group.

Mouse-over the Note to see the Note that was added for a Group.

Extension Group Settings

Name

A unique name for this group.

Note

A descriptive note for this item. This is displayed when you hover over the note icon.

Voicemail/Fax Mailbox Quota (MB)

A number of megabytes as 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.

- Leaving blank, indicates unlimited space.
- Entering a 0 (zero), indicates 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.

Manage Group Members

This collector box lets you find and collect extensions and other Extension Groups that belong in this Extension Group.

To put an extension or Extension Group into the Group Members collection:

- Type a name or extension number into the text field. As you type, Switchvox offers suggestions of matching extensions.
- Click the Find icon to display the extensions available in Switchvox. You can sort this view by the column headers. If you are working with phone-type extensions, you can expand the window to show an extension's profile.

To delete an item from the collection, select it and click the **Delete** icon.

To put items in the right order, select the items you want to move, then click the Up, Top, Down, or Bottom button.

To select multiple items in the collection:

- Control-click to select two or more items that aren't together in order.
- Click an item, and then Shift-click another item to select both items and all items between them.

Templates

Extension Templates contain almost all of the same settings as the Phone and Virtual Extensions, except for the personal information of the extension-owner.

Templates even contain the membership in Extension Groups, so using well-defined Templates and Groups can make it simple to set up new employees.

If most of your phone-type extensions will be set up the same way, you can modify the default template to suit your needs. If not, you can create multiple templates for different types of employees.

Each time you create a SIP, Virtual, or Analog extension, you choose an Extension Template. Then, when the Create Extension page opens, all of the settings match that Template, and you don't have to do all of that work again.

Extension settings can always be changed. If you have a special case, you can create the extension with a template, and then modify the extension to change one or two settings.

Permissions

The Extension Permissions page displays the existing extension permission mappings in a table that includes extension holders, their respective target extensions, and the associated permissions. You can modify the information or delete the mapping entirely.

If you need to create a permission mapping, click the **Create Extension Permissions Mapping** button. This displays the Create Extension Permissions Mapping page. On this page, you choose the type (extension group or individual extension) who will be the Permission Holder and the type (extension group or individual extension) who

will be the Permission Target. Then you specify the permissions. The following are options:

- **Can View Calls.** Let the Holders view the caller ID of the target's current calls.
- **Can Record Calls.** Let the Holders record the target's current calls.
- **Can Monitor Calls.** Let the Holders silently listen in on the target's current call. This includes monitor, whisper, or barge.
- **Can Pickup Ringing Calls.** Let the Holders answer a call that is ringing the target's extension.
- **Can Intercom.** Let the Holders initiate an intercom connection with the target extension.

When finished, click **Save Extension Permission Mapping**. The Extension Permission page is re-displayed showing the new mapping.

Settings

The Extension Settings page allows you to set the default extension settings by specifying the following:

- **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.
NOTE: If you specify an extension length that is longer than some existing extensions, there may 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.** YES indicates 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.
- **Require Strong Extension Passwords.** YES indicates you require extension-owners to use strong extension passwords, and the extension-owner will not be able to change his or her password unless it is evaluated as strong. A strong password has at least 6 numbers, no more than 2 of the same number in a row (e.g., 111), and no sequences (e.g., 123). Also, it does not contain the extension.

For details about the extension setting that this affects, see [“Profile Information” page 8](#)

- **Require all extension owners to change their password if it is not already strong.** YES indicates that if strong passwords are required, then you also require that all extension-owners change their password if it is not already strong. This means extension-owners cannot use the Web Suite until they have created a strong password. This is similar to Login Actions, but this setting applies to all extension-owners who do not have a strong, or very strong password.
- **Require Strong Phone Registration Passwords.** YES indicates you require phones register with Switchvox using a strong password. A strong password has at least six characters, one of each character type (uppercase letter, lowercase letter,

number, and symbol), no sequences (abc, 123), no more than 2 of the same character type in a row, and includes 2 symbols. For example: m!2T^1

If you do not want to use symbols, you can just make the password longer. For details about the extension setting that this affects, see [“Phone Password” page 13](#).

NOTE: If you are using Digium Phones, or you used Phone Feature Packs to configure a phone, then the phone is already configured with a very strong password.

Click **Save Extension Settings** when finished.

Phones

The Phones tool allows you to select either Digium Phones or Phone Feature Packs to make phone assignments.

Digium Phones **34**
Phone Feature Packs **35**

Digium Phones

When you select this option, it displays the Digium Phone Assignment page, which lists the unassigned extension by default. Phone assignments define the Switchvox SIP-Phone extension assigned to a Digium Phone's MAC address. An assignment must define Line Key 1 as a Switchvox SIP-Phone extension, but an assignment can also define lines 2-6. Those lines can be another SIP phone extension on Switchvox or an external extension.

You can filter the phone assignments for viewing from among the following types:

- **Unassigned Phones.** Digium Phones on the network that do not already have a Switchvox extension assigned to them.
- **Assigned Phones and Extensions.** Displays the following:
 - MAC Address* for the Digium Phone
 - Model* number for the Digium Phone
 - Configured By* indicates if the assignment was created by the Server (the Switchvox Web Suite, Bulk Import, or the Extend API), or by the Phone itself (using the numeric keypad to select the extension for Line Key 1). Server-based assignments cannot be changed from the phone itself.
 - User Details* indicate the extensions assigned to the phone. Line 1 must be a Switchvox phone extension, but lines 2-6 (depending on the model of your Digium Phone) can be a Switchvox extension, or an external extension.
- **Ignored Phones and Extensions.** Digium Phones or Switchvox extensions that you do not want to assign and you do not want to see in the Unassigned lists. If

you want to delete a Digium Phone entirely from Phone Assignments, first ignore it, then you can delete it.

Quick-Create Assignments. The Quick-Create Assignments button lets you assign multiple phones and extensions quickly. Using this option, the extension is always assigned to Line 1.

This is most useful if you have a barcode scanner, so that you can scan each phone box, and easily associate it with the correct extension.

The Populate Unassigned Extensions button allows you to add all unassigned phone extensions as rows in order by extension number, first name, or last name.

Phone Assignment Options. These options can be used for creating phone-based assignments using the phone's numeric keypad to select an extension from a list. You don't need to use these options if you create server-based assignments (meaning, assignments created from the Switchvox Admin Suite, Bulk Import, or the Extend API).

Server Name. This is the name of the Switchvox server as it appears in the phone's list of Digium Configuration Servers.

Assignment Authentication. To ensure that your phones are assigned correctly, you can choose a method of authentication to be used before the phone will be assigned:

- **Assignment Code:** One code displays the extension list and lets any extension be assigned.
- **Extension / Password:** The extension list is always displayed, but once an extension is chosen, that extension's password must be entered.

To find an extension in the list, enter the first name, last name, extension number, or first and last initial.

Enable Open Enrollment Period. This option sets a period of time where phone-based assignments can be created without any authorization. Use this option with caution.

Phone Feature Packs

The following must be in place before you use Phone Feature Packs to set up your phones:

- Switchvox is up and running.
- Your phones are on the same network as Switchvox.
- You have a Phone Feature Pack 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 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 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" page 34*. If you have a Polycom 300 or 500 phone, you must set up your phone manually (you cannot use the Phones tool). For help, see the documentation for your phone.

The Phones 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:

- Place each phone where it should be located, plug it into your network, and turn it on. 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.
- In the Admin Suite, select **Setup > Extensions > Phones**. 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.

- 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.

- 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:

- Click the **Modify** icon
- Enter an extension number, or search for the extension that you want.
- After each phone has the correct extension, check its box and click **Confirm Checked Phones**.

Switchvox pushes out the final configuration to each phone, and sends a command to each phone to reboot again.

After 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 Extension-Owner User Guide, [“Phone Features” page 12](#)).

To modify a configured phone, click its **Modify** icon.

Main Line. Set an extension and an alternate host for this phone.

Additional Lines. To add a line, enable it, then specify whether this is a Switchvox Extension or not, and then enter the extension information.

- 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.

Polycom phones

- Reset the phone back to its factory defaults:
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 Enter. Once the phone reboots, press Setup during the countdown screen. Then press 4, 5, 6, on the keypad, then select Ok. Go into Server Menu to find the settings.
The phone reboots with the factory settings.
- Go to the Network Configuration for your phone: press Menu, 3,2, 456 (the password), Enter,1, 1. Scroll to Server and select it.
- Set server type to http and server address to IP_ADDRESS_OF_YOUR_PBX/pc. For example, if your PBX has an external IP address of 192.100.0.1 then set the server address to 192.100.0.1/pc.

snom Phones

Reset the phone back to its factory defaults.

Set the Setting URL to 'http://SWITCHVOX_IP_ADDRESS/pc/{mac}.xml'.

For example, if Switchvox's IP address is 192.100.0.1, the Setting URL is http://192.100.0.1/pc/{mac}.xml.

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.

Phone Setup Options

Alternate Host Default Option

YES indicates that the phone use the default for the Alternate Host option. The default is the PBX's external IP/hostname. The default is only applied if a phone does not have a setting in the Alternate Host option.

Only Accept Calls from the PBX

YES indicates that 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.

Call Routing

Tools are available for managing the following call routing functions:

<i>Channel Groups</i>	39
<i>VoIP Providers</i>	47
<i>Outgoing Calls</i>	56
<i>Incoming Calls</i>	58

Channel Groups

Channel Groups displays all the channels available and lets you group channels together by signaling type. You can also modify or delete existing groups.

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

- Enter a name for the channel group.
- Specify the device type.
- Select the channels to add to the group.
- 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.

NOTE: Unless you are an advanced user, do not change any of the other settings.

When you are done, click **Save 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. The following options are available:

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 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

This is 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

This is 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 long Switchvox 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 Buffer Policy

Set the Fax Buffer Policy for this Channel Group. To turn this option off, select Disabled. The default is Half.

This option lets you set a buffer policy to prevent fax failures on high latency systems. See the related option [“Number of Fax Buffers” page 42](#).

You must install a fax license and the fax software to make faxing available. For details, see [“Digium Addon Products” page 80](#).

Fax Detection

Set the Fax Detection for this Channel Group. For more information, see [About Faxing](#).

Both. 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” page 80](#).

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.

Number of Fax Buffers

Set the number of fax buffers. The default is 12.

This option lets you set a buffer policy to prevent fax failures on high latency systems. See the related option [“Fax Buffer Policy” page 40](#).

You must install a fax license and the fax software to make faxing available. For details, see [“Digium Addon Products” page 80](#).

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 preferable 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.

Wait Duration

This forces the system to wait the specified number of seconds before answering the call, allowing time to receive all of the call data before answering the call. The default is 1. Most likely you do not need to change the default.

VoIP Providers

This tool 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.

From the VoIP Provider page, you can create and list the 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.

SIP Providers

Click the Create SIP Provider button to displays the create page with the Provider Information tab opened. The other tabs are for advanced settings that you mostly likely do not need to change from the defaults.

After you have created a provider, you are prompted to go to the Connection 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. Or if the status is Unreachable, you have the option of running diagnostic tests on the provider information.

The following are the Provider Information fields:

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” page 80](#)*.

DTMF Mode. DTMF tones are the sounds emitted when you press buttons on your phone. Options in the dropdown include: Info, Inband, Auto, and RFC2833.

- 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).

Peer Settings Tab

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 the Same PBX. 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 uses the default setting, which 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.
- No 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 outgoing calls through this Switchvox.

- To allow calls, put an Outgoing Call Rule into the collection as Allow.
- To deny calls, put an Outgoing Call Rule into the collection as Deny.
- To ignore an Outgoing Call Rule for this provider, do not put the Rule into the collection.

Caller ID Settings Tab

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 Tab

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: Leave this option set to Yes unless instructed otherwise. This option lets the provider send you calls without authenticating. Most SIP providers do not authenticate, therefore setting this to No may cause problems.

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 most 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 Tab

The Call settings determine how to manage calls with this provider.

Audio Codecs. Some providers support more than the normal audio 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.

The following table describes the audio codecs. If you entered a license key for the G729 Codec in **Tools > Digium Addon Products**, you will also see G729 in the list of 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
G729	30	Good	Medium
SPEEX	33	Great	High
GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

Video Codecs. The video codecs H.263, H.263+, and H.264 are available.

NOTE: Video calling is not available over IAX, so these codec settings are not available in the IAX VOIP Provider section.

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.

Call Settings.

- **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 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 is 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 Tab

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” page 80](#).

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.
- **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.

Fax Detection

This option describes the method of fax detection. The default is Yes.

Yes. Detect incoming faxes based on the receipt of a CNG tone or a T.38 invite.

CNG. Detect incoming faxes based on the receipt of a CNG tone.

T.38. Detect incoming faxes based on the receipt of a T.38 invite, without CNG tone.

No. 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” page 80](#).

Error Correction Mode for T.38 Faxes

Error Correction Mode (ECM). The default is FEC.

Initiate T38 reinvite after timeout

This option initiates a T.38 re-invite if it is not received within 10 seconds. The default is Yes.

Enable Audio Failover

This option causes Switchvox to revert to audio fax mode in the event that T.38 negotiation fails. The default is Yes.

IAX Providers

Click the Create IAX Provider button to displays the create page with the Provider Information tab opened. The other tabs are for advanced settings that you mostly likely do not need to change from the defaults.

After you have created a provider, you are prompted to go to the Connection 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. Or if the status is Unreachable, you have the option of running diagnostic tests on the provider information.

The following are the Provider Information fields:

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.).

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.

Peer Settings Tab

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 PBX. 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” page 53* for more information about each outgoing rule.

Caller ID Settings

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 Tab

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 condi-

tion 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 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 Tab

The Call settings determine how to manage calls with this provider.

Audio Codecs. Some providers support more than the normal audio 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.

The following table describes the audio codecs. If you entered a license key for the G729 Codec in **Tools > Digium Addon Products**, you will also see G729 in the list of 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
G729	30	Good	Medium
SPEEX	33	Great	High
GSM	35	Good	Low
ADPCM	54	Good	Low
LPC10	22	Poor	Medium

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.

Fax Settings Tab

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” page 80](#).

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.

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.

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. It is comprised of three tabs: Outgoing Call Rules; Caller ID Rules; and Call Diagnostics. All rules are displayed in a table on the Outgoing Call Rules page. Clicking the Create Outgoing Call Rule button displays the Outgoing Call Rule page.

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 route (such as a VoIP 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. You can modify or delete those Rules, and create your own.

Rule Name. A unique name for this rule.

Rule Note. A descriptive note for this item. This is displayed when you hover over the note icon.

Pattern to match. Enter the call rule information in the boxes as follows:

- Number begins with the digits [].
- 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. This indicates which of your Channel Groups or Providers to use for this call, or it's an internal call (within Switchvox).

Failover Call Through lets you set up to 5 more providers if Switchvox cannot place the call through the previous provider. This helps you protect against downtime if one of your providers is experiencing technical difficulties.

Switchvox tries to complete the call in the order that the Failover Providers are shown.

Rule Final. 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.

Outgoing Caller ID Rules Tab

Outgoing Caller ID rules let you modify the outgoing caller ID name and number.

Caller ID rules can apply to:

- A single extension.
- Any extension. (Use a Single Rule, and put the word any in the Select Extension field.)
- A range of extensions.
- An extension group.

If you use a ranged Caller ID rule, you can derive an extension's correct caller ID by adding and prepending numbers to the 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 Tab

Outgoing Call Diagnostics can be used to determine which Outgoing Call Rule is being used and why.

If you set Enable Diagnostics to YES, diagnostic information is read over the phone to all callers from all extensions.

For example, If you were to Enable Diagnostics 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" page 53](#)).

Incoming Calls

The Incoming Calls tools let you manage incoming call rules and routes via two tabs: DID Routes, the default; and Caller ID Rules. A DID Route maps numbers coming in

on a particular Provider or Channel Group to a particular extension., and Incoming Call ID Rules let you perform actions on calls based on who is calling you. These rules are system wide and happen before any of the extensions' call rules are triggered.

DID Routes Tab

By default, Switchvox creates a DID 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 routing based on information sent by the provider or deduced from the channel group.

To create a new DID route, click Create Single DID Route or Create Ranged DID Route. A Single DID Route always routes calls to a single, specific extension. A Ranged DID Route derives the correct extension number from the DID.

You can put all of the DID Routes in order, to ensure that call reach the correct extension.

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.

Single DID Route Settings

This option 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

To create a single DID route, enter/specify the following information:

- **Rule Name** to identify the rule.
- **Note.** Clarifying information regarding the rule.
- **Incoming DID to Match.**
- **Incoming Provider.** Select provider from the list of existing Providers.
- **Incoming Call Type.** Select from dropdown options: Voice Calls, Faxes, or both.
- **Extension to Route Call.** Search for or enter the extension to route.
- **Process All Calls as a Fax.** If you have fax capabilities, and you have selected both types of calls, you call indicate YES; otherwise, this option is not available.

NOTE: You must install a fax license and the fax software to make faxing available. For details, see [“Digium Addon Products” page 80](#).

When finished, click **Save Single DID Route**.

Ranged DID Route Settings

This option 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.

Enter the following information:

- **Rule Name** to identify the rule.

- **Note.** Clarifying information regarding the rule.
- **Beginning Range of Incoming Call.** First number in the range of numbers.
- **End Range of Incoming Call.** Last number in the range of numbers.
- **Incoming Call Type.** Type of calls to affect: Voice Calls, Faxes, or Voice Calls and Faxes.
- **Incoming Provider.** Select provider from the list of existing Providers.
- **Number of digits to trim.** Enter the number of digits to trim from the front of the phone number.
- **Derive Resulting Extension by adding this to the trimmed call.** Enter a value to add to the result or you can leave this blank.

Incoming Caller ID Rules Tab

To create a new Incoming Caller ID Rule, click either Create Call Transfer Rule or Create Block Number Rule.

Create Transfer Call Rule

This option lets you create a rule that transfers 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, enter/select the following information:

- **Rule Name** to identify the rule.
- **Note.** Clarifying information regarding the rule.
- **Incoming Number Type** from the dropdown. Options are Phone Number or Prefix.
- **Incoming Number.** Enter the phone number or prefix that you want to affect.
- **Extension to Transfer to.** Enter the extension the call should be transferred to.
- **Timeframe.** Select Any Time, During Business Hours, Outside Business Hours, or Weekends Only from the dropdown. (For more information, see [“Time Frames” page 76.](#))

When finished, click Save Call Transfer Rule and a message is displayed indicating the new rule was successfully saved.

Create Block Number Rule

This option 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 any-time this number calls you. You can block whole prefixes of numbers by choosing to block Prefix rather than Phone Number.

Enter/select the following:

- **Rule Name** to identify the rule.
- **Note.** Clarifying information regarding the rule.

- **Incoming Number Type** from the dropdown. Options are Phone Number or Prefix.
- **Incoming Number**. Enter the phone number or prefix that you want to affect.
- **Block Action**. Enter the blocking actions from the dropdown options: Play Busy Signal, Play Congestions, or Hang Up.
- **Timeframe**. Select Any Time, During Business Hours, Outside Business Hours, or Weekends Only from the dropdown. (For more information, see [“Time Frames” page 76.](#))

When finished, click Save Block Number Rule and a message is displayed indicating the new rule was successfully saved.

Modifying Caller ID 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 Caller ID 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

Admins

The Admins tools give you a means of managing Switchvox accounts:

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Manage

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 Admin to display the Create Admin page.

General Information Tab

Enter a new Login name for the admin, then enter and re-enter a new password. The password is case-sensitive, and it can contain special characters. The strength of the 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 to use the password that you created. If the sub-admin creates a weak password, Switchvox shows you with an alert.

Permissions Tab

You can set this sub-admin's access for each of the main sections in Switchvox. You can block the sub-admin 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 Save Admin, and the Manage Admins page is displayed with the new sub-admin included in the Admins list.

To modify a sub-admin, click its Modify icon.

To delete a sub-admin, click its Delete icon.

My Account

This is the information for this administration account.

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

If you choose to use a language other than English for US, be aware that all other languages and locales handle dates with the format of dd/mm/yyyy (as opposed to mm/dd/yyyy).

New Password

The password for this account. Leave blank if there are no changes.

Administration accounts can have extensive control over Switchvox, so you should use a very strong password to protect this account. Strong passwords are fairly long, combining letters, numbers, and special characters.

Retype New Password

For verification, enter the password again.

Minimalist Mode

This mode presents the Web Suite interface with reduced graphics.

If Minimalist Mode is set to Yes:

- The background graphic and larger graphics on buttons and tabs are removed.
- Some text, especially in section headers, is smaller.
- Animations are removed.

Tools

This section describes the functions under the Tools category, which are used to set up and manage the PBX tools such as Time Frames, IVR Menus, Voicemail, and Music on Hold.

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PBX Features

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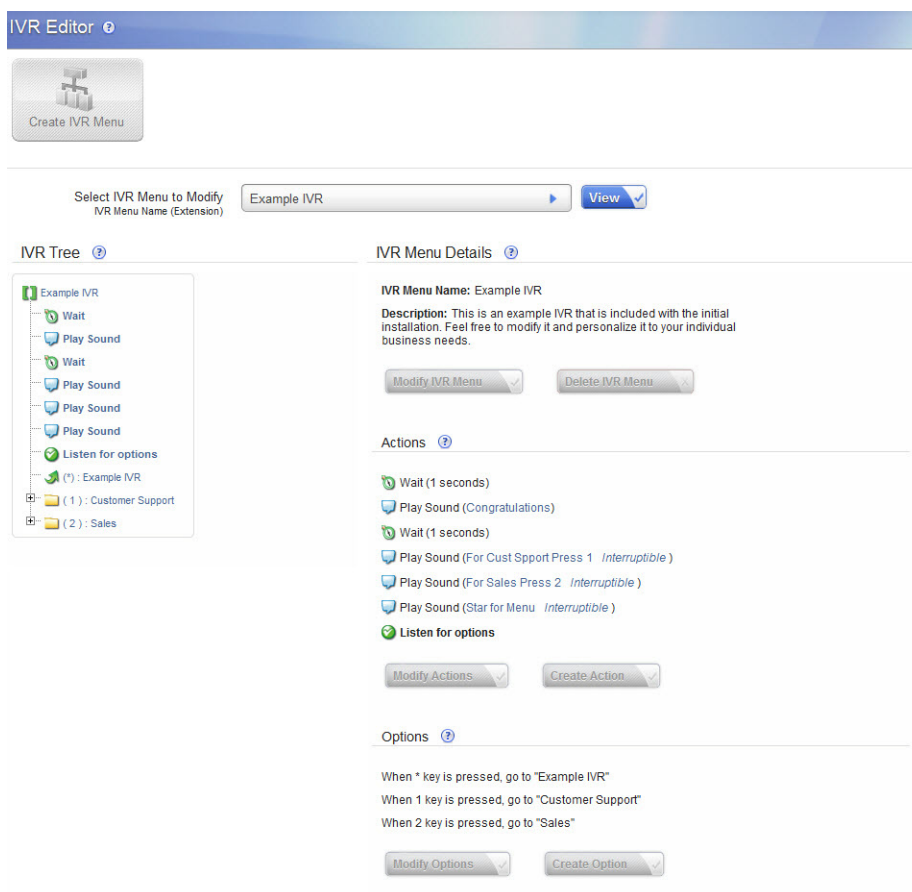
Interactive Voice Response (IVR) Editor

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. **IVR Editor** is the entry to this process.

Managing IVR Menus

Select the **IVR Editor** from the PBX Features section. The IVR Editor page is comprised of a Create IVR Menu button and the following sections for any default or already created IVR menus: IVR Tree, IVR Menu Details, Actions, and Options, and a dropdown to select an IVR to view or modify

See following figure..



Planning a New IVR Experience

Before creating an IVR Menu, you should plan in advance the entire IVR experience you want to offer your callers. Then create the sequence starting from the end and working towards the beginning as is shown in the following example:

- Determine all of the recorded sounds you want to play for the caller (instructions, information, etc.) and create those sounds in the Sound Manager (see page 93).
- 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, see page 69.)
- Determine the factors that 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 need to create an IVR menu for each situation.
- After you have your sounds and a set of component IVRs built, then you can create the Main IVR Menu that answers your calls.

Create IVR Menu

Click **Create IVR Menu** to display its menu page.

- Enter a **unique name** and a **description** for the new IVR.
- Click **Create IVR Menu**.

This displays the **IVR Editor** page with a message indicating the new IVR Menu was created. The new menu is displayed in the IVR Tree and IVR Menu Details.

After you are done building your main IVR, you need to create an extension that reaches it. For details, see IVR 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.

Modifying an IVR Menu

To modify an existing IVR Menu, select it from the **Modify Existing IVR Menu** dropdown. The dropdown presents your IVRs in two sections: IVRs that have an extension number and IVRs that do not have an extension number. After selecting a menu, the IVR Tree and Menu Details, including Actions and Options, are updated with information for the selected IVR Menu.

IVR Tree

The IVR Tree displays an overview of the selected menu. **IVR Actions** are listed first with various icons to indicate the kind of action. The last action in the list is always **Listen for options**, which cannot be modified or deleted. This is the action that handles the IVR Options.

IVR Options are listed inside folder icons. Options are an entry point into another IVR. When you click the plus next to an Option folder, it expands and displays the actions in that IVR menu.

Click on any action, folder, or option name to see details in the **IVR Menu Details**.

IVR Menu Details

IVR Menu Details 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, it displays the details for that action.

IVR Actions

You can add, modify, or delete actions associated with an IVR Menu. Most actions have a configuration page on which you can specify the details of the action, then save your changes.

Adding an Action to an IVR Menu

1. Click **Create Action** from the IVR Menu Details page, or the Modify IVR Actions page. This displays the Create Action page.
2. Select an action from the dropdown. After you make a selection, the Action Description displays a short description of the selected action.
3. Click **Next**.
4. Complete the detail of the action as appropriate, then click **Save Settings**.

Modifying an existing action

1. Click **Modify Actions** in the IVR Menu Details. This displays the Modify IVR Actions page.
2. Click the **Modify** button for the action you want to modify. This displays the Modify Action page.
3. Make your changes and click **Save Settings**.

Deleting an existing action

1. Click **Modify Actions** in the IVR Menu Details. This displays the Modify IVR Actions page.
2. Click the **Delete** button for the action that you want to remove. This displays the Delete Action page. Click **Yes, Delete Action**.

Actions Types

The following are Action Types available for IVR Menus.

- Interruptible Actions
- Play Sound
- Record Sound
- Play Recorded Sound
- Send an Email
- Send Recorded Sound to Voicemail
- Store Recorded Sound in Sound Manager
- Upload Recorded Sound
- Merge Sounds
- Play Sound From URL
- Gate Keeper
- Conditional Clause
- Time Based Clause
- Change Language
- Alter Caller ID
- Wait
- Give Busy Signal
- Hang Up
- Dial Extension
- Send to Voicemail
- Send to External Number
- Go To IVR Menu / Action
- Record Digits
- Say Digits/Letters
- Say A Number
- Say Date/Time
- Play DTMF Tones
- Perform Math

- Set Variable
- Set Global Variable
- Get Global Variable
- Concatenate Variables
- Send Call Values to URL
- Get Extension Status
- Get Extension Type
- Check User Password
- Set Distinctive Ring Hint
- Remove Distinctive Ring Hint

Interruptible Actions

Some IVR actions can be interrupted if the caller presses a key on his or her keypad. Although this can offer a better caller experience, you must plan for it carefully because it has repercussions throughout your IVRs. Take care about the order of actions in your IVR and where interruptions are allowed.

Here are the rules 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
 2. Digits action. They will not be used to dial an extension or to select an option.
 3. 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 IVR Options).
 4. 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.

This sound is interruptible. Check this box if this is interruptible. For details, see “Interruptible Actions” (page 64).

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" (page 76) 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:

```
<?xml version="1.0" encoding="ISO-8859-1"?>
<!DOCTYPE ivr_info SYSTEM "http://www.mybz.com/xml/ivr.dtd">
<response>
<result>
<ivr_info>
<variables>
<variable>
<name>balance</name>
<value>12323</value>
</variable>
<variable>
<name>first_name</name>
<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:

- IDLE
- INUSE
- BUSY
- 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
- ivr
- sip
- conference
- group_pickup
- intercom
- voicemail
- callpark
- agentcallbacklogoff
- call_queue
- agentcallbacklogin
- autodial
- directory
- 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.
- 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.

IVR Options

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 during this IVR. For example, the IVR might play a sound "if you know your party's extension, please dial it now." This setting enables that feature.

To specify which extensions can be dialed from this IVR, enter an Extension Group that includes the extensions that you want to allow.

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.

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).

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 modify or delete these time frames, or create new ones.

IMPORTANT: A time frame is considered valid when the current date and time match any one of a time frame’s conditions. Not all of the conditions in the time frame have to match the current date and time.

Creating a New Time Frame

Time frames consist of Time Frame Conditions. The time frame itself is considered valid when any of its conditions are met.

To modify a Time Frame with Conditions:

- Click Create Time Frame Conditions.
- Enter a Date Range, Day of the Week Range, or a Time Range, or any combination.
- Continue adding conditions 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 12:00am to 11:59pm, and Wednesdays 12:00am to 11:59pm

:

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

Early Morning

Start Date	End Date	Start Day	End Day	Start Time	End Time
07/01/2011	07/07/2011				

The First Week in July 2011

Start Date	End Date	Start Day	End Day	Start Time	End Time
01/21/2011					

After January 21, 2011

Start Date	End Date	Start Day	End Day	Start Time	End Time
02/01/2011	02/28/2011	Monday	Friday	6:00AM	7:00AM

Weekdays in February, 2011 between 6:00 and 7:00 in the morning

Modifying a Time Frame

- Click its Modify button. This displays the Modify Time Frame Conditions page.
- Click a Delete button to remove a condition that no longer applies. (You cannot edit a condition.)
- Click Create Time Condition to add a condition.

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 displayed 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 so that Switchvox automatically records 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 or her 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.

Scheduling a 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 [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.

Viewing & Playing Recorded Calls

The Recorded Calls list shows the information for each recorded 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.

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>
```


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.

Digium Addon Products

This lets you register and enable Digium Addon products.

To add a product,

- Select the product from the dropdown.
- Enter your License Key.
- Click Submit.

NOTE: If you install a Fax license, you will be prompted to download and install the related fax software.

Finding Product Addons

For information about what Addons are available for your Switchvox, go to Switchvox Product Addons

On the Switchvox Product Addons website, you must enter your Switchvox Registration Code to find your Addons. You can find your Registration Code on your Switchvox DVD.

Your Addons could include your Switchvox Fax License, other freely available Addons, and other available Addons that are specific to your Switchvox.

Event Triggers

Event Triggers let you set up Switchvox to interact with external web-based applications. When certain events occur in Switchvox, it can do an HTTP POST that includes information about the event, and then respond to 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 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. New Voicemail

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.

QUEUES. 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:

```
<response>
<result>
<call_info>
<caller_id_number>8585551212</caller_id_number>
<caller_id_name>John Doe</caller_id_name>
<transfer_extension>850</transfer_extension>
<display_url>http://intranet.example.com/crm.cgi?custid=2</display_url>
<continue_outgoing_reporting>1</continue_outgoing_reporting>
</call_info>
</result>
</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. If this XML is a response to the event On Outgoing Call, this can be an external number. Make sure to enter this number as you would dial it from your phone. For example, if you must dial 9 for an external number, an external number here must begin with a 9. **WARNING:** It

is possible to break Switchvox by creating routing loops with `transfer_extension`.

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.

Voicemail

Voicemail Routing

Voicemail Routing lets you handle several voicemail-related actions.

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.

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 Import 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” (page 14).

NOTE: 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.

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SMTP Settings

When voicemail is received for a phone-type extension, a notification email message is sent to the extension's email address.

SMTP Settings let you set a 'From' address for those email messages, and the SMTP server.

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 bounce email that comes from the localhost.localdomain domain.

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).

Templates

Notification Templates let you customize voicemail notifications. You can create multiple templates and tailor them to different types of accounts. These templates are

available to an extension owner when he or she sets up voicemail notifications (along with any templates he or she creates).

You can use variables in your template that are substituted with real values when the notification is 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 Notification 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.

Modifying Notification 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 Notification 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.

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Contacts

Internal Directory

By default, the Internal Directory is available on all of your Digium phones, and it includes all of the Switchvox extensions. (The Internal Directory is also available on Polycom phones configured with a Phone Feature Pack, under the Switchvox Directory application.)

You can customize what extensions are in the Internal Directory by building one or more Extension Groups and adding them to the directory. If you do this, be sure to remove the existing group, **All Extensions**.

Download the Internal Directory

Click the floppy disk icon to download the XML for the Internal Directory. This is useful for a peered Switchvox to upload as an External Contacts file.

External Contacts for Digium Phones

To add external Contacts (Contacts that aren't extensions on this Switchvox) to all Digium phones, you can upload a CSV or XML file.

To see an example of the XML used by Contacts, download the Internal Directory's XML file.

CSV headers:

```
first_name, last_name, organization, job_title, location,
email_address, primary_number, primary_number_label,
secondary_number, secondary_number_label, jabber_id,
subscription_url
```

Example of the CSV record for one of your vendors:

```
Jane,Doe,Bunches of Flowers,Designer,San
Diego,jane@bunchesofflowers.com,918585551212,Flower
Stand,918585553333,Mobile,,,
```

Distinctive Ringtones

Switchvox can store and play distinctive ringtones that can be used on Digium Phones (or a Polycom phone with a Phone Feature Pack). See Phone Features

There are limitations for the total number of ringtones, and the total amount of disk-space that can be used. But you should have plenty of room to add the ringtones that you like.

To create a ringtone:

- Click Create Ringtone.
- Click in the Choose a File text box, and browse your computer to find a .wav sound file.
- In the Name field, enter a name for this ringtone.
- Click Save Ringtone.

The 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 Digium Phone (or a Polycom phone). 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.

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

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.

Switchboard Panels

Switchboard Panels let you add powerful panels to your 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 Switchboard 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.

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

Custom Switchboard 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.

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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 MP3 and Ogg Vorbis codecs for MOH files. This means that any file you want to use for MOH needs to end in .mp3 or .ogg, or it won’t play properly.

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**.

Adding a Group

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

Enter/select the following:

Group Name. A unique name for this group.

Description. a unique name for this 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.

When finished, 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 includes the new group highlighted in blue.

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 you really want to delete the group, then click **Yes, Delete Group**. This redisplay the Music On Hold main page, and a message indicates that your group has been deleted.

Uploading a Song

Song to Upload

- Select **Upload Song** from the Add New dropdown, then click **Go**. This displays the Upload Song page.
- 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.

NOTE: The file must be .mp3 or .ogg.

- Select a group this song should belong to from the Add to Group dropdown.
- Click **Upload Song**. The Upload Song Main page is displayed again, with a success (or failure) message, and the new song is listed.

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 Music on Hold page is displayed, and a success message indicates that the files were deleted.

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 redisplay 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 redisplay 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.

To obtain sound packs for different languages, go to Switchvox Product Addons.

To add a sound pack:

- Download the sound pack to your desktop computer.
- Go to Tools > Media > Sound Manager.
- Select Sound Pack from the Add New: dropdown.
- Click Go.
- Click Browse and locate the file on your computer.
- Click Add Sound Pack.

The sound pack is added to the Sound Manager, and the language of the sound pack is available for use.

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 default language, click Set as Default for that language. 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.

Reporting

This section describes the Reporting category, which describes how to generate detailed usage reports about your calling patterns and queue activity.

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Call Reports

Concurrent Calls **98**
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Concurrent Calls

The Concurrent Calls Report shows the total calls happening at the same time, over a date and time range that you define.

Call Logs

Call Logs 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. Opens the report in the browser window.

Output to .xls file. Downloads a .xls file that you can open with an application such as Microsoft Excel.

Call Reports

Call Reports lets you generate a report about call activity. Call reports are generated for the date-range and criteria you specify, and they include the information 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 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 the following output 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.

Queue Reports

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Queue Logs

Queue Logs return a list of queue calls based on the criteria that you specify.

Date Range. The log will include queue calls that happened during this date/time range.

Queue. The log will include calls made to this queue.

Type of Queue Call. The log will include only these types of queue calls.

Queue Reports

Call Queue Reports

Queue reports can be broken down 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 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.

Queue Status

Queue Status provides real-time information for each call queue in Switchvox:

- Overall statistics.
- Member status and statistics.
- Calls waiting in the queue.

The information in Queue Status is only for one day. For historical information, see Queue Logs or Queue Reports.

Statistics

Queue Statistics show you how a queue has been functioning for the following categories:

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. Longest time that a call was in the queue and was answered by a queue member.

Average 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 Talk Time. Average time that queue members spent talking with callers from the queue.

Member Status

Member Status 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

Current Calls Waiting 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.

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 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.

Server

This section describes the tools under the Server category, which describe how to perform maintenance tasks, manage your server's network settings, and diagnose issues.

Diagnostics	104
Maintenance	111
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Diagnostics

The Diagnostics tools give you views of the call activity in Switchvox and help you evaluate potential problems:

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<i>Current Calls</i>	106
<i>Server Information</i>	106
<i>Disk Usage</i>	107
<i>SNMP</i>	107
<i>Error Logs</i>	110
<i>Advanced Debugging</i>	110
<i>Technical Support</i>	110

Connection Status

This tool displays Connection Status information for the following entities connected to Switchvox:

- VOIP Providers
- SIP Phones
- Hardware Devices
- Fax Print Jobs

VOIP Providers

This tool lists the following information for each VoIP provider:

- **Type**
- **Name** of the provider as entered when the provider was created
- **Host** that Switchvox connects to when registering and sending calls

- **Account ID**
- **Callback Extension** that incoming calls are sent to by default
- **Latency (ms)**
- **State** of the connection. This value is *Registered* when the system is operating properly. If it is not, outgoing calls will fail and incoming calls cannot be received. Here are some 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, 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 tool lists all phone Extensions with their respective Caller ID, User Agent, IP Address, and Phone State.

Phone States. The following are possible phone states:

- *Registered.* This indicates the phone is configured properly and is on the network.
- *Timeout.* This indicates 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 the phone is not registered with Switchvox.

If a phone is not registered but it should be, check the following:

- The phone should be 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.
- The phone must know the Switchvox IP address, and it must have the same password as the "Phone Password" for the extension. For more information on configuring your phone, see your phone's manual or call your phone's technical support.
- The phone must be ON, its Ethernet cord plugged into the phone and the correct Ethernet jack, and it must be on the same LAN as Switchvox (not on a separate partitioned LAN subnet).
- There should be no firewall between your phone and Switchvox that could block your phone from accessing Switchvox.

NOTE: 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 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 tool 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 that the device is plugged into the correct line and that you have the appropriate signaling type selected.

Fax Print Jobs

This tool lists the current Print Jobs for faxes.

These Print Jobs occur when an extension-owner is creating a fax from his or her desktop computer.

Current Calls

This tool lists 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**, then click **Yes, Hangup**.

Server Information

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.

memory. If **memory** is over threshold, one option is to turn the system off (wait a few moments), 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.

Advanced Server Information. A Switchvox Technical Support representative may ask you check this area for a specific issue. In most cases, you will not need to use this tool.

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 General Settings (**Setup > Extensions > Manage > Phone and Virtual Extensions > Profile Information > Advanced Options > General Settings**).

SNMP

SNMP (Simple Network Management Protocol) is used to monitor devices on an IP network. If you are not already using SNMP, then you can 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

.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 call-back 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	first phone state
.1.3.6.1.4.1.22736.10.4.3.[2-xxx].[1-6]	details for each additional phone
.1.3.6.1.4.1.22736.10.4.2	number of reachable SIP phones

.5 Current Calls

.1.3.6.1.4.1.22736.10.5.1	total current calls
.1.3.6.1.4.1.22736.10.5.2	number of calls in state 'agent_login'
.1.3.6.1.4.1.22736.10.5.3	number of calls in state 'agent_logout'
.1.3.6.1.4.1.22736.10.5.4	number of calls in state 'checking_voicemail'
.1.3.6.1.4.1.22736.10.5.5	number of calls in state 'conference'
.1.3.6.1.4.1.22736.10.5.6	number of calls in state 'directory'
.1.3.6.1.4.1.22736.10.5.7	number of calls in state 'faxing'
.1.3.6.1.4.1.22736.10.5.8	number of calls in state 'intercom'
.1.3.6.1.4.1.22736.10.5.9	number of calls in state 'ivr'
.1.3.6.1.4.1.22736.10.5.10	number of calls in state 'leaving_voicemail'
.1.3.6.1.4.1.22736.10.5.11	number of calls in state 'monitoring'
.1.3.6.1.4.1.22736.10.5.12	number of calls in state 'parked'
.1.3.6.1.4.1.22736.10.5.13	number of calls in state 'queued'
.1.3.6.1.4.1.22736.10.5.14	number of calls in state 'receiving_fax'
.1.3.6.1.4.1.22736.10.5.15	number of calls in state 'ringing'
.1.3.6.1.4.1.22736.10.5.16	number of calls in state 'talking'
.1.3.6.1.4.1.22736.10.5.17	number of calls in state 'unknown'

.6 Telephony Cards

.1.3.6.1.4.1.22736.10.6.1	number of devices
.1.3.6.1.4.1.22736.10.6.2	number of state 'Ok' devices
.1.3.6.1.4.1.22736.10.6.3.1.1	first device id
.1.3.6.1.4.1.22736.10.6.3.1.2	first device name
.1.3.6.1.4.1.22736.10.6.3.1.3	first device channel
.1.3.6.1.4.1.22736.10.6.3.1.4	first device channel group
.1.3.6.1.4.1.22736.10.6.3.1.5	first device signalling
.1.3.6.1.4.1.22736.10.6.3.1.6	first device state
.1.3.6.1.4.1.22736.10.6.3.[2-xxx].[1-6]	details for each additional device

.7 Switchvox Subscription

.1.3.6.1.4.1.22736.10.7.1	expire date
.1.3.6.1.4.1.22736.10.7.2	days left

.8 RAID Controller

1.3.6.1.4.1.22736.10.8.1	controller status
1.3.6.1.4.1.22736.10.8.2	logical device status
1.3.6.1.4.1.22736.10.8.3	device #0 status
1.3.6.1.4.1.22736.10.8.4	device #1 status

Other OIDs

The Switchvox agent provides some additional OIDs that are not specific to the Switchvox application.

.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.

Error Logs

These tools are for our customer service representatives to help you with problems 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

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.

The information can be useful in identifying the source of a problem; however, these sessions place an additional load on Switchvox system resources.

Technical Support

These tools are for Switchvox Technical Support representatives to help you with problems that you might have with Switchvox.

IMPORTANT: You do not need to, and should not use these tools unless it is at the request of a Switchvox Technical Support representative.

Technical Support Access lets us access your machine to help diagnose a problem. You must enable Technical Support Access to use the VPN connection. If you disable it, then all access to your system will be closed, including an open VPN connection. A Technical Service Request (TSR) can be downloaded, then sent to Switchvox Technical Support.

Maintenance

The following system Maintenance tools are available:

Telephony Cards **111**
Backups **112**
Registration **113**
Updates **113**
System Clock **115**
System Reload **115**

Telephony Cards

The Telephony Cards tools let you scan for and configure new telephony hardware in the Switchvox server. You can also delete hardware. If hardware has not been detected, make sure all cards are installed properly and that the card you are using is supported. See the Digium web site for installation instructions.

Hardware Devices. Each device connected to the Switchvox server is listed under this section. To configure a listed device, click **Configure** for that device. To delete a listed device, click **Delete** for that device, then verify the deletion request. 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 in which your Switchvox server is located, select it from the dropdown, then click **Set Country**. This setting also sets the call-progress tone scheme that determines the tones sent to your callers.

Hardware Echo Cancellation. This option disables the hardware's echo cancellation so all echo cancellation is handled by Switchvox. For details about those settings, see "Advanced Options" page 38 under Channel Groups.

Echo Reduction. This tool allows Switchvox to run a series of tests, after which it calibrates the analog lines to the results. Click **Remove Echo Reduction Settings** to

revert the settings back to their default values. We recommend running Echo Reduction if you move lines to other ports, or add new lines.

NOTE: 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.

Backups

The Backups tool lets you create and download a backup file containing much of the data and audio files for Switchvox. This file may be used at any time to restore Switchvox to the same state it was in at the time the backup was created.

NOTE: Always backup your Switchbox system before making any changes to it, including restoring the system from a backup.

IMPORTANT: A best practice for backups is to schedule a daily automatic backup that stores the backup file on an FTP server. For details, see “Set Up Automatic Backups” page 112.

Create a Backup. This tool displays the Create Backups page. Select any optional components you want to include in the backup.

NOTE: Call Recordings are backed up separately. For details, see “Call Recording” page 78.

When ready to begin, click **Create Backup**.

WARNING: The backup process may take several minutes depending on the amount of content. Do not click any links or buttons during the process.

After the backup is finished, the Backups page is displayed with the new backup listed in the Backups table. Click **Download** to save the Backup (which may include multiple files). Store the Backup in a location from which it can be easily accessed if necessary to restore your system.

Set Up Automatic Backups. This tool displays the Automatic Backups page. Enter the following information:

- **Day and Time.** Set the day of the week and time to run the backup process.
- **Number of Copies.** Enter the number of backups to store. 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 you use to access the server, and the path to the directory to store the backup files. Click **Test FTP Settings** to test this information to be sure your backup files can be transferred.
- **Optional Components.** Select optional files and logs you want to include in the backup.

After you have entered the backup information, click **Save Settings**.

To disable automatic backups, clear the FTP information, then click **Save Settings**.

Upload and Restore Backups. Be absolutely sure you want to restore your system from any backup file before clicking a Restore option because restoring your system from a backup file causes the following to happen:

- Your current Switchvox configuration is overwritten with the contents of the selected backup file.
- All current calls are dropped.
- All agents will be logged out of queues.
- All data created since the backup was created will be lost, including extensions and voicemail.

- **Restoring From the List.**

If the backup file you want to use is listed in the table of Backups, you can restore from the list. Find the backup file in the list and click its **Restore** button. Verify you want to restore using this backup, by clicking **Yes, Restore Backup**.

- **Restoring From a Backup on Your Computer.**

If the backup file you want to use is not listed, then you need to upload the backup file from your hard drive, then restore. To use this process, do the following:

- Click **Upload a Backup File**.
- Browse to find the file on your computer. If the backup is contained on multiple files, select the meta file.
- If you select a meta file, you are prompted to Browse for all of the files.
- After selecting the files, click **Upload and Restore Backup**.

- **Deleting a Backup File from the List.**

- Click the **Delete** button for the backup.
- Verify you want to permanently delete the file by clicking **Yes, Delete Backup**. This deletes and removes the backup from the list, and the Backup page is displayed with a message indicating that the file has been deleted.

Registration

Registration is usually completed when Switchvox 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.

Updates

This tool displays the following information about Switchvox:

- Current Switchvox software version number
- Maximum number of extensions and concurrent calls
- The expiration date for Subscriptions
- Any Switchvox software updates that are available for you to 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 Features” page 12 in the Switchvox Extension-Owner User Guide.
- **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” page 55 in the Switchvox Extension-Owner User Guide.
- **Distinctive Ringtones.** Set the audio file of your choice to ring on incoming calls, depending on the caller. See “Ring Rules Tab” page 17 in the Switchvox Extension-Owner User Guide.
- **Auto-answer.** Set an automatic answer in Intercom mode, depending on your caller. See “Ring Rules Tab” page 17 in the Switchvox Extension-Owner User Guide.
- **Combine Ringtones and Auto-answer.** Combine a ringtone and auto-answer, depending on the caller. See “Ring Rules Tab” page 17 in the Switchvox Extension-Owner User Guide..
- **Phone Applications.** Access many of the Switchvox features right from your phone.

The Phone Features available on your phone vary, depending on the type of phone you have.

- General Options are available on snom and Polycom phones.
- 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.
- If you use Switchvox remotely (your phone is not on the same network as Switchvox), you will not see Switchvox Caller Profiles.

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 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 are one or more updates listed on this page, each one is listed with a more info... link that displays the release notes for that update. The release notes describe the new enhancements in the update, and any issues that are resolved in that update.

Each update has two actions: Download, and Download and Apply.

--**Download.** This option downloads the update to Switchvox with a progress indicator so that you can see how long it will take. After the download is complete, the update is

ready to be applied, and the button **Apply This Update** appears. You can apply the update whenever it is convenient. After you apply an update, the files for that update, and any previous updates that you may have downloaded, are removed from Switchvox.

--**Download and Apply.** If you click Download and Apply, the update is downloaded to Switchvox and immediately applied.

System Clock

System Clock 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.

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, first make a system backup file (see “Create a Backup” page 112), then inform your Switchvox users that calls will be dropped and members will be logged out of their queues.

Networking

Networking includes tools to manage the following:

IP Configuration **115**
Network Proxy Settings **117**
Access Control **118**
Phone Networks **121**

IP Configuration

This tool lets you set the IP configuration for Switchvox. These settings are similar to settings you would set on any other computer on your network. Enter the following information, then click **Save IP Configuration**.

Gateway Address. Enter the IP address of the machine to which Switchvox sends outbound traffic. It is typically the address of your router.

DNS Addresses. 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” page 118 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. Enter 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.

Interface. Enter 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. If you need to change an Advanced Option, click **Show Advanced Options** and enter the following information:

--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>.

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.

Network Proxy Settings

These settings direct all outgoing HTTP and HTTPS traffic to a web proxy.

- Proxy Hostname or IP Address
- Proxy Port
- Proxy Username
- Proxy Password
- Custom SSL Certificates

Use these fields 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 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.

Access Control

The Access Control tool lets you manage which networks have access to Switchvox.

- Access Control Rules 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.
- Blocked IPs are automatically blocked by Switchvox. This tool lets you see what has been blocked and lets you unblock it. It also lets you change the blocking options and search for blocked IPs.
- **Create Access Control Rule.**

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.

Create an Access Control Rule. Click **Create Access Control Rule** and enter or set the following information for the rule:

- Enter a unique Rule Name.
- Enter the network IP address and netmask that the rule applies to.
- Set access to YES for each service that you want to allow the network to access.
 - Never Block IPs.** Switchvox will not automatically block a network. For example, you can select YES for Local Network to Never block IPs. For details about how Switchvox automatically blocks IP addresses and how to change it, see "Blocked IPs" page 119.
 - 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" page 107.

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.

Digium Config Server. The phone configuration server for Digium Phones. This must be On to allow access from a remote Digium Phone (meaning, Digium Phones on an external network).

Blocked IPs. Switchvox automatically blocks an IP if it is attempting to log in or register a phone with a bad username and password. This is based on the assumption that if someone is trying to reach Switchvox without valid login information, that person probably does not have a legitimate reason to reach Switchvox.

- To change the options for automatic blocking, click **Edit Blocking Options**.
- To unblock an IP address or netmask, click **Unblock** on that item.
- To find an IP address that has been blocked, use the Search field. You can enter an IP address, an extension number, or an administrator's login.

NOTE: If you have physical access to the Switchvox server, you can use the Basic Server Function Unblock Local IPs to remove the blocks for local IP addresses. (See “Basic Server Functions” page 120)

Edit Blocking Options

The IP Blocking Options determine when Switchvox blocks an IP address from logging into or connecting to Switchvox.

Web Suite and API Logins. --

--Login attempts before lock out. If an IP address attempts to log in as a Switchvox extension or administrator, but fails this number of times, then the IP address is locked out of Switchvox. That means the IP can reach the Switchvox Web Suite and API, but no login information can be entered.

--Login attempts before IP block. If an IP address attempts to log in as a Switchvox extension or administrator, but fails this number of times, then the IP address is blocked from connecting to Switchvox. That means that Switchvox does not let that IP display any part of the Web Suite, or access the API.

--Number of blocked IPs from the same network to trigger a netmask block. If this number of blocked IPs are from the same network, then the /24 network is blocked. In other words, if the first 3 sets of digits match on more than 25 blocked IPs, then the netmask block (e.g., 216.239.51.0/24) replaces all of the original IP blocks. That means that all of the 256 possible IP addresses in that netmask are denied access to Switchvox.

SIP Registration

--Registration attempts before IP block. If a phone attempts to register, but fails this number of times, then its IP address is blocked from connecting to Switchvox.

--Number of blocked IPs from the same network to trigger a netmask block. If this number of blocked IPs are from the same network, then the /24 network is blocked. In other words, if the first 3 sets of digits match on more than 25 blocked IPs, then the netmask block (e.g., 216.239.51.0/24) replaces all of the original IP blocks. That means that all of the 256 possible IP addresses in that netmask are denied access to Switchvox.

General

--Share blocked IPs with Digium. If this is checked, it allows Switchvox to send the blocked IP addresses to Digium. This may help us identify common IP addresses that are being blocked.

Manage 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.

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.

- **View System Info.** This displays the URL of the Switchvox Admin Tool Suite.
- **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 Configuring Your Network Installing Switchvox). You can only do this from here at the server.
- **Reboot PBX.** This reboots Switchvox. You can also do this from Server > Maintenance: System Reload (see System Reload).
- **Shutdown PBX.** This shuts down Switchvox and turns off the power on the appliance. You can also do this from Server > Maintenance: System Reload (see System Reload).
- **Reset HTTPS certificate.** This resets your SSL certificate. You can only do this from here at the server.
- **Tech Support Access.** This lets Switchvox technical support representatives log into your Switchvox Appliance. You can also do this from Reporting > Diagnostics: Tech Support (see Technical Support).
- **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.
- **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.
- **Unblock Local IPs.** This removes the block(s) for the local network. That means any Blocked IPs where the first 3 sets of digits match the first 3 sets in Switchvox's IP (e.g., a /24 CIDR). You can only do this from here at the server. (See also: Blocked IPs.).
- **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.
- **Lock 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.

Phone Networks

Phone Networks determine how a Digium Phone should behave, depending on the IP address and netmask of the Phone as it sits on the network. If the Phone cannot get on the network because it is trying in vain to use the wrong Phone Networks information, use the Phone's **Select Network** option to select the Phone Network that matches the network you want the phone to use. This is accessed via the Phone's Main Menu > Preferences > Select Network.

IMPORTANT: for a remote phone (a phone that is not on the same network as Switchvox), make sure that you have an Access Control Rules in place (see "Create an Access Control Rule" on page 118.), and that your router is forwarding the appropriate ports to Switchvox.

When creating a Phone Network, enter following information:

Name. A unique name for this network.

Network. The IP address and Netmask of the network where Digium Phones could be located. This should be the IP address that the phone obtains and uses to identify itself on this network. For example, this might be a subnet in your office, or the IP address of a phone at an employee's home. (This is not a remote employee's external IP address, which you would use in an Access Control Rule.)

Direct Port Access. Set to YES if the phone, while on this network, has access to these ports on Switchvox:

Port 80, for getting phone firmware. (If a remote phone cannot reach your Switchvox to get phone firmware, the phone will get the firmware from Digium's servers).

Port 443, for getting additional files and accessing Switchvox's API.

Port 5062, for getting configuration information.

Normally a phone just uses port 5060 for all of these activities and for handling calls. However, if set to YES, the phone uses those ports as indicated. This is useful because those protocols are better suited for those activities. That's why the default phone network Internal is set to YES.

VLAN Discovery Mode.

LLDP is the default. In this case, a Digium Phone uses LLDP to discover Virtual Local Area Network information. If the phone does not get a response from LLDP, it will assume that there is no VLAN. (VLAN information cannot be obtained via DHCP.)

Manual allows you want to set a specific VLAN ID and QoS priority number for the phone to use. The VLAN configuration-information attributes are

lldp-med.policy.vlan.vid

lldp-med.policy.priority

None should be selected if you do not want the phone to attempt to use LLDP, and you do not want the phone to use a specific VLAN ID and QoS number.

PBX Address. Switchvox's IP address to be used by a Digium Phone located on this network.

The default Phone Networks entries (Internal and All Networks) are automatically updated so that Host Address is the same as Switchvox's internal and external IP addresses in IP Configuration. However, if you make a change to Host Address in either one of the default entries, then changes you make in IP Configuration will not ever affect that entry.

Alternate PBX Address. In the event that the Phone can no longer reach Switchvox at the PBX Address above, the Phone will register to this IP address.

If this happens, the Phone will be in Failover Mode, and many of its Phone Applications will not be available. However, the phone should be able to make and receive calls over this alternate host.

NTP Server. By default, Digium Phones use NTP to retrieve time settings. Unless you have a particular reason, do not change these settings.

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