



ADMINISTRATOR GUIDE

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Poly VVX Business IP Phones, OBi Edition

Getting Help

For more information about installing, configuring, and administering Poly/Polycom products or services, go to Polycom Support.

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Before You Begin

Topics:

- [Related Poly and Partner Resources](#)
- [Trunks, Endpoints, and Terminals Usage](#)
- [SIP/SP Services](#)

This guide describes how to administer, configure, and provision Poly VVX business IP phones with OBi Edition software.

Related Poly and Partner Resources

See the following sites for information related to this product.

- The [Poly Online Support Center](#) is the entry point to online product, service, and solution support information including Video Tutorials, Documents & Software, Knowledge Base, Community Discussions, Poly University, and additional services.
- The [Poly Document Library](#) provides support documentation for active products, services, and solutions. The documentation displays in responsive HTML5 format so that you can easily access and view installation, configuration, or administration content from any online device.
- The [Poly Community](#) provides access to the latest developer and support information. Create an account to access Poly support personnel and participate in developer and support forums. You can find the latest information on hardware, software, and partner solutions topics, share ideas, and solve problems with your colleagues.
- The [Poly Partner Network](#) is a program where resellers, distributors, solutions providers, and unified communications providers deliver high-value business solutions that meet critical customer needs, making it easy for you to communicate face-to-face with the applications and devices you use every day.
- The [Poly Services](#) help your business succeed and get the most out of your investment through the benefits of collaboration.

Privacy Policy

Poly products and services process customer data in a manner consistent with the [Poly Privacy Policy](#). Please direct comments or questions to privacy@poly.com

Trunks, Endpoints, and Terminals Usage

The phone and the auto attendant are considered *endpoints* where calls can terminate, and *trunks* rely on the corresponding service providers to terminate the call.

Like the trunks, each endpoint is represented by a two-letter abbreviation and an instance identifier equal to 1:

- PH = phone
- AA = auto attendant

Unless stated otherwise, abbreviated endpoint names are case insensitive.

A trunk or an endpoint is also referred to as a *terminal* in this document.

SIP/SP Services

You can configure as many as six SIP/SP service accounts for your phones.

In this document and in the system web interface, *internet telephony service provider (ITSP)* is used generically to describe the logical entity providing the SIP trunk service to your phone. When your phone is used with an IP PBX, ITSP refers to the IP PBX.

Each ITSP configuration is grouped as an ITSP profile. There are six ITSP profiles available: ITSP profile A, B, C, D, E, and F. SP service account specifics are grouped under the heading SP_n service, where $n = 1$ to 6.

ITSP profiles include the following parameters:

- ProxyServer
- OutboundProxy
- DigitMap

Note: ITSP profiles don't include account-specific parameters.

SP services include the following account-specific parameters:

- AuthUserName (usually, but not necessarily, the same as the phone number of the account)
- AuthPassword
- CallerIDName
- X_ServProfile (which ITSP profile to apply the ITSP-specific parameters from)

You must set the following parameters to enable the phone to the SP_n service using ITSP profile X, where $X = A$ to F and $n = 1$ to 6:

- ITSP Profile X - `SIP::ProxyServer = <value>`
- `SPn Service::Enabled = true` (or selected in the system web interface)
- `SPn Service::AuthUsername = <value>`

Note: If you don't set these parameters, your phone considers the service disabled.

If two SP services use the same ITSP, you may be able to refer both SP services to the same ITSP profile. However, you may need to configure multiple ITSP profiles for the same ITSP and have each individual SP service using that ITSP to point to a different ITSP profile.

Getting Started

Topics:

- [Poly VVX Business IP Phones, OBi Edition, Overview](#)
- [Configuration Options](#)

VVX business IP phones, OBi Edition, offer high-resolution LCD displays with a customizable user interface, as well as multiple fully programmable feature keys.

Manage the phone's network interaction directly from your phone or through the system web interface.

Find information on how to use the IVR system and other concepts in this guide in the *Poly OBi Device Technical Reference*, at the [Poly Online Support Center](#).

Poly VVX Business IP Phones, OBi Edition, Overview

VVX business IP phones, OBi Edition, support Polycom HD Voice technology and implement the following features and functionalities:

- Six SIP accounts with universal inter- and intraservice two-way call bridging among the six accounts and the OBiTALK service.
- High-quality voice encoding using G.711, G.729, G.722, iLBC, and OPUS algorithms.
- Recursive digit maps and associated call routing (outbound and inbound).
- Fully programmable line keys and softkeys.
- Programmable feature keys with preassigned functions and labels.
- Support for all standard SIP-based IP PBX and ITSPs/VSPs.
- Support for 3CX PBX and uaCSTA interoperability.
- Suited for all service provider and enterprise deployment environments, regardless of size.
- Ideal for self-service installations—home users, small business owners, or corporate IT departments
- Integration with softswitch architectures.
- Cloud management enabled via [OBiTALK.com](#) with both a user portal and an ITSP partner portal with an optional REST API.

Overview of Poly VVX Business IP Phone Hardware

VVX business IP phones consist of the following components (depending on model):

- VVX 150, 250, 350, and 450 desk phone unit:
 - VVX 150 business IP phones feature a compact monochrome display.
 - VVX 250, 350, and 450 business IP phones feature a high-resolution color active matrix TFT LCD display with a customizable user interface.
- VDC power adapter: Powers your phone when Power over Ethernet (PoE) isn't available.

- Polycom Wi-Fi wireless network adapter: Connects your phone to a wireless network (VVX 250, 350, and 450 only).
- USB flash drive: Use to copy ringtones, background images, and other media to your phone (VVX 250, 350, and 450 only).
- USB headset: Use to answer your phone without using the handset or speakerphone (VVX 250, 350, and 450 only).
- VVX EM50 expansion module: Adds additional lines to your phone (VVX 450 only).

Note: VVX 150 doesn't have a USB port, so you can't connect any USB devices.

VVX business IP phones have a three-port Ethernet switch. The switch consists of one internally connected port to the phone processor for traffic to and from the phone and two Ethernet ports, labeled **SW** and **PC**.

Configuration Options

Configure and manage your VVX business IP phones using either the phone's local interface, the system web interface, or the PDMS-SP portal.

Choose the method that best suits your deployment scenario.

Find the Phone's IP Address

To access the device local interface and system web interface, you must have the IP address of your phone.

Each phone must have a valid IPv4 address to connect to the network and communicate with other devices or cloud-based services.

Procedure

- » Do one of the following:
 - Launch the **Product Info App** from the **Main** menu.
 - Go to **Settings > Network**.
 - Dial *******, then select option **1** to enter the **Voice Admin** menu.

Related Links

[Access the System Web Interface](#) on page 12

Access the Phone's Local Interface

Access and configure VVX business IP phones using the phone's local interface.

Procedure

1. Press the **Home** key.
2. Using the navigation keys and the **Select** key, go to **Settings**.
3. Configure and make changes to your device as necessary.

Note: Use the same administrator password you use for the system web interface to access the **Voice Services** and **Device Administration** options in the phone's local interface.

Access the System Web Interface

VVX business IP phones have a system web interface that you can access from any standard web browser.

You must first obtain your phone's IP address to access system web interface.

The system web interface offers configurable options and status information organized into a number of web pages. There are many configurable parameters available on the phone, organized into a number of device configuration web pages. By browsing through the web pages, you can view all the parameters that you can configure and read or set their values.

Each web page is divided into three frames:

- A customizable top frame
- A left frame that lists the links to available pages
- A main frame that shows the parameters of the currently selected page

Use the menu tree on the left side of the page to navigate the various configuration parameter sections of your phone.

Note: You must submit changes made on each configuration page before moving to another page. Otherwise, you lose any changes you made when you navigate to another page. Most changes require a restart of the phone to take effect.

Procedure

1. Enter your phone's IP address into a web browser.
2. When prompted, enter `admin` for the username and `admin` for the password.
If the default username and password change, log in with the updated credentials.
3. Enter your changes in the system web interface.

Note: Always select **Submit** to save your changes before going to another page.

Related Links

[Find the Phone's IP Address](#) on page 11

Access the PDMS-SP OBiTALK Portal

PDMS-SP OBiTALK is a device management portal website that serves Poly customers and service providers deploying OBi devices.

PDMS-SP OBiTALK uses remote provisioning to manage OBi Edition devices. It stores or dynamically generates on demand a configuration file for each managed device and periodically checks in with the OBiTALK server for configuration updates.

You can register your phones on OBiTALK, which enables you to access all your phones without having to enter the URL `http://Device-IP-Address/` for each phone.

The appearance between the PDMS-SP OBiTALK portal and the system web interface is different, but administrators have identical access to settings.

Once you add your devices on OBiTALK, you can manage them on the PDMS-SP OBiTALK portal.

Procedure

1. In a web browser, enter the URL `www1.obitalk.com`.
2. Enter your username and password.
3. Select **Sign in**.

Setting Up the System

Topics:

- [Power Your Phone](#)
- [Poly VVX EM50 Expansion Modules](#)
- [Configure the Primary Line](#)
- [Set the Line Capacity](#)
- [Set Service Account Credentials](#)

After you set up and power on your VVX business IP phones, you can register your handsets and start configuring features.

For more information about setting up your phone, go to the [Poly Online Support Center](#) and view the following documentation:

- *Poly VVX 150 and VVX 250 Business IP Phones Setup Sheet*
- *Poly VVX 350 and VVX 450 Business IP Phones Setup Sheet*
- *Wallmount and Face Plate Instructions for Poly VVX 150, 250, 350 and 450*

Power Your Phone

You can power your phone using the provided AC power adapter or using PoE.

Procedure

- » Do one of the following:
 - Plug a PoE cable to the phone's **SW** Ethernet switch port.
 - Plug your phone's AC power adapter into a power source.

Note: Use either the AC power adapter or the PoE cable as a power source, but not both.

Poly VVX EM50 Expansion Modules

Add additional lines to your VVX 450 business IP phone by connecting a Poly VVX EM50 expansion module.

Note: VVX 150, 250, and 350 business IP phones don't support expansion modules. You can use Poly VVX EM50 expansion modules only on VVX 450 business IP phones, OBi Edition, with OBi Software 6.4.0 or later.

The VVX EM50 expansion module supports the following features and functionality:

- Assign a function to each line key.
- Each expansion module supports 90 lines (three pages with 30 line keys each).

- Connect up to two expansion modules to a VVX 450 business IP phone to support a total of 228 lines.
- Use the USB port to connect low-powered devices.

The line keys on VVX 450 business IP phones, OBi Edition, and VVX EM50 expansion modules are numbered sequentially. For example, the VVX 450 business IP phone, OBi Edition supports 48 lines, numbered 1 to 48. The first line on the connected expansion module is line 49.

The line keys on the expansion module are divided into 15 left and 15 right keys on each page.

Set Up a VVX EM50 Expansion Module

VVX EM50 expansion modules are powered and signaled by VVX business IP phones and require minimal setup.

Expansion modules automatically configure and update when you connect them to the phone.

Procedure

- » Connect a VVX EM50 expansion module to the USB port on the side of the VVX 450 business IP phone.

The LED lights on the expansion module line keys flash red and green as it initializes.

Load a Background Image on the Expansion Module

The default background image on the VVX EM50 expansion module (side car) is the same as the phone background.

When you set the background image on the phone, the expansion module background automatically updates. If you attach more than one EM50 expansion module, both show the same background picture.

Note: The optimal image resolution is 480 px wide by 800 px high.

Procedure

1. In the system web interface, go to **User Settings > User Preferences**.
2. In the **Value** column for the `SideCarBackgroundPicture` parameter, enter the website/server/URL that is hosting the specific image file that you want to have as the background image of the EM50.

Use an empty value to use the same background image as your phone.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Primary Line

The primary line is the default service used to make calls when not using an explicit access code prefix.

The reserved name *pli*, which is found in the phone `DigitMap` and `OutboundCallRoute` parameters, is substituted by the corresponding name of the primary line:

Primary Line	Parameter Value
SP1	sp1
SP2	sp2
SP3	sp3
SP4	sp4
SP5	sp5
SP6	sp6
OBI TALK	pp
Trunk Group 1	tg1
Trunk Group 2	tg2

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.
2. In the **Value** column for the `PrimaryLine` parameter, enter the service you want to set as the primary line (values are case sensitive):
 - SP1 Service
 - SP2 Service
 - SP3 Service
 - SP4 Service
 - SP5 Service
 - SP6 Service
 - OBI TALK Service
 - Trunk Group 1
 - Trunk Group 2
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set the Line Capacity

Line capacity refers to the maximum number of simultaneous calls that can be active per line. Some of the calls may be on hold.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service > Calling Features** (where $N=1$ to 6 services).
2. In the **Value** column for the `MaxSessions` parameter, enter the maximum number of simultaneous calls to allow on the service.

Note: Set the number equal to or less than what the underlying service provider can support. The default value is 2 for all lines.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Service Account Credentials

All SP services except PDMS-SP OBiTALK require credentials. You must configure a username for SP service accounts.

Procedure

1. In the system web interface, go to **Voice Services > SP n Service > Credentials** (where $n = 1$ to 6 services).
2. In the **Value** column, configure the following parameters:

Parameter	Description
AuthUserName	Enter a username to authenticate the connection to the server.
AuthPassword	(Optional) Enter a password to authenticate the connection to the server
URI	Set a username if the username for SIP authentication is different from the account username.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Network Settings

Topics:

- [Network Connectivity](#)
- [DNS Lookup](#)
- [NTP Servers and Local Time](#)
- [Configuring Proxy Server Settings](#)
- [SIP Registration](#)
- [SIP Server DNS Settings](#)
- [SIP Privacy](#)
- [NAT Traversal](#)

Network settings include the system network configurations.

You also can register your system with SIP for calling.

Network Connectivity

VVX business IP phones offer two interfaces for networking: Ethernet (indicated as **WAN** in the local interface and system web interface) and Wi-Fi.

You can use both network options at the same time, but Ethernet takes precedence over Wi-Fi.

For more information on parameters, see the appropriate data model for your phone:

- [VVX 150 Data Model](#)
- [VVX 250 Data Model](#)
- [VVX 350 Data Model](#)
- [VVX 450 Data Model](#)

Configure the External Ethernet Ports

You can configure the two external Ethernet ports from the system web interface.

Procedure

1. In the system web interface, go to **System Management > WAN Settings**.
2. Configure the following settings:

Parameter	Values
WAN Settings - Switch	• Auto
Port::SpeedWAN Settings - PC	• Full
Port::Speed	• Half
	• Full
	• Half
	• Full
	• Disabled
Poly recommends using only the Auto or Disabled settings.	

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the WAN Interface

The WAN interface on the phone refers to the internal Ethernet switch port connected directly to the phone processor.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Internet Settings**.
2. Configure the following settings:

Setting Group	Description
VLAN	The phones support VLAN tagging in compliance with 802.1p/q. If you enable VLANEnable, the phone tags outbound traffic according to the VLANID and VLANPriority parameters. The phones ignore inbound traffic that doesn't belong to the same VLAN.
LLDP	The phones support LLDP-MED to automatically discover Network Policy (VLAN and DSCP) settings and perform other related handshake functions.
IP Address Assignment	The phones support different methods of acquiring an IP address assigned to its WAN interface. Configure the method using the AddressingType parameter, which can have one of the following values: • DHCP: Request address assignment from a DHCP server. • Static: Use the statically assigned IP address, subnet mask, and default gateway from the IPAddress, SubnetMask and DefaultGateway parameters, respectively.
DNS Servers	You can specify as many as two DNS servers to use with the WAN interface using DNSServer1 and DNSServer2. Note: If the DHCP offer includes DNS Servers, the phone takes as many as 16 servers from the list and uses them together with the explicitly configured servers.

Setting Group	Description
Cisco Discovery Protocol (CDP)	The phones support CDP for automated network setting discovery. Common values included in CDP broadcast messages are: Device Type and Model, Duplex/Speed Setting, VLAN Setting, and PoE Class (Power Draw).

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Wi-Fi Interface

VVX 250, VVX 350, and VVX 450 business IP phones support Wi-Fi via the Poly Wi-Fi wireless network adapter.

Connect the adapter to one of the USB ports on the phone.

Note: VLAN and LLDP features aren't available on Wi-Fi.

Procedure

1. In the system web interface, go to **OBiWiFi Configuration > Wi-Fi Settings**.
2. Configure the following settings:

Setting Group	Description
IP Address Assignment	The phones support different methods of acquiring an IP address assigned to its Wi-Fi interface. Configure the method using the <code>Basic Settings::AddressingType</code> parameter, which can have one of the following values: • DHCP: Request address assignment from a DHCP server. • Static: Use the statically assigned IP address, subnet mask, and default gateway from the WAN Settings - Internet Settings::IPAddress, SubnetMask and DefaultGateway parameters, respectively.
DNS Servers	You can specify up to two DNS servers to use with the Wi-Fi interface using <code>Internet Settings::DNSServer1</code> and <code>DNSServer2</code>. Note: If the DHCP offer includes DNS Servers, the phone takes as many as 16 servers from the list and uses them together with the explicitly configured servers.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

DHCP Options

VVX business IP phones support DHCP network options for both Ethernet and Wi-Fi.

Configure DHCP Options

The options that the phone tries to extract from the DHCP offer is a comma-separated list of option numbers that you can configure using the `ExtraOptions` parameter.

The phone doesn't recognize any other option numbers.

For more information on parameters, see the appropriate data model for your phone:

- [VVX 150 Data Model](#)
- [VVX 250 Data Model](#)
- [VVX 350 Data Model](#)
- [VVX 450 Data Model](#)

Procedure

1. In the system web interface, go to **System Management > WAN Settings > DHCP Client Settings**.
2. In the **Value** column for the `ExtraOptions` parameter, enter the extra DHCP options, separated by commas.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set the Config File URL

Set the URL that the system uses for your phone in the system's config file.

You can use the following macros to refer to the values of these options in any of the configuration parameters:

- `$DHCPOPT66`
- `$DHCPOPT150`
- `$DHCPOPT159`
- `$DHCPOPT160`
- `$DHCPOPT161`

Procedure

1. In the system web interface, go to **System Management > Auto Provisioning > ITSP Provisioning**.
2. In the **Value** column of the `ConfigURL` parameter, enter the URL of the config file.
The default value of the `ConfigURL` is `tftp://$DHCPOPT66/$DM.xml`.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

DNS Lookup

The following DNS behavior applies to both the WAN and Wi-Fi network interfaces.

Configure Lookup Order

When your phones obtain DNS servers from both DHCP and statically configured values, the phone queries all the servers of one type before moving to the other type.

Use the `DNSQueryOrder` parameter to control the order the phone queries the server types.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > DNS Control**.
2. In the **Value** column, select the available DNS servers for the `DNSQueryOrder` parameter in the order you want them to be queried by the phone.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure DNS Query Delay

When there are multiple DNS servers available, the phone queries as many DNS servers as necessary to resolve a domain name.

You can configure the `DNSQueryDelay` parameter to insert a short delay between each query to stop once the phone receives a positive response.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > DNS Control**.
2. In the **Value** column, select a value for the `DNSQueryDelay` parameter to insert the delay.
The query order follows the `DNSQueryOrder` setting. If you set the delay to 0, the phone queries all the DNS servers at the same time.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define Local DNS Records

You can define as many as 32 local DNS records in the phone configuration.

Once you define the DNS records, you can enable the phone to search through these records before hitting the external DNS services when attempting to resolve a domain name. These records can be A or SRV records.

Note: The only way to provide a list of redundant servers to the phone is through the use of DNS A or DNS SRV records.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Local DNS Records**.
2. In the **Value** column, enter a local DNS record for each parameter name.
3. Select **Submit**.

4. Reboot your system when you complete your changes.

NTP Servers and Local Time

The phone keeps track of the current time by querying NTP servers (using SNTP).

Configure NTP Servers

You can configure up to two NTP servers using the `NTPServer1` and `NTPServer2` parameters.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Time Service Settings**.
2. In the **Value** column, enter the host names or IP addresses of the NTP servers for the `NTPServer1` and `NTPServer2` parameters.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Disable SNTP Discovery

By default, the phone discovers the SNTP server using DHCP option 42 and discovered servers take precedence.

You can disable SNTP discovery in DHCP and use the configured SNTP servers instead.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > DHCP Client Settings**.
2. In the **Value** column, delete option 42 from the extra options list for the `ExtraOptions` parameter.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set the Local Time Zone

The phone determines the local time by the `LocalTimeZone` parameter.

The phone queries the NTP servers once per hour to update the current time and the phone uses its own local clock in between NTP refreshes.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Time Service Settings**.
2. In the **Value** column, select your local time zone from the list for the `LocalTimeZone` parameter.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Daylight Saving Time

You can enable, set the start and end rules, and configure the amount of time to adjust for daylight saving time.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Time Service Settings**.
2. In the **Value** column for the `DaylightSavingTimeEnable` parameter, select the check box to enable daylight saving time.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Specify Start and End Rules for Daylight Saving Time

You can set the start and end rules so the phone automatically adjusts for daylight saving time.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Time Service Settings**.
2. In the **Value** column for the `DaylightSavingTimeStart` parameter, enter the date to start daylight saving time.

The format for the date is `mon/day/weekday/h:m:s` using the following values:

Variable	Value
<code>mon</code>	1 to 12
<code>day</code>	1 to 31 or -1 to -31 (if counting from the end of the month)
<code>weekday</code>	0 for the exact day 1 = Monday 2 = Tuesday 3 = Wednesday 4 = Thursday 5 = Friday 6 = Saturday 7 = Sunday
<code>h:m:s</code>	hour, minute, second

3. In the **Value** column for the `DaylightSavingTimeEnd` parameter, enter the date to end daylight saving time.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Amount of Time to Adjust for Daylight Saving Time

You can configure the amount of time to adjust when daylight saving time is in effect.

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Time Service Settings**.
2. In the **Value** column for the `DaylightSavingTimeDiff` parameter, enter the amount of time to adjust for when daylight saving time takes effect.

The format is `h:m:s` or `-h:m:s`.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable SIP Signaling

You can also enable the phone to use SIP signaling to tell time.

When the phone renews registration with a SIP proxy server, the server may include a Date header in the response to the phone that indicates the current GMT time.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > X** (where X=the ITSP profile for the service provider).
2. In the **Value** column, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Proxy Server Settings

Configure the proxy server for the correct settings you need for your deployment.

For TCP/TLS, your phone must start a TCP/TLS connection with the `ProxyServer` and use the same connection to exchange all subsequent SIP messages. If you specify an `OutboundProxy`, your phone starts the TCP/TLS connection with the `OutboundProxy` instead. With the `OutboundProxyTransport` parameter, it's possible to choose a different transport to use with the `OutboundProxy` and with the `ProxyServer`.

Configure the Proxy Server for SIP/SP Services

For SIP/SP services that don't use TCP/TLS, configure the proxy server as needed for your deployment.

Procedure

1. In the system web interface, go to **Service Providers > ITSP > X** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the following parameters, configure the proxy server:

Parameter	Description
<code>ProxyServer</code>	Enter the host name or IP address of the SIP proxy server.
<code>ProxyServerPort</code>	Configure the destination port to connect to the SIP server if the listening port is nonstandard.
Note: The <code>RegistrarServer</code> and <code>RegistrarServerPort</code> parameters are rarely needed. Your phone assumes that the SIP Registrar is the same as the SIP Proxy Server if they aren't specified separately.	

Configure the Transport Protocol for SIP Messages

You can set the transport protocol for SIP messages.

Your phone supports UDP, TCP, and TLS protocols for SIP transport.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > x > SIP** (where x=the ITSP profile for the service provider).
2. In the **Value** column for the `ProxyServerTransport` parameter, select the transport protocol to connect to the SIP server.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Local Contact Port for SIP Messages

You can configure the local contact port for sending and receiving SIP messages.

The `SPn` services for $n = 1 - 6$ each use a different local contact port for sending and receiving SIP messages (the defaults are 5060, 5061, 5062, 5063, 5064, and 5065 respectively).

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1-6$).
2. In the **Value** column for the `X_UserAgentPort` parameter, enter the UDP, TCP, or TLS port for sending and receiving SIP packets.

The default server port is 5060 for UDP/TCP and 5061 for TLS.

Note: `ProxyServer` and `RegistrarServer` must use the same transport protocol for SIP messages that you set in the `ProxyServerTransport` parameter.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

SIP Outbound Proxy Server

You can configure an outbound proxy server on your phone, such that all outbound requests are sent via the outbound proxy server instead of directly to the SIP proxy server or registration server.

Use an outbound proxy when the device-facing server is a session border controller (SBC). Configure the outbound proxy destination port if the outbound proxy isn't listening at the standard port.

Configure the Port for Connecting to the Outbound Proxy Server

If the outbound proxy server is listening at a nonstandard port, you must specify the correct port value.

Procedure

1. In the system web interface, go to **Service Providers > ITSP > x > SIP** (where x=the ITSP profile for the service provider).
2. In the **Value** column for the `OutboundProxyPort` parameter, enter the destination port you want to use when connecting to the outbound proxy.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Transport Protocol for the Outbound Proxy Server

The **outbound proxy** server may use a different transport protocol from the proxy server, which you can configure.

If the **OutboundProxyTransport** protocol is TCP or TLS, your phone initiates a TCP or TLS connection only with the **outbound proxy**. All subsequent messages exchanged between your phone and the servers must use the same connection. If for any reason the connection is closed, your phone attempts to re-establish the connection with the **OutboundProxy** following an exponential back-off retry pattern.

Note: Even though your phone only exchanges messages directly with the outbound proxy server, the `ProxyServer`, `ProxyServerPort`, and `ProxyServerTransport` parameters are still very important since the SIP requests sent by your phone to the server are formed based on these values, not the **OutboundProxy** value. Never configure your phone to include the **OutboundProxy** value in the SIP requests generated by your phone, unless the **OutboundProxy** has the same value as the **ProxyServer**.

Procedure

1. In the system web interface, go to **Service Providers > ITSP > x > SIP** (where x=the ITSP profile for the service provider).
2. In the **Value** column for the `X_OutboundProxyTransport` parameter, select the transport protocol to connect to the outbound proxy server.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Bypass the Outbound Proxy Server After Initial Call Setup

Some server implementations include the outbound proxy server in a Record-Route header such that your phone shouldn't respect the locally configured **OutboundProxy** value after the initial INVITE is sent for a new call.

You can achieve this behavior by enabling the `X_BypassOutboundProxyInCall` parameter. However, this option has no effect when the **OutboundProxyTransport** is TCP or TLS, as your phone always uses the same connection to send messages to the server.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > x** (where x=the ITSP profile for the service provider).
2. In the **Value** column for the `X_BypassOutboundProxyInCall` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

SIP Registration

The main purpose of SIP registration is to create and maintain a dynamic binding of the SIP/SP account to your phone's local contact address.

Service providers can also rely on this periodic message to see if your phone is online and functional.

Enable Registration with the ProxyServer or RegistrarServer

You can set up your phones to periodically register to the **ProxyServer** or the **RegistrarServer**.

ProxyServer is a required parameter that you must configure on your phone, while **RegistrarServer** is optional and assumed to be the same as the **ProxyServer** if not specified in the configuration.

Note: If the server isn't listening at the standard port, configure the correct port value in **ProxyServerPort** (and **RegistrarServerPort** as needed).

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > x > SIP** (where x=the ITSP profile for the service provider).
2. In the **Value** column, enter the host name or IP address for one of the following parameters:
 - `ProxyServer`
 - `RegistrarServer`
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Third-Party Registration

Third-party registration registers the system for an Address of Record (AOR) that isn't the same as the account user-id.

That is, the user-id in the TO header of the SIP REGISTER request is different from that in the FROM header, which always carries the account user-id.

Procedure

1. In the system web interface, go to **Voice Services > SP n Service > Share Line Features** (where $n=1-6$ services).
2. In the **Value** column for the `X_ShareLineUserID` parameter, enter the third-party user-id to register to for the shared line.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Registration Period

Configure the nominal registration Expires header value (implemented as a Contact header parameter value in seconds) used by your phone in REGISTER requests.

For more information about the registration period, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > x > SIP** (where x =the ITSP profile for the service provider).
2. In the **Value** column for the `RegistrationPeriod` parameter, enter the nominal interval between device registration, in seconds.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Registration Margin

Your phone computes the next renewal time by subtracting a percentage of the Expires value derived from the 2xx response returned by the server.

Use the `X_RegistrationMargin` parameter to control how the margin to subtract is computed.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile > x > SIP** (where x =the ITSP profile for the service provider).
2. In the **Value** column for the `X_RegistrationMargin` parameter, enter the number of seconds before registration when the phone should re-register.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

SIP Proxy Server Redundancy and Dual Registration

You must enable device registration to use the server redundancy feature.

Other SIP requests, such as INVITE or SUBSCRIBE, are sent to the same server that your phone is currently registered to. Server redundancy specifically refers to your phone's capability to do the following:

- Look for a working server to register to from a list of candidates
- Switch to another server once the server that it currently registers to becomes unresponsive

For more information on SIP proxy server redundancy and dual registration, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Configure Server Redundancy

If your phone uses an outbound proxy server, the phone applies server redundancy to the outbound proxy server instead of the registration server.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_ProxyServerRedundancy` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Secondary Registration

You can configure your phone to maintain a secondary registration with a server that has lower or equal priority than the primary server.

If the `X_ProxyServerRedundancy` parameter is disabled, the `X_SecondaryRegistration` parameter doesn't have any effect. Your phone resolves only one IP address from the server's domain name and doesn't attempt to try other IP addresses if the server isn't responding.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_SecondaryRegistration` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable the Interval for Checking Primary and Secondary Fallback Lists

You can configure the interval for checking the primary and secondary fallback lists for higher priority proxy servers, which you can use as the primary and secondary proxy servers.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_CheckPrimaryFallbackInterval` parameter, enter the interval (in seconds) for checking the primary fallback list.
3. In the **Value** column for the `X_CheckSecondaryFallbackInterval` parameter, enter the interval (in seconds) for checking the secondary fallback list.
Setting the value to 0 disables this periodic check.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

SIP Server DNS Settings

Configure the DNS and external address settings for your phones.

Configure the SIP Server DNS Lookup

When sending out SIP requests to the server, your phone looks up the IP address of the server using a DNS query if the server is specified as a domain name instead of an IP address.

If you configure an outbound proxy server, your phone uses it instead of the SIP proxy server or SIP registration server.

You can enable the `X_DnsSrvAutoPrefix` parameter to prepend the service prefix for the domain name DNS query, using the following values:

- UDP: `_sip._udp.`
- TCP: `_sip._tcp.`
- TLS: `_sip._tls.`

If the result from the DNS query is an SRV record, the phone takes the server port from that record and ignores the server port value configured on your phone. Otherwise, the phone takes the server port from the configured value. If no value is specified, the phone uses 5060.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_DnsSrvAutoPrefix` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable DNS NAPTR

You can enable DNS Name Authority Pointer (NAPTR) for each ITSP profile.

If enabled, the phone only attempts NAPTR lookup of the domain name specified in the `OutboundProxy` parameter (if configured). Otherwise, the phone uses the `ProxyServer` parameter.

With NAPTRlookup, your phone can discover the hosts to access the SIP service for a given domain, the SIP transport, preference, and order of each host, and what types DNS records to use for each listed host (SRV or A record). This feature complies with RFC 2915.

For more information about DNS NAPTR support, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_DnsNAPTR` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Discover the External IP Address and Port

Your phone uses the discovered external IP address and port to replace its private address and port by enabling the `X_DiscoverPublicAddress` parameter.

This enables your phone to use the public address and port as the SIP contact address.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_DiscoverPublicAddress` parameter, select the check box.

Note: Enabling the `X_UsePublicIPAddressVia` parameter also substitutes the Via address, but this usually isn't necessary.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

SIP Privacy

Your phone observes inbound caller privacy and decodes the caller's name and number from SIP `INVITE` requests by checking the `FROM`, `P-Asserted-Identity` (`PAID` for short), and `Remote-Party-ID` (`RPID` for short) message headers.

All these headers can carry caller's name and number information.

Find more information on SIP privacy in the *Poly OBi Device Technical Reference*, at the [Poly Online Support Center](#).

Enable an RPID Header

For outbound calls, the phone can state the caller's preferred privacy setting in an `RPID` header of the outbound `INVITE` request.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_InsertRemotePartyID` parameter, select the check box (the default value of this parameter).
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Block Outbound Caller ID

You can instruct your phone to use `sip:anonymous@localhost` in the `FROM` header to block outbound caller ID.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).

2. In the **Value** column for the `X_UseAnonymousFROM` parameter, select the check box.
Your phone also includes a `Privacy: id` header if the `X_InsertPrivacyHdr` parameter is enabled.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

NAT Traversal

If your phone sits behind a NAT router, it can discover the mapped external address corresponding to its local SIP contact address.

The server uses the following options:

- The `"received="` and `"rport="` parameters of the VIA header of the REGISTER response sent by the server, which tell your phone its mapped IP address and port number. The server uses this method if you enable periodic registration on your phone.
- The response to a STUN binding request your phone sent to a STUN server. Keep alive messages are sent to the same server where the phone sends a REGISTER request.

Find more information on NAT Traversal considerations in the *Poly OBi Device Technical Reference*, at the [Poly Online Support Center](#).

Configure Keep Alive Messages

Keep the NAT pinhole open by configuring your phone to send out periodic keep alive messages on the same network path.

Poly recommends that you send the keep alive messages to the same proxy server responsible for registration.

Note: The server may drop the keep alive messages before processing them. If you use STUN binding requests as keep alive messages, Poly recommends that the server return a valid STUN binding response to each request.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1$ to 6 services).
2. Configure the following parameters:

Parameter Group	Parameter	Description
SPN Service	<code>X_KeepAliveEnable</code>	Enable sending of keep alive message. If enabled, your phone sends periodic keep-alive messages to the destination specified in <code>X_KeepAliveServer</code> and <code>X_KeepAliveServerPort</code> , at the interval specified in <code>X_KeepAliveExpires</code> . The content of this message is the ASCII string keep-alive\r\n
SPN Service	<code>X_KeepAliveExpires</code>	Keep alive period in seconds.

Parameter Group	Parameter	Description
SPn Service	X_KeepAliveMsgType	The type of keep-alive messages to send out periodically if keep-alive is enabled. One of the following: <code>keep-alive</code>: The string keep-alive for SIP transport over UDP, or CRLF (for example, carriage return and line feed characters) for SIP transport over TCP or TLS. <code>empty</code>: A blank line <code>stun</code>: A standard STUN binding request. Your phone uses the binding response to form its contact address for REGISTRATION. <code>custom</code>: use the value of X_CustomKeepAliveMsg <code>options</code>: a valid SIP OPTIONS request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled. <code>notify</code>: a valid SIP NOTIFY request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled.
SPn Service	X_CustomKeepAliveMsg	Defines the custom message to be used when X_KeepAliveMsgType is custom. The value should have the following format: <pre>mtid=NOTIFY;event={whatever};user={anyone}</pre> Where <code>NOTIFY</code> may be replaced by any other SIP method, such as <code>PING</code>, the event parameter is optional and is only applicable if the method is <code>NOTIFY</code>. - If event isn't specified, the 'keep-alive' event is used with <code>NOTIFY</code>. - The user parameter is optional. If not specified, the <code>request-uri</code> won't have a user ID, and the TO header field uses the same user ID as the FROM header, which is the local account user ID. If user is specified, it's used as the user ID in the <code>Request-URI</code> and TO header. SIP messages for <code>keep-alive</code> are sent only once without retransmission. Responses to the SIP messages are ignored by the phone.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

STUN and ICE

Your device supports standard STUN based on RFC3489 and RFC5389 for passing inbound RTP packets to devices sitting behind NATs.

Find more information about STUN and ICE in the *Poly OBi Device Technical Reference*, at the [Poly Online Support Center](#).

Configure the STUN Feature

You can enable the STUN feature, configure the IP address or domain name of the external STUN server, and add the destination listening UDP port to connect to the STUN server.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > General** (where X=the ITSP profile for the service provider).
2. In the **Value** column, configure the following parameters:

Parameter	Description
STUNEnable	Select the check box to enable the STUN feature (default is NO or FALSE).
STUNServer	Enter the IP address or domain name of the external STUN server to use.
X_STUNServerPort	Enter the destination port to connect to the STUN server.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable the ICE Feature

Interactive Connectivity Establishment (ICE) is a technique that finds ways for two computers to talk to each other in peer-to-peer networking. Configure the ICE feature on your phone using the `X_ICEEnable` parameter.

ICE is done on a per-call basis for automatically discovering which peer address is the best route for sending RTP packets.

ICE is effective if STUN is also enabled; however, STUN isn't a requirement for using ICE on your phone.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_ICEEnable` parameter, select the check box (the default is NO or FALSE).
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Securing the System

Topics:

- [Load Custom Device Certificates](#)
- [Set the 802.1X Authentication Mode](#)
- [Set Port Number for the Management Web Server](#)
- [Access Control for the System Web Interface](#)
- [Configuring Passwords](#)

You can download and add custom certificates to VVX business IP phones, OBi Edition.

All certificates must be in DER or PEM format. When using (EAP) TLS, the client certificate file must also include the private key (PEM) file appended to the client certificate. For more information on supported certificates, see the *Certificate Updates for Poly VVX Business IP Phones, OBi Edition* at the [Poly Online Support Center](#).

Load Custom Device Certificates

You can load custom device certificates in the system web interface.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Custom Device Certificate *n*** (where *n*=1-2).
2. In the **Value** column for the `DownloadURL` parameter, enter the URL of the certificate to be downloaded.
3. Go to **TLSPlatform Profile *n*** (where *n*=1-2).
4. In the **Value** column for the `DeviceCert` parameter, select the custom device certificate.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Set the 802.1X Authentication Mode

You can set the authentication mode using the `802_1XMode` parameter.

VVX business IP phones support the following 802.1X authentication modes:

- Disable
- MD5
- TLS
- TTLS/MSCHAPv2
- PEAP-MSCHAPv2 (optional for all parameters)

Procedure

1. In the system web interface, go to **System Management > WAN Settings > Internet Settings**.
2. Depending on selected mode, configure the following additional authentication parameters:

Parameter	Description	(EAP) MD5	(EAP) TLS (1.0)	TTLS/ MSCHAPv2
802_1XIdentity	A user name. If the value isn't needed, set the value as an empty string.	Required	Required	Required
802_1XPassword	A password or passphrase. If a password or passphrase isn't needed, set the value as an empty string.	Required	Required	Required
802_1XAnonymousID	When empty, anonymous identity isn't used in authentication.		Required	Required
802_1XTLSSecurityProfile	Security profile for the 802.1X authentication.		Required	

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Port Number for the Management Web Server

You can set the port number value for the management web server.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Web Server**.
2. In the **Value** column for the **Port** parameter, enter a number for the port web server (for HTTPS, enter 443 as the value).
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Access Control for the System Web Interface

You can protect access to the system web interface by configuring two passwords on the phone: an administrator password and a user password.

The system only prompts for a user password if you configure one in the system.

Unlike configuration parameters, you can't control the functions under **System Management/Device Update** via provisioning. For these functions, the following restrictions always apply when the current login is the user:

- **Firmware Update:** Removed so that user can't update firmware or AA prompts.
- **Backup AA User Prompts:** Same as admin login.

- **Backup Configuration:** Backup parameters set with user read-only or read-write permission only.
- **Restore Configuration:** Restore parameters set with user read-write permission only.
- **Reset Configuration:** Reset parameters set with user read-write permission only.

Configuring Passwords

You can change the administrator and user passwords.

Poly recommends that you change the default administrator and user passwords at your earliest convenience.

Change the Default Password

If you use the default password, a warning message `Default password is used` appears at the top right of the page.

Change this password and record it in a safe place. If you lose this password, you might need to return your phone for service.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Web Server**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
AdminPassword	Enter a new, case- sensitive administrator password.
AdminPin	Enter a new, case- sensitive PIN to enter the updater menu.
UserPassword	Enter a new, case- sensitive user password.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Passwords in the System Web Interface

Set both admin and user passwords in the system web interface.

Procedure

1. In the system web interface, go to **System Management > Device Admin**.
2. Configure the following settings:
 - For **AdminPassword**, enter an administrator login password.
 - For **UserPassword**, enter a user login password. To choose the default value, select the check box under **Default**.

Set Passwords in the Phone's Local Interface

Set system passwords in the phone's local interface.

Procedure

1. Press the **Home** key and go to **Settings > Device Admin**.
2. Configure the following settings:

Parameter	Description
Port	The web server listen port. Default is 80, the standard HTTP port. Note: Setting this value to 0 disables all web server access.
AdminPassword	Admin login password. Default is <code>admin</code> .
UserPassword	User login password. Default is <code>user</code> .

Set the Minimum Password Length

The service provider can specify the minimum length of the password.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Web Server**.
2. In the **Value** column for the `PasswordMinimumLength` parameter, enter the minimum length for the password.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Call Routing

Topics:

- [Edit the Outbound Call Route Configuration](#)
- [Configure Inbound Call Rules](#)
- [Set Up Digit Maps](#)
- [Configure Outbound Call Rules](#)
- [Customize Service Route Access Codes](#)
- [Auto Attendant](#)

Call routing is the process your system uses to set up an endpoint call based on information such as the trunk on which the call originates, the caller's number, and the called number.

For more information about call routing parameters, see the following data models:

- [VVX 150 Data Model](#)
- [VVX 250 Data Model](#)
- [VVX 350 Data Model](#)
- [VVX 450 Data Model](#)

Edit the Outbound Call Route Configuration

You can edit the routing rules for outbound calls made from the phone.

Note: Poly recommends keeping the default parameters.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.
2. In the **Default** column, clear the check box for the parameters you want to edit.
3. In the **Value** column for these parameters, enter your changes.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Inbound Call Rules

Incoming calls come to the phone via any configured line.

How the phone routes the incoming call is based on the rules defined in the `InboundCallRoute` parameter of each line.

The typical way to route an incoming call is to ring the phone, so the default `InboundCallRoute` value for all services is `ph`.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1-6$).
2. In the **Value** column for the `InboundCallRoute` parameter, enter the routing rule for inbound calls.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Up Digit Maps

A digit map is a succinct way of describing a set of number patterns.

The `DigitMap` parameter determines the set of number patterns that users can dial. You can refer to named digit maps with the `(Mname)` syntax.

For example, the default `DigitMap` parameter refers to digit maps defined for SP1, SP2, SP3, SP4, and OBiTALK services with the reserved names `(Msp1)`, `(Msp2)`, `(Msp3)`, `(Msp4)`, and `(Mpp)` respectively:

```
([1-9]x?* (Mpli) | [1-9]S9 | [1-9] [0-9]S9 | *** | **0 | **1 (Msp1) | **2 (Msp2) | **3 (Msp3) |
**4 (Msp4) | **9 (Mpp) | (Mpli) )
```

This way, the digit map is more readable and organized.

Note: `(Mpli)` refers to the digit map of the primary line.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.
2. In the **Value** column for the `DigitMap` parameter, enter a digit map to set the number patterns users can dial.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Outbound Call Rules

After a user dials a complete number, the phone determines which service to use for the call by applying the call routing rules defined in the `OutboundCallRoute` parameter.

This parameter may refer to named digit maps defined elsewhere in the phone configuration.

The default value refers to `(Msp1)`, `(Msp2)`, `(Msp3)`, `(Msp4)`, and `(Mpp)`, which are digit maps defined for SP1, SP2, SP3, SP4, and OBiTALK services, respectively:

```
{ ([1-9]x?* (Mpli) ) :pp }, { **0 :aa }, { *** :aa2 }, { (<**1:> (Msp1) ) :sp1 },
{ (<**2:> (Msp2) ) :sp2 }, { (<**3:> (Msp3) ) :sp3 }, { (<**4:> (Msp4) ) :sp4 },
{ (<**9:> (Mpp) ) :pp }, { (Mpli) :pli }
```

Note: `pli` refers to the primary line.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.

2. In the **Value** column for the `OutboundCallRoute` parameter, enter the routing rule for outbound calls made from the phone.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Customize Service Route Access Codes

If necessary, you can modify the service route access codes with the `DigitMap` and `OutboundCallRoute` parameters.

Customizing the service route access codes help when you prefer to not use the default service route access code for SP1 (**1), and want to change it to something else.

Note: The phone handles the **PrimaryLine** setting by substituting internally all occurrences of `pli` with the abbreviated name of the trunk named as the primary line in the `DigitMap` and `OutboundCallRoute` parameters of the same parameter group.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.
2. In the **Value** column, configure the following parameters:

Parameter	Value
<code>DigitMap</code>	Modify the digit map to limit numbers that you can dial.
<code>OutboundCallRoute</code>	Modify the routing rule for outbound calls made from the phone.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Auto Attendant

Your phone has an auto attendant (AA) feature that you can configure by including `aa` as the destination of the inbound call routing rules of a trunk (such as SP1 or OBiTALK) and have incoming calls matching those rules on that trunk routed to the AA.

Note: In the phone configuration, the AA as described here is referred to as AA1 or Auto Attendant 1. Throughout this document and the phone configuration, AA is the same as AA1. AA2 on the other hand refers to the IVR system that is used for basic phone configuration.

Enable Auto Attendant

You must enable the Auto Attendant feature on the phone to use it.

Procedure

1. In the system web interface, go to **Voice Services > Auto Attendant > Auto Attendant 1**.
2. In the **Value** column for the `Auto Attendant- Auto Attendant 1` parameter, select the check box to set this parameter to `True`.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the AA Callback Service

Your phone offers two methods for the AA to call you back at the calling number or a number that you pick.

You can statically configure the routing rule for inbound calls on a trunk with the `InboundCallRoute` parameter. The outbound service to be used for the AA to call back is determined according to the `OutboundCallRoute` parameter.

The `CallbackAnswerDelay` parameter controls the number of milliseconds before AA answers when a callback number is specified.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service > Calling Features**.
2. In the **Value** column for the `X_InboundCallRoute` parameter, enter a routing rule for inbound calls on the trunk to direct AA to call back the caller's or another number if the caller hangs up before AA answers.
3. Go to **Auto Attendant > Auto Attendant 1**.
4. In the **Value** column, configure the following parameters:
 - `OuboundCallRoute`: Enter a routing rule for outbound calls made via this AA.
 - `CallbackAnswerDelay`: Enter the number of milliseconds before AA answers when a callback number is specified.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Customizing Service Route Access Codes for the Auto Attendant

You can customize the service route access codes for calling via the Auto Attendant by modifying the `Auto Attendant::DigitMap` and `Auto Attendant::OutboundCallRoute` parameters.

Voice Services

Topics:

- [ITSP Profiles](#)
- [The OBiTALK Service](#)
- [Voice Service Bridges](#)
- [Trunks](#)
- [Voice Gateways](#)

A voice service, also known as a line or trunk, is an individual user account with an ITSP.

You can configure the following voice services on your phone's IP port:

- SP1
- SP2
- SP3
- SP4
- SP5
- SP6
- PDMS-SP OBiTALK

An SP n service can be a generic SIP voice service. SIP/SP services include an extension from a PBX or a subscriber account with a service provider.

PDMS-SP OBiTALK is a built-in service that Poly provides for customers to manage their devices and use for technical support and device-to-device calling among OBi devices.

No matter what technology the service provider equipment uses, you must provision it into your phone configuration as a domain name or an IP address along with a port number if the server isn't listening at the standard port (5060). Since OBiTALK servers are known by your phone, there's no need to configure the server domains for this service.

ITSP Profiles

The configuration of an SP service comprises two parts: a service provider and a service subscriber.

The service provider comprises parameters that are common to all service subscriber accounts from that service provider. Each service provider part is an ITSP profile that has its own parameter groups. You can configure as many as six ITSP Profiles (A through F) in a phone configuration.

The service subscriber comprises parameters that may vary for each specific subscriber account from the service provider. Every SP service user account requires a user name and usually a password for authentication. The service provider assigns an extension number or DID number to the user account. The assigned number may be the same as the user name of the account.

Select a Service Provider Profile

The service subscriber, known as an SP service, includes the `X_ServProvProfile` parameter that binds the SP service with an ITSP Profile.

By default, the `X_ServProvProfile` parameter for all SP services points to ITSP Profile A. If you want to use two different service providers with your phone, configure the settings for them in ITSP Profile A and ITSP Profile B respectively.

Note: A common mistake is not setting the `X_ServProvProfile` parameter correctly to point to the corresponding ITSP profile.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1$ to 6 services).
2. In the **Value** column for the `X_ServProv_Profile` parameter, select a service provider profile (A through F).
3. Select **Submit**.
4. Reboot your system when you complete your changes.

The OBiTALK Service

OBiTALK is a proprietary protocol for communications among VVX business IP phones, OBi Edition, and OBiTALK device management servers.

The protocol is intended for two main purposes:

- Peer-to-peer calling between OBi devices
- Device management by OBiTALK servers

Every OBi Edition phone comes with one instance of the OBiTALK service with the (fixed) factory-assigned 9-digit device OBi number as the user ID of the service. OBi devices can call each other by dialing the other party's OBi number.

The OBiTALK service also enables you to view and change the settings of your phones from the PDMS-SP OBiTALK portal. If you disable the OBiTALK service in your phone's configuration, you can't place OBiTALK voice calls or manage device features through the PDMS-SP OBiTALK portal.

Enable the OBiTALK Service

The OBiTALK service is enabled by default, unless disabled through ZT customization.

For information on ZT customization, see the *Poly VVX Business IP Phones, OBi Edition Provisioning Guide*.

Procedure

1. In the system web interface, go to **Voice Services > OBiTALK Service > OBiTALK Service Settings**.
2. In the **Value** column for the `Enable` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Limit OBiTALK Calls

You can limit OBiTALK calls to just the OBiHai echo server.

A simple way to disable OBiTALK voice calls completely is by setting the `MaxSessions` parameter to 0. If you do, however, you can't perform an echo test.

Procedure

1. In the system web interface, go to **Voice Services > OBiTALK Service > OBiTALK Service Settings**.
2. In the **Value** column for the `DigitMap` parameter, enter a value of (`<ob>222222222 | ob222222222`).

You can change or add more OBi numbers to this digit map by following the same pattern.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Voice Service Bridges

Your phone is also a voice service bridge (VSB) that can bridge calls across multiple supported voice services.

A *call bridge* connects two calls on the same or different voice services. Your phone allows four concurrent independent call bridges.

Trunks

Each supported voice service on your phone is referred to as a *trunk*, a physical wire or wires that deliver phone services to homes or businesses.

Each trunk is represented with 2-letter abbreviation and a 1-based instance identifier:

- SP1 = the SP1 Voice Service (with ITSP A, B, C, D, E, or F)
- SP2 = the SP2 Voice Service (with ITSP A, B, C, D, E, or F)
- SP3 = the SP3 Voice Service (with ITSP A, B, C, D, E, or F)
- SP4 = the SP4 Voice Service (with ITSP A, B, C, D, E, or F)
- SP5 = the SP5 Voice Service (with ITSP A, B, C, D, E, or F)
- SP6 = the SP6 Voice Service (with ITSP A, B, C, D, E, or F)
- PP1 = the OBiTALK Service

When configuring your phones, you can omit the instance identifier if it's equal to 1. For example, PP is equivalent to PP1. You use these short-hand notations heavily when configuring your phone, as they're found in call routes, call forward numbers, and speed dials parameters. Unless stated otherwise, the abbreviated trunk names are case insensitive.

Set the Trunk Capacity

Trunk capacity is the maximum number of simultaneous calls allowed on the trunk.

Set the value for each service using the `MaxSessions` parameter.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service > Calling Features** (where $N=1$ to 6 services).
2. In the **Value** column for the `MaxSessions` parameter, enter the maximum number of simultaneous sessions that the endpoint can conference.

The default value is 2 for all services. For PDMS-SP OBiTALK service, the maximum value is 4.

Note: For other SIP/SP services, don't set the value higher than the maximum number of simultaneous calls allowed by the service provider.

Trunk Groups

A trunk group is a group of trunks.

If a call is routed to a trunk group, your phone picks one of the available trunks from the group to make the call.

Availability of a trunk is based on the following criteria:

- Whether the trunk's digit map allows the number to call, AND
- Whether the trunk has capacity to make one more call

You can configure as many as four trunk groups on your phone.

Configure a Trunk Group

You can reference a trunk group and its associated digit map using the short name `TGn` and `(Mtgn)` and reference them in other digit maps and call routing rules so that the system can route calls to a particular trunk group.

Only trunks can be added to a trunk group. These include: PP, SP1 - SP6, VG1, VG2, ..., VG8, TG1, TG2, ... and TG4.

Note: A trunk group can include another trunk group (it can be recursive). However, you must make sure this doesn't result in infinite recursion.

Procedure

1. In the system web interface, go to **Voice Services > Gateways and Trunk Groups**.
2. In the **Value** column, configure the following parameters:

Parameter	Value
Enable	Select the check box to enable the use of the trunk group
Name	Enter a user-friendly name to describe the trunk group.
TrunkList	Enter a comma-separated list of trunks to include in this group.
DigitMap	Enter a digit map to direct calls to use this trunk group.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Voice Gateways

A gateway is another phone that enables two stage dialing.

Incoming OBiTALK callers call the gateway first with a normal OBiTALK call, get the auto attendant (AA), and then dial the target number. For authentication, the AA may ask the user to enter a PIN before establishing the second call.

A gateway is conceptually a trunk with its own digit map, and you can specify as many as eight gateways. Address each gateway using its factory-assigned OBi number and refer to a gateway and its associated digit map with the short trunk name *VGn* and *(Mvgn)*. You can use *VGn* and *(Mvgn)* in call routing rules and digit maps just like other real trunks.

Find more information on voice gateways in the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Configure a Gateway for Direct Dialing

You can configure a gateway with one-stage or direct dialing so that the caller can dial the target number directly without going through the AA.

Because a user can't enter a PIN when direct dialing, you can configure an optional user ID and password so the device can automatically authenticate with the gateway.

Procedure

1. In the system web interface, go to **Voice Services > Gateways and Trunk Groups**.
2. In the **Value** column, configure the following parameters:
 - **AuthUserID:** Enter a user ID
 - **AuthPassword:** Enter a password
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Call Settings

Topics:

- [Conference Calls](#)
- [Phone Number Formats](#)
- [Configure the Media Loopback Service](#)
- [Configuring Auto-Answer](#)
- [Configure Speed Dial Keys](#)
- [Caller ID](#)
- [Call Forwarding](#)
- [Configure Do Not Disturb](#)
- [Configuring Do Not Ring](#)
- [Configure Message Waiting Indicators](#)
- [Disable the Handset](#)
- [Configure the Speaker and Headset Buttons](#)
- [Display or Hide the Notification Icons](#)
- [Display or Hide the LTE Device Not Found Message](#)
- [Configure Multicast Page Groups](#)
- [Configure Music On Hold](#)
- [PTT Mode](#)
- [Premium Call Features](#)

You can configure call settings for your VVX business IP phones.

Conference Calls

The VVX business IP phones support users joining two or more parties into a conference call.

The phones support two methods of conference calls:

- Local mixing or bridging
- External conference bridge

Local Mixing or Bridging

After starting a 3-way conference, users can see the two remote parties both in the Connected state.

Note: The Opus audio codec doesn't support three-way calling when both legs of the call bridge are using Opus.

With locally mixed N-way calls, users can see N individual call items on the screen, one for each call-leg of the N-way call. Hence, you can also individually control each call-leg, such as holding or ending any one of them.

External Conference Bridges

When using an external conference bridge, the bridge limits the conference size.

Check with your service provider on the conference size limit.

Enable an External Conference Bridge

You can configure the conference bridge to enable call participants on the same SP service or using the same ITSP profile as the conference bridge to be added to conference calls.

For participants that are transferable, your system keeps them in the conference using local mixing. They're then subject to the local mixing limit.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. Enter the user ID or (SIP) URL of the external conference bridge in the `x_ConferenceBridge` parameter.
3. Go to **IP Phone > Phone Settings > Calling Features**
4. In the **Value** column for the `UseExternalConferenceBridge` parameter, select the check box to use an external conference bridge for conference calls.

The phone assumes that only conferees that are on the same SP service or using the same ITSP profile as the conference bridge can be referred to the bridge. For conferees that are referable, the phone keeps them in the conference using local mixing and are subject to the local mixing limit. For example, if you have a conferee who is connected through the OBiTALK service, the phone keeps the call with that conferee in the Connected State, as well as the call to the conference bridge in the Connected State, and applies local mixing to the two calls.

Phone Number Formats

There are places within your configuration that specify a target phone number, such as a speed dial number or a call forwarding number.

The following two formats specify a target phone number:

- Short number: The number itself, such as 3231234
- Full number: The number and the service to use with the number, such as sp3 (14089993312)

Use the case-insensitive service names for each service in the full number per the following:

- sp *n* for SP*n* Service for *n* = 1 to 6
- pp for the OBiTALK Service

When you only specify a short number, your phone determines the service to use, where necessary, by going through normal digit map and call routing processing. With a specified full number, your phone uses the number and service as specified without any modification.

Configure the Media Loopback Service

VVX business IP phones support the following media loopback features:

- Loopback modes: loopback-source and loopback-mirror
- Loopback types: rtp-media-loopback and rtp-packet-loopback
- Loopback packet formats: encapsrtp, loopbkprimer

For more information about Media Loopback Service, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
AcceptMediaLoopback	Select the check box to accept incoming call that requests media loopback.
MediaLoopbackAnswerDelay	Enter a delay in ms before answering a media loopback call. Default is 0.
MediaLoopbackMaxDuration	Enter a maximum duration in seconds to allow for an incoming media loopback call. Default is 0, which means the duration is unlimited.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Auto-Answer

Your phone supports two methods to auto-answer a call:

- Call-Info: The phone inserts an `answer-after=0` parameter in a Call-Info header in the INVITE request.
- Alert-Info: The phone inserts `info=alert-autoanswer;delay=0` parameters in an Alert-Info header in the INVITE request.

For more information on auto-answer and intercom, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Enable Auto-Answer

You can enable auto-answer using the `X_AutoAnswerMethod` parameter.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X** (where X=the ITSP profile for the service provider).

2. In the **Value** column for the `X_AutoAnswerMethod` parameter, select the signaling method you want to use to auto-answer.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Disable Auto-Answer

You can disable auto-answer with the `AutoAnswerEnable` parameter.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `AutoAnswerEnable` parameter, clear the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Auto-Answer Incoming Calls Based on Inbound Call Routing Rules

You can let the phone auto-answer certain incoming calls based on inbound call routing rules by specifying a rule that routes the call to `ph(autoans)`.

For example:

```
{someid:ph(autoans;nobeep)},{@.4089991234):ph(antoans;delay=2)},{ph}
```

The `autoans` syntax supports two attributes:

- `nobeep`: Don't play a beep tone on answering. The default is to play the beep according to the user preference settings.
- `delay={value in seconds}`: The number of seconds to ring before auto-answering the call. The default is 0.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service > Calling Features** (where $N=1$ to 6 services).
2. In the **Value** column for the `X_InboundCallRoute` parameter, enter a rule that routes the inbound call to `ph(autoans)`.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Auto-Answering Calls When a Caller Requests to Barge-In

You can configure your phones that use the OBiTALK service to enable a caller to barge-in (auto-answer to join) on an active call.

Many softswitches offer this feature by performing the audio mixing at the server end.

Configure the Star Code for Barge-In

Your phone offers the barge-in feature at the device end with the OBiTALK service (available for OBi-to-OBi calls only).

The calling phone requests to barge-in when you dial a star code before dialing the target OBi number. The called phone automatically answers and conferences with the calling phone, subject to resources available at that moment.

Note: The barge-in operation is only applicable if there is at least one active call on the called phone. Otherwise, the phone rejects the incoming call to barge-in.

Procedure

1. In the system web interface, go to **Star Codes > Star Code Profile *N*** (where *N* is A or B).
2. In the **Default** column, clear the check box for a parameter name with an empty value column.
3. In the **Value** column, enter a star code script in the following format: `code, name, action1`. For Code29, the star code script is `*96, Barge In, set($Bar1,1)`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Barge-In as a Coach

The calling phone to barge in may request to coach the called phone instead of fully participate in the active call.

Procedure

1. In the system web interface, go to **Star Codes > Star Code Profile *N*** (where *N* is A or B).
2. In the **Default** column, clear the check box for a parameter name with an empty value column.
3. In the **Value** column, enter a star code script in the following format: `code, name, action1`. For Code30, the star code script is `*99, Barge In, set($coach1,1)`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Speed Dial Keys

Configure one or more feature keys as speed dial keys by assigning the `Function` parameter.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function`, clear the check box.
4. In the **Value** column for the `Function` parameter, select **Speed Dial**.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Caller ID

During a call, the caller's name and number displays on the screen when available.

You can specify an Action URL in the Call-Info header to show caller information when a user presses the **ciurl** softkey (you must include this key in one of the applicable softkey sets). For example:

```
Call-Info: <https://abc.com/user/info.php?user=john.j.smith>;purpose=info
```

Configure a Caller ID Display Name

You can configure the name that displays when a user places a call.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service > Calling Features** (where $N=1$ to 6 services).
2. In the **Value** column for the `CallerIDName` parameter, enter a name for the caller that displays on the screen during a call.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Blocking Caller ID

Blocking caller ID (or anonymous call) enables users to prevent their name and number information from appearing when they make an outgoing call.

You can enable this is a feature for all calls on the phone or only for calls on a specific service. For calls that apply to an SP service, the feature may be offered locally by the phone or by the softswitch serving that SP service on the phone.

Block Caller ID for All Calls

You can block caller ID for all calls on the phone.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X = the ITSP profile for the service provider).
2. In the **Value** column, configure the following parameters:

Parameter	Description
<code>X_InsertRemotePartyID</code>	Select the check box to insert a SIP Remote-Party-ID header in all outbound INVITE.
<code>X_InsertPrivacyHdr</code>	Select the check box to insert a Privacy:id header in INVITE for anonymous calls.

Parameter	Description
X_UserAnonymousFROM	Select the check box to use <code>sip:anonymous@localhost</code> in the FROM header of INVITE to block outbound caller ID.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Block Caller ID for Calls on an SP Service

You can block caller ID for calls on an SP service with the `AnonymousCallEnable` and `Anonymous Call` parameters.

Procedure

1. In the system web interface, go to **Voice Services > SP n Service** (where $n = 1$ to 6 services).
2. Go to **Calling Features** and in the **Value** column for the `AnonymousCallEnable` parameter, select the check box to enable **Anonymous Call**.
3. Go to **Network Provided Services** and in the **Value** column for the `Anonymous Call` parameter, select the check box to enable the anonymous call service.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable the Block Caller ID Feature Key

You can configure the block caller ID function on a feature key with the `AnonymousCallEnable` parameter.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `AnonymousCallEnable` parameter, select the check box to enable **Anonymous Call**.
Enabling this option is equivalent to enabling the feature for all services.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Block Caller ID on a Line or Programmable Key

You can enable Block Caller ID on a line or programmable key.

Note: This service requires ITSP support.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where $n = \text{Left or Right}$).
 - Go to **IP Phone > Programmable Keys**.
- 2.
3. Select a key (for example, **Key 1** or **Key 2**).

4. In the **Default** column for the `Function` parameter, clear the check box.
5. In the **Value** column for the `Function` parameter, select **Block Caller ID**.
6. Select **Submit**.
7. Reboot your system when you complete your changes.

Enable the Block Anonymous Calls Feature Key

You can enable a feature key to block incoming calls that don't have an identifying caller ID number.

Incoming calls receive a busy signal or busy call treatment (such as **Call Forward On Busy**). This feature has both phone versions and line versions.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function` parameter, clear the check boxes.
4. In the **Value** column for the `Function` parameter, select `Block Anonymous Call`.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Call Forwarding

Call forwarding enables users to send incoming calls to another number.

Users can forward calls to a landline, VoIP, or OBiTALK number. Each line has one set of call forward settings, and there's one additional set at the phone level. Incoming calls on a particular line are processed by the call forwarding rules at that line level first, then at the phone level, whichever is applicable.

You can configure signaling options for call forwarding. If you don't use signaling, the phone tries to bridge the caller and forward to the target internally.

Configure Call Forwarding Settings

Configure Call Forwarding settings in the system web interface.

VVX business IP phones support three types of call forwarding:

- Call forward unconditional/Call forward all
- Call forward on busy
- Call forward on no answer

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile *X* > SIP**.
2. In the **Value** column for the `X_Use302ToCallForward` parameter, select the check box.
3. Go to **Voice Services > SP*n*Service > Calling Features**.
4. In the **Value** column, configure the following parameters:

Parameter	Description
<code>CallForwardUnconditionalEnable</code>	Select the check box to enable the Call Forward Unconditional feature on this line.
<code>CallForwardUnconditionalNumber</code>	Enter directory number to forward all incoming calls on this line.
<code>CallForwardOnBusyEnable</code>	Select the check box to enable call forwarding-on-busy on this device.
<code>CallForwardOnBusyNumber</code>	Enter a directory number to forward all incoming calls on this line when busy.
<code>CallForwardOnNoAnswerEnable</code>	Select the check box to enable call forwarding-on-no-answer on this device.
<code>CallForwardOnNoAnswerNumber</code>	Enter a directory number to forward all incoming calls on this line when no answer.
<code>CallForwardOnNoAnswerRingCount</code>	Enter number of rings to be considered as no answer.

5. Select **Submit**.
6. Reboot your system when you complete your changes.

Configure Call Forward Signaling

You can configure call forwarding using signaling options.

When using signaling for call forwarding, the phone attempts to forward the caller by replying a 302 response to the INVITE request with the forward target's SIP URL in a Contact header.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > General**.
2. In the **Value** column for the `SignalingProtocol` parameter, select **SIP**.
3. Go to **Service Providers > ITSP Profile X > SIP**.
4. In the **Value** column for the `X_Use302ToCallForward` parameter, select the check box to enable the use of the 302 response to INVITE for call forward.
5. Select **Submit**.
6. Reboot your system when you complete the changes.

Configure Do Not Disturb

Configure **Do Not Disturb** (DND) on the phone to forward calls directly to voicemail.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Calling Features**.
2. In the **Value** column for the `DoNotDisturbEnable` parameter, select the check box.
3. Go to **Network Provided Services**.

4. In the **Value** column for the `DoNotDisturb` parameter, select the check box.
5. Go to **IP Phone > Phone Settings > Calling Features > Phone Settings > Calling Features**.
6. In the **Value** column for the `DoNotDisturbEnable` parameter, select the check box.
7. Go to **User Preferences Settings**.
8. In the **Value** column for the `DoNotDisturbFeatureProvider` parameter, select the provider for the DND feature.
9. Select **Submit**.
10. Reboot your system when you complete your changes.

Configuring Do Not Ring

Enabling **Do Not Ring** disables the ringer, but the phone screen still indicates when an incoming call.

You can also assign a feature key to turn Do Not Ring on or off.

Enable Do Not Ring

Enable Do Not Ring using the system web interface.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `DoNotRingEnable` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure a Do Not Ring Feature Key

Define a feature with the Do Not Ring function using the `Function` parameter.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function` parameter, clear the check box.
4. In the **Value** column for the `Function` parameter, select **Do Not Ring**.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Configure Message Waiting Indicators

Visual and audio message waiting indicators notify the user when a new voice message is available.

Configure these settings in the system web interface.

For more information about message waiting indicators, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Calling Features**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
MWIEnable	Select the check box to enable Message Waiting Indication on the phone for this service.
X_VMWIEnable	Select the check box to enable Visual Message Waiting Indication on the phone for this service.
MessageWaiting	Select the check box to indicate if new messages are waiting.

3. Go to **Service Providers > ITSP Profile X > SIP**.
4. In the **Value** column, configure the following parameters:

Parameter	Description
X_MWISubscribe	Select the check box to enable SUBSCRIBE for MWI.
X_MWISubscribeExpires	Enter an interval (in seconds) for MWI subscription renewal.

5. Select **Submit**.
6. Reboot your system when you complete your changes.

Disable the Handset

You can use the `HandsetEnable` parameter to disable the handset when using speakerphone for all calls. The default setting is Enabled.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Default** column for the `HandsetEnable` parameter, clear the check box.
3. **Value** column, select one of the following options:

Option	Description
Enabled	Enables the handset for the phone.
Disabled	Disables the handset.

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure the Speaker and Headset Buttons

Configure the speaker and headset buttons to answer or end calls. The buttons are enabled by default.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `SpeakerKeyEnable` and `HeadsetKeyEnable` parameters, select one of the following options:

Option	Description
Enabled	Enables the speaker button or headset button. This is the default setting.
Disabled	Disables the speaker button or headset button.
Turnoffonly	Enables users to press the speaker button or headset button to end an incoming call.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Display or Hide the Notification Icons

When the admin disables the handset, the speaker button, or the headset button on VVX business IP phones, a notification icon is displayed at the top of the phone.

You can use the `ShowHandsetDisabledIcon`, the `ShowSpeakerDisabledIcon`, or the `ShowHeadsetDisabledIcon` parameters to display or hide the notification icons. The default setting for all three parameters is true (the icons are displayed).

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
<code>ShowHandsetDisabledIcon</code>	Select the check box to display the icon when the handset is disabled. The check box is selected by default. Clear the check box to hide the icon.
<code>ShowSpeakerDisabledIcon</code>	Select the check box to display the icon when the speaker button is disabled. The check box is selected by default. Clear the check box to hide the icon.

Parameter	Description
ShowHeadsetDisabledIcon	Select the check box to display the icon when the headset button is disabled. The check box is selected by default. Clear the check box to hide the icon.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Display or Hide the LTE Device Not Found Message

VVX 350 and VVX 450 business IP phones (OBi2 SKU only) can display a message when they don't detect an LTE device.

Use the `ShowLteNotFoundInSettingsNetwork` parameter to display or hide a message when the phones don't detect an LTE device. The default setting is `false` (the message doesn't display).

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `ShowLteNotFoundInSettingsNetwork` parameter, do one of the following:
 - Select the check box to display the message.
 - Clear the check box to hide the message.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Multicast Page Groups

Each phone supports two multicast groups that you can configure, and your phone supports up to 10 page groups.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Page Group *n*** (where *n* = 1 to 10 page groups).
2. In the **Value** column, configure the following parameters:

Parameter	Description
GroupName	A user-friendly name to label the group on the phone UI.
MulticastAddress	Must be a valid IPv4 Multicast Address. Default is 224.1.1.100 for both paging groups.
MulticastPort	Default is 65322 for page group 1 and 65324 for page group 2. Note that each group must use a different port number.

Parameter	Description
PushToTalk	A Boolean option to control whether the page group key (softkey or feature key) should act as a PTT key. That is, you can talk only while pressing+holding the key down. Default is false.
TTL	TTL value of outgoing (multicast) RTP packets. Default is 2.
AudioCodec	Audio codec to use for outgoing page. Default is G711U.
TxPacketSize	Outgoing RTP packetization in milliseconds. Default is 20.
RTCPtxInterval	Interval in milliseconds between sending outgoing RTCP packets when paging. No RTCP packets sent if value is 0, which is the default. Note that an RTCP Bye packet is always sent when ending an outgoing page regardless of this setting.
SilenceSuppression	A Boolean option to control if Silence Suppression should be used for outgoing page. Default is false.
ParticipantName	A string value to identify the current user who is paging to the group in outgoing RTCP packets.
PushToTalk	Select the check box to enable PTT.
PlayToneOnIncomingPage	A Boolean option to control whether to play a short Paging Tone before playing a new incoming page. Default is true.
PreferredAudioDevice	Select the preferred audio device to play incoming page. Enter either System, Headset, or Speaker. Default is System to let the phone picks according to other user preferences
PageTimeout	Limit of outbound page duration in number of seconds to avoid accidentally jamming the page group channel. The duration would be unlimited if the value is 0, which is also the default.
EnableJoinLeaveGroupUserControl	A Boolean option to control whether to enable you to join or leave a page group by pressing the pg1 through pg10 softkey. Default is true. When set to false, press the key once to talk and press it again to go back to listen. Although in that case you can't choose the leave the page group, you can still use the AutoAnswerIntercom option from the User Preferences menu not to play any of the incoming pages.
SpeakerVolume	Select the speaker volume when paging. 0 to follow current ringer volume.
Stats	A prepopulated field to show statistics of packets received in the past hour, minute, and 10 seconds. This field is updated every 10 seconds.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Music On Hold

When you enable this setting, this phone plays music to all callers that are on hold.

Note: You can't customize or configure the music that plays.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `MOHServiceNumber` parameter, enter the number to call to get music on hold.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

PTT Mode

The phone supports push-to-talk (PTT) mode with speed dial, Busy Lamp Field, and page groups 1 through 10 feature keys.

Configure PTT for Page Groups

Configure PTT mode with the feature key function page groups 1 through 10.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Page Group n** (where $n=1$ to 10).
2. In the **Value** column for the `PushToTalk` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure PTT for Speed Dial Numbers

Configure PTT mode on a speed dial by including the ptt flag in the `Number` parameter of the speed dial.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where n = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key that is designated as a Speed Dial key (for example, **Key 1** or **Key 2**).
3. In the **Value** column for the `Number` parameter, enter the ptt flag using the following syntax:

`Number = {target-number}[:,ptt][:send={digit-codes}]` where {target-number} can be an empty value if the number is unassigned, and {digit-codes} is a sequence of the following case-sensitive codes.

Codes	Description
0-9,*,#,a,b,c,d	The DTMF digit to send to the peer. Each digit is sent with 100 ms on and 100 ms off.
S	Pause for 3 seconds.
s	Pause for 1 seconds.
U" {prompt} "	Prompts you to enter one or more digits manually on the phone with the given {prompt} shown on the screen. Press the OK softkey to continue.
A	Wait for the called party to answer before continuing.

Note: If a voice service is specified for the speed dial feature key, if the Service field or the value of the Number field is a full number (one that includes voice service information), the phone doesn't apply a digit map before calling the speed dial. Any *codes in the number aren't processed, and the result may not be desirable. You can include the *codes to be processed by the phone by enclosing them in a pair of [...] at the beginning of the number field. For example: you may set the number field to [*96]SP1(2113). The phone then interprets *96 locally (to request auto-answer on the called party) and makes the call to 2113 using the SP1 service.

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Premium Call Features

VVX business IP phones provide various premium call features.

These features include:

- Busy Lamp Field (BLF)
- Managing call park, hold, and pickup
- Managing shared call appearances (SCA)
- Creating and managing buddy lists
- Recording calls
- Making emergency calls

Busy Lamp Field (BLF)

(SIP/SP only.)

Busy Lamp Field (BLF) is a common collaborative feature for users to monitor the extensions of other users from their phone.

When BLF is supported by a specific SP service on your phone, you can assign the BLF function to a feature key bound to that service to monitor an extension. A feature key assigned to the BLF function is referred to as a *BLF key*. One BLF key monitors one extension only. You can assign as many BLF keys as there are feature keys available.

Assign BLF to a Feature Key

When BLF is supported by a specific SP service on the phone, you can assign the BLF function to a feature key bound to that service.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Value** column for the `Function` parameter, select **Busy Lamp Field**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure the Speed Dial Number for BLF

You can assign a BLF key to act as a speed dial function key.

When using a BLF key as a speed dial key, your phone determines the number to call based on the following attributes (in order) specified in the `Number` parameter of the BLF key:

- If the optional `spd` attribute is specified, call that attribute.
- If the optional `ext` attribute is specified, call that attribute.
- Call the `{userid}` attribute, which you **MUST** specify for a BLF key.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select the key that you defined with the **BLF** function.
3. In the **Value** column for the `Number` parameter, enter the number, extension, or `userid` attribute associated with the assigned function.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Directed Call Pickup for BLF

Configure the method for Directed call pickup under the ITSP Profile of the SP service that is bound to the BLF key.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Configuration**.
2. In the **Value** column for the `X_DirectedCallPickupMethod` parameter, select one of the following methods

Parameter	Description
<code>Feature Code</code>	Your phone sends a normal INVITE to the number formed by concatenating the <code>Feature Code</code> with the number of the monitored extension (the <code>ext</code> attribute of the <code>Number</code> parameter (if it exists) or the <code>{userid}</code> attribute). Used with BroadSoft and FreePBX.
<code>INVITE+Replaces</code>	Your phone sends an INVITE with a <code>Replaces</code> header that identifies to the softswitch the ringing call to pick up from the monitored extension. Used with MetaSwitch.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Barge-In for BLF

Configure a barge-in feature code under the ITSP Profile of the SP service that is bound to the BLF key.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Codes**.
2. In the **Value** column for the `BargeIn` parameter, enter a feature code to barge in on an active call on another extension.

Your phone sends a normal INVITE to the number formed by concatenating the `Feature Code` with the number of the monitored extension (the `ext` attribute of the `Number` parameter (if it exists) or the `{userid}` attribute).

Used with BroadSoft and FreePBX.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Call Pickup for BLF

Configure a call pickup feature code under the ITSP Profile of the SP service that is bound to the BLF key.

Note: The resume operation is intended to resume (and take over) a holding call on the monitored extension. Currently, there's no known softswitch that supports this operation.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Codes**.
2. In the **Value** column for the `CallPickup` parameter, enter a feature code to retrieve a parked call from the ITSP.

Your phone sends a normal INVITE to the number formed by concatenating the `Feature Code` with the number of the monitored extension (the `ext` attribute of the `Number` parameter (if it exists) or the `{userid}` attribute).

Used with BroadSoft and FreePBX.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Floating BLF Key Assignment

You can reserve a block of BLF keys to members of a group of extensions without hard-coding the extension for each reserved key.

To reserve a BLF key to a member of the group, specify the `Number` parameter of a BLF key as: `Number = {group-name}/`

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select the key that you defined with the BLF function.
3. In the **Value** column for the `Number` parameter, assign the number to the group name using one of the following options:
 - `Number={group-name}/`: Specifies the group that the phone reserves the key for.
 - `Number={group-name}/{preferred-extension}?:` Specifies the preferred extension that the phone uses for the key within the group. The phone monitors the extension and uses it if it's returned by the server in a NOTIFY message. If the extension isn't returned, the phone can assign this BLF to another key as needed.

Example: `Number = sales-team/`

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure BLF with the OBiTALK Service

Configure BLF to work with the OBiTALK service with the `CallStatusAggregator` parameter.

The phone can monitor the following statuses:

- Offline

- Idle (no calls)
- Ringing (at least one call ringing)
- Holding (at least one call on hold)
- Connected (at one call connected)

The phone can apply the following operations to the monitored OBi number:

- Call (Call the monitored OBi number)
- (Directed) Pick Up (Pick up the oldest ringing call on the monitored phone)
- Barge-In (Barge in a connected call on the monitored phone)
- Coach (Coach the monitored phone on a connected call)

Procedure

1. In the system web interface, go to **Voice Services > OBiTALK Service > Calling Features**.
2. In the **Value** column for the `CallStatusAggregator` parameter, enter an OBi number that aggregates the call status for OBi devices.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Call Park and Call Pickup

Call park and call pickup/call retrieval are similar to call hold and call resume.

However, a user can pick up a call on extensions other than the one that parks (holds) the call.

You can set up a parking lot for calls using the softswitch. A call may be parked at an unoccupied slot from one extension and picked up by the same or another extension from the same slot.

Configure Call Park Methods

Your phone supports both call park and call pickup.

Set the method to use for call park independently for each SP service under the ITSP Profile bound to that SP service.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Configuration**.
2. In the **Value** column for the `X_CallParkMethod` parameter, select one of the following methods:

Method	Description
Feature Code	You must use this option for Park-Against-An-Extension. Specify the call park feature code under the same ITSP Profile, using the ITSP Profile > SIP > Feature Codes::Park parameter. In the context of parking a call, the local extension of the SP service is taken from the SPn Service::X_MyExtension parameter, if it is specified.
REFER	You must use this option for Park-In-An-Orbit.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Call Park Monitor and Call Pickup Methods

Configure your call park monitor and call pickup methods depending on how the call is parked.

When a call is parked-against-an-extension with a feature code, you can pick it up by dialing the call pickup feature code followed by the extension number the call is parked against. With the park-against-an-extension method, you can define a group of extensions to serve as a more generic parking lot and use BLF keys to monitor the call park status of each key. With the `spd` attribute of the `Number` parameter, these BLF keys include the call park feature code as a prefix to act as a shortcut to park a call in an empty slot.

Similarly when a call is parked-in-an-orbit and if the softswitch has a way to allow calls to be picked up from an orbit that is equivalent to making a call to a special number, users can make those calls without any special support from the phone. There are scenarios when your phone can monitor if certain extensions have calls parked against them, or if certain park orbits are holding calls.

Call Park Monitor Feature Keys

For calls parked-in-an-orbit, you can define a **Call Park Monitor** feature key to monitor each orbit (one feature key per orbit).

Your phone treats the Call Park Monitor function like the BLF function. The softswitch provides call park status information, as summarized in the following table.

Call Park Status Information

Call Park Monitor Status	Description	Available Operation	Icon	LED Pattern
One Call Ringing	A call is parked in this orbit	Pick up the call via Directed Call Pickup		Fast blinking red
No Calls	No call parked in this orbit	Park the call in this orbit		Off
Offline	Status of the park orbit is unknown	None		Steady amber

You configure a **Call Park Monitor** feature key like a BLF feature key, except that the `{userid}`, `{extension}`, and `{speed-dial}` attributes pertain to a park orbit instead of a real extension, and the `ptt` flag is not applicable.

When the user presses the **Call Park Monitor** key, one of the following applies:

- If there is no call parked, the highlighted call on the screen is parked.
- If there is a call parked, pick up the call.

Enable Call Park Status

Enable call park status using the system web interface.

For calls parked-against-an-extension, your phone may monitor call park status by directly (SIP) subscribing to an extension's call park status or indirectly through BLF monitoring of that extension.

Note: Your phone supports direct subscription to call park status of the underlying SP's extension, but not an arbitrary extension. If the underlying SP is a shared line, then the direct subscription to call park status of that extension isn't necessary if the softswitch already includes call park status in the notifications of the shared line status to the phone.

Procedure

1. In the system web interface, go to **Voice Services > SP n Service > Network Provided Services** (where $n = 1$ to 6).
2. In the **Value** column for the `CallParkStatus` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Call Park Ring

When you define one or more Call Park Monitor feature keys, you can enable Call Park Ring to play a short ring at regular intervals as an indication that at least one of the monitored call park orbits has a call parked.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Default** column of the `CallParkRingEnable` parameter, clear the check box.
3. In the **Value** column of the `CallParkRingEnable` parameter, select the check box.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Call Park Short Ringtones

Configure the short ringtones that play to indicate call park status in the system web interface.

Procedure

1. In the system web interface, go to **User Settings > User Preferences**.
2. In the **Value** column for the `CallParkRingtone` parameter, enter the string value with the following format: `{ring-tone-url},duration={d-value},interval={i-value}`

Attribute	Description
<code>{ring-tone-url}</code>	This value can be <code>ding</code> or <code>blorp</code> or an external URL that starts with <code>http://</code> or <code>https://</code>. For best results, limit this ringtone to 1 or 2 seconds long.
<code>{d-value}</code>	Duration parameter (optional) The total duration in seconds to play the ring, as long as there's a parked call. It must be greater than or equal to 0 (0 = to play indefinitely). Default value is 60.

Attribute	Description
{i-value}	Interval parameter (optional) The interval between the ring bursts in seconds. It must be greater than 0. Default value is 10.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Call Park Ring in the User Preferences Menu

Configure call park ring in the **User Preferences** menu in the system web interface.

Note: You can also configure the call park ring option so that the phone users can change the `CallParkRingEnable` option from **User Preferences** on the local interface of the phone. The default onscreen label for this option is **Call Park Ring**, and users can set the value to either **Enable** or **Disable**.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > GUI Menus**.
2. In the **Value** column for the `PreferenceMenu $n$` parameter (where $n = 1$ to 6), enter `cpr` in any of the **GUI menus**.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the Call Hold Notification Tone

You can configure your phone to play a notification tone when you have another call on hold.

The notification tone is defined using the `CallParkRingtone` parameter.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `CallHoldRingEnable` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Shared Line and Shared Call Appearances

(SIP/SP only) A shared line is a service account or extension installed on a group of phones.

All phones on the shared line are notified when a phone uses that extension. There can be multiple simultaneous calls on a shared line.

Each call on a shared line is shared call appearance (SCA), sometimes referred to as a bridged line appearance (BLA).

Related Links

[Configure Feature Keys](#) on page 84

Configure the SCA Method

There are two common implementations for SCA: the Call-Info method and the SLA/BLA (dialog) method. Configure the SCA method using the `X_ShareLineMethod` parameter.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Configuration**.
2. In the **Value** column for the `X_ShareLineMethod` parameter, select one of the following methods:

Method	Description
call-info	This is the method used by BroadSoft. The phone subscribes to the call-info event package with the proxy server to receive notification of share call appearance state updates. This method also uses the line-seize event package for seizing a SCA before making a call.
dialog;sla	This is based on the Bridged Line Appearance draft (draft-anil-sipping-bla-02). The phone accepts subscription to the dialog;sla event package from the state-agent, and also subscribes back the same with the state-agent. The phone and state agent then exchange notifications of share call appearance state updates.
dialog;ma	This is similar to the dialog;sla method, and is based on a more current version of the same draft (draft-anil-sipping-bla-04). The name of the event package name is changed to dialog;ma.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Designate an SP Service as a Shared Line

You can designate an SP service as a shared line with the `X_ShareLine` parameter.

You can use shared line when you want to give calling or answering rights to a set of phones under the same SP Line. For example, for businesses like a restaurant, when a customer calls in, all phones tied to that same SP line can answer the incoming call and at the same time, they can make a call using that same line.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Share Line Features**.
2. In the **Value** column of the `X_ShareLine` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Maximum Number of Call Appearances on a Shared Line

Set the maximum number of call appearances on a shared line with the `MaxSessions` parameter.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Calling Features**.

2. In the **Value** column of the `MaxSessions` parameter, enter the maximum number of simultaneous sessions that the endpoint can conference.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure a Call Appearance Feature Key

If your phone has a shared line configured as one of the services, you must have as many call keys defined on the phone as are bound to that service.

Procedure

1. In the system web interface, go to **IP Phone > Line Key Customization > Call Appearance**.
2. In the **Value** column for the `Enable` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Subscription Expires Values

SCA implementations are based on subscribe/notify framework.

You can configure the subscription expires value to the call-info and line-seize events.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Configuration**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
<code>X_CallInfoSubscribeExpires</code>	Enter <code>CallInfo</code> (SCA) subscription renewal (in seconds).
<code>X_LineSeizeSubscribeExpires</code>	Enter <code>Line-seize</code> event subscription renewal (in seconds).
<code>X_DialogSubscribesExpires</code>	Enter <code>DialogSubscribe</code> subscription renewal (in seconds).

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Buddy Lists

A buddy list is a contact list with presence information incorporated.

Your phone supports buddy lists based on the XMPP standard.

Configure a Buddy List Service

Enable a buddy list service for each SP service independently.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service Credentials**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
X_XmppDomain	Enter the XMPP server domain name. The phone resolves this name as DNS A Record only. For example, <code>impliop1.broadsoft.com</code> .
X_XmppUserName	Enter a user name for authentication to the XMPP server. For example, <code>ObihaiTester@impliop1.broadsoft.com</code> .
X_XmppPassword	Enter a password for authentication to the XMPP server.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable the Buddy List App

Enable the buddy list app for your phones using the following parameters.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
BuddyList	Select the check box to enable the buddy list app.
QueryVcard	Select the check box to enable the Vcard.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set up a Buddy List

Set up a buddy list using the following parameters.

The status values are available to all SP services and are stored as a comma-separated list of phrases in the `MyStatusHistory` parameter.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Buddy List Setup**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
MyPresence	Select the user selectable presence.
MyStatus	Enter the user selectable status value. Users can add or remove status values using the New and Remove softkeys.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure a Presence Monitor Feature Key

Configure a feature key using the Presence Monitor parameter to monitor the presence of a buddy in a buddy list.

Each presence monitor key monitors only one buddy.

You can configure as many presence monitor keys as you need, but you can't allocate the same buddy to more than one key.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Value** column, configure the following required parameters:

Parameter	Description
Function	Select the presence monitor function assigned to this key.
Service	Select the SP service where the XMPP service is offered, such as SP3.
Number	Enter the number, extension, or user ID of the buddy you want to monitor. Note: This value typically looks like an email address, such as user.name@somecompany.com .

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Emergency Call Settings

You can define one or more numbers as emergency numbers by adding the prefix EM# to those numbers using the `DigitMap` parameter and a corresponding rule with `OutboundCallRoute` to route those calls to a specific voice service to handle the call.

When the phone determines a call is to an emergency number, note the following:

- Users can't hold or end the call. Only the remote party can end the call.
- Users can start a call with a headset, but they can't switch to use a headset if they start the call using the handset or speakerphone. Users can only switch between the handset and speakerphone during the call.
- Users can't start or resume any other calls.
- Users can't press the **Home** key to get to the **Home** screen to start another app.
- Call waiting is disabled.
- All feature keys are disabled.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
DigitMap	Enter a digit map by adding the prefix EM# to those numbers. For example, DigitMap = (<EM#>911 other rules
OutboundCallRoute	Enter the routing rule for outbound calls made from this phone. For example, OutboundCallRoute = {(911):sp1},{(311):sp2}, other rules ...

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Emergency Call Webhook

You can add the capability to trigger a webhook that dials the chosen emergency provider when making an emergency phone call.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `EmergencyCallActionURL` parameter, enter an Action URL to launch when making an emergency call.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring System Settings

Topics:

- [Ringtones](#)
- [Configuring Star Codes](#)
- [Customizing Softkey Sets](#)
- [Feature Keys](#)
- [Disable the Calls App](#)
- [Set Audio Codec Priority](#)
- [Real-Time Voice Transcription](#)

Customize your VVX business IP phones using applications and various keysets.

Ringtones

Ringtones provide audible call progress indicators to the user.

Set a Custom Tone Pattern

You can create customized tone patterns to provide audible call progress indicators to the user.

Note: Tone Profile A default settings use North American telephone standards. Tone Profile B default settings use Australian telephone standards. You can download tone profiles for other countries from the OBiTALK forum.

For more information about tone patterns and tone profiles, see the *Poly OBi Device Technical Reference*.

Procedure

1. In the system web interface, go to **Tone Settings > Tone Profile A** or **Tone Settings > Tone Profile B**.
2. Select a tone pattern.
3. In the **Default** column, clear the check box for `TonePattern`.
4. In the **Value** column, enter the tone pattern you want to use.
For example tone patterns, see the *Poly OBi Device Technical Reference*.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Distinctive Ringtones and Patterns

Add Alert-Info to the SIP invite header to configure distinctive ringtones based on how you set up incoming call parameters.

In a SIP-based distinctive ring, the `RingName` is matched against the Alert-Info of the form `Alert-Info: <http://127.0.0.1/ring-name>`, where *ring-name* is one of the preloaded ringtones.

Set the Call Waiting Indicator Tone

Set a distinctive tone for incoming calls using the `X_DefaultRing` parameter.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1$ to 6 services).
2. In the **Value** column for the `X_DefaultRing` parameter, select the default ring pattern for the phone to use for incoming calls.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Assign a Default Ringtone

Assign a different default ringtone to each trunk on your phone.

An ITSP can instruct your phone which ringtone to use (by name) for a call routed to SP_n by inserting an Alert-Info header in the SIP INVITE sent to your phone. The Alert-Info must include a URI. For example:

Alert-Info: <http://www.xyz.com/some-folder/bellcore-dr4>

When your phone receives this instruction, it looks for a ringtone name in the ringtone profile that matches the Alert-Info URI. Ringtone names aren't case-sensitive when the phone compares them. If the phone finds a match, your phone plays the corresponding ringtone. Otherwise, your phone plays the default ringtone.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service** (where $N=1-6$).
2. In the **Value** column for the `X_RingtoneFile` parameter, enter the name of the default ringtone for the phone to use for incoming calls.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Star Codes

Star codes are short sequences of characters that users enter to perform certain operations.

Each sequence usually starts with the star (*) key followed by a two-digit code (such as *69).

A star code script contains a number of predefined variables and actions. Each variable represents one or a group of configuration parameters. An action can be checking or setting a variable's value, collecting a phone number, or calling a certain number.

Set the Star Code Profile

You can select a Star Code Profile (A, B, or None) for interpreting the star codes users enter on the phone.

You can select which star code profile to use with the `StarCodeProfile` parameter.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Calling Features**.
2. In the **Value** column for the `StarCodeProfile` parameter, choose one of the following star code profiles:
 - **A**
 - **B**
 - **None** (if you aren't using a star code)

A star code script is defined with the help of a number of predefined variables and actions. Each variable represents one or one group of configuration parameters. An action can be checking or setting the value of a variable, collecting a phone number from the user, or calling a certain number.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Program a Star Code

You can program star codes for any enabled feature, in addition to the default star codes available on VVX IP business phones.

Procedure

1. In the system web interface, go to **Star Codes > Star Code Profile N**.
2. In the **Default** column, clear the check box for a Parameter Name with an empty value column.
3. In the **Value** column, enter a star code script in the following format: `code, name, action1`
 For example, for Code30, the star code script is `*01, Page Group 1 Talk, pg1tx`. Using the star code `*01` sends a page to Group 1.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Customizing Softkey Sets

The set of softkeys available on the phone screen at any given time is a softkey set. The current app state determines which softkey set displays on the phone.

You can customize some of the softkey sets by entering a list of softkey specifications as a comma-separated list in the corresponding softkey parameters.

The value of a softkey set parameter uses the following general format: `{softkey-spec} [{softkey-spec}] , {softkey-spec} [{softkey-spec}]` .

A `{softkey-spec}` is a softkey specification and the `|{softkey-spec}` syntax lets you specify an optional alternative softkey to show when the given softkey is hidden. A softkey is hidden when its hidden condition is matched. The hidden condition for each softkey is defined internally.

Find more information on softkey specifications in the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Assign Softkeys

You can assign softkeys to the configurable soft key sets.

Procedure

1. In the system web interface, go to **IP Phone > Soft Keys > Soft Key Sets**.
2. In the **Value** column, assign softkeys for the following parameters:

Parameter	Description	Softkey Examples
Home	This parameter represents the Home screen.	redial,cfa,dnd,missed lines
SCAInUse	SCA is being used by another phone.	sca.barge sca.monitor sca.whisper newcall
CallParked	Call is parked.	pickup,,,newcall
CallConnected	Call is connected.	end,hold,tousb,tospk,conf,transfer,privhold,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume
CallHolding	Call is holding.	end,resume,add2conf,conf,transfer,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume
Dialtone	Dial tone playing.	redial,phbk,lines,mode
Dialing	Dialing a number.	redial,dial,backspace,mode
OnDialing	Dialing a number while on-hook.	switch.line,dial,backspace,mode
Ringing	Incoming call is ringing.	answer,reject
CallConnecting	Trying outgoing or called party is ringing.	end,,,newcall
ConfTrying	Calling second conferee.	end
ConfRinging	Second conferee ringing.	end,,conf.now
ConfConnected	Connected with second conferee.	end,hold,resume,conf.now

Parameter	Description	Softkey Examples
ConfBridgeControl	Connected with an external conference bridge.	end,deaf,mute
Conferencing	Merged conference call item.	end,split,newcall
XferTrying	Calling transfer trying.	end
XferRinging	Transfer target ringing.	end,,xfer.now
XferConnected	Connected with transfer target.	end,hold,resume,xfer.now
NetServices	Network Services top level.	ns.bci,ns.dnd,ns.cfa,ns.bac,ns.acd
BLFCall	BLF call options.	blf.call,blf.answer,blf.pickup,blf.barge,blf.coach,blf.monitor
Settings	This represents the Settings menu.	reboot,reset,fwupdate

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define the Softkey Style

All softkeys support the optional label and icon attribute, the text and icon that shows on the screen inside the softkey window. In addition, all softkeys support the optional style attribute that specifies how to style the icons and labels for the softkey.

The value of the style attribute is equal to the ID attribute of the `<style>` elements in the `<SoftKeyStyles>` XML specified in the `SoftKeyStyles` parameter.

Procedure

1. In the system web interface, go to **IP Phone > Screen Item Customization**.
2. In the **Value** column for the `SoftKeyStyles` parameter, clear the check box.
3. Specify any of the following aspects for the softkey styles:
 - Softkey background picture when the key is in normal state, pressed down or highlighted state, and alert state. You can set the picture as an internal picture file or an http or https URL.
 - The placement of the label and icon within the softkey window (left/right/center/top/bottom), which you can adjust with x and y offset.
 - The color, size, and font for the label.
 - The height and width of the softkey icon and label.

Note: If you don't specify the style attribute or it's not found, the `<style>` element with `id=default` is applied to the softkey. Otherwise, the softkey is styled with the internal default style.

Feature Keys

You can configure feature keys on your phone or VVX EM50 expansion module to perform predefined functions.

You can configure side car keys, virtual line keys, and programmable keys.

You can define multiple feature keys with the same function, such as **Call Appearance** and **Speed Dial**. However, don't assign a monitor function to the same entity using more than one key. For example, don't assign more than one BLF key to monitor the same extension or assign more than one **Message Status** key to monitor the same mailbox. The phone can update only one of the keys when the status of the monitored entity changes.

A call feature key or call key is a virtual line key (VLK) with the call function assigned. Each call carried out on the phone must have an assigned call key, and you must have at least one call key to make or receive a call. Each call key can hold exactly one call, so Poly recommends that you define at least a few call keys on the phone to handle multiple call scenarios, such as call waiting and conferencing.

Assign Feature Keys

You can assign a function to a feature key bound to that service.

For more information on the specific values for each feature key, see the *Poly OBi Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > *n* Line Keys** (where *n* = Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. In the **Value** column, configure the following parameters:

Parameter	Description	Notes
Function	Select the function to assign to this feature key.	

Parameter	Description	Notes
Service	The service to bind the key to.	Required for: • ACD Sign On or Off • Busy Lamp Field • Call Park Monitor • Disposition Code • Hoteling • Exec Filter On or Off • Exec Assistant • Line Monitor • Message Status • Presence Monitor • Security Class Optional for: • Block Anonymous Call • Block Caller ID • Call Appearance • Call Forward • Do Not Disturb • Transfer • Speed Dial
Number	The number, extension, or user ID of the entity associated with the assigned function.	Required for: • BLF • Call Park Monitor • Presence Monitor • Speed Dial Optional for: • Transfer • Blind Transfer
Name	A nickname used to refer to the entity specified in the <code>Number</code> parameter.	Optional for all functions.
MaxCalls	Maximum number of calls to overload on the key.	This is only applicable if the function is Call Appearance. This parameter isn't available under Programmable Keys and Side Car Keys.
Group	A short name referencing the Line Key Customization Group to use to customize the layout of this Line Key.	

Configure Feature Keys

You can configure each feature key to perform various functions.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > X Line Keys** (where X=Left or Right).
 - Go to **IP Phone > Side Car n** (where n=1 or 2).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Value** column, configure the following parameters:

Feature Key Function	Description
Call Appearance	Makes or receives one call, and the key is known as a call key in this case. You must have an unused (idle) call key available to make or receive a new call. The phone administrator should allocate as many call keys on the phone as the maximum number of concurrent calls it's expected to handle. The VLKW (Virtual Line Key Window) shows nothing but the idle phone icon (shown on the right column) when no active call is assigned to the key. Otherwise, the icon changes to reflect the current call state (Call States and Call State Icons are described in the Section Making and Receiving Calls and VLKW shows call information, if available. VLKW may also change its background color to further reflect the current state. A call key may be bound to a voice service and is called a bound call key in that case (otherwise it's known as an unbound call key). A bound call key is used to make/receive calls on the bound voice service only. This is one of the ways for a user to select a specific line to make a call. A call key may be bound to a service that is a Shared line. In that case when no call is on that key, the VLKW information reflects the state of the respective Shared Call Appearance (SCA). Parameters: • Service: Optional. The service or line with which to bind the key • MaxCalls: Number of calls managed by this call key

Feature Key Function	Description
Line Monitor	Monitor a Line (a voice service installed on the phone). The Line events that are monitored include: • Idle: no calls • Ringing: at least one incoming call • In Use: at least one active call • Holding: at least one call holding VLKW shows the monitored service name and account username (usually same as the account DID number or extension). This function must be bound to the specific voice service that it monitors. Parameters: • Service: Required. The service or line to monitor
BLF (Busy Lamp Field)	Monitor the call state of another extension. A BLF key must be bound to a service (as configured by the phone administrator). The call events that are monitored include: • Ringing: at least one incoming call • Holding: at least one call holding • Busy: at least one active call • Idle: no calls • Call parked (against the monitored extension) VLKW shows the bound service name and the monitored extension (or DID number, or account username). This function must be bound to a specific voice service. Parameters: • Service: Required. The service that provides the monitoring function • Number: Required. The extension (on the specified service) to monitor. It may contain multiple attributes ◦ ptt - (no value). Make the call a Push-to-Talk when calling the monitored extension ◦ spd - An alternative number to call when calling the monitored extension ◦ bx - (no value). Enable one-touch blind transfer behavior

Feature Key Function	Description
Call Park Monitor	Monitor the call park status of a single orbit in a parking lot. Per default LED setting, the key's LED is turned off when no call is parked in that orbit, or otherwise solid red. When the orbit isn't occupied, press the CPM key once to park the highlighted call on screen onto that orbit (the call must be in the Connected or Holding state to be parked). When the orbit is occupied, press the CPM key to retrieve the call as an additional call on the current phone. Parameters: • Service: Required. The service that provides the parking lot orbit: parking and monitoring function • Number: Required. The parking lot orbit to park and monitor
Presence Monitor	Monitor the presence/status of one buddy in a Buddy List. It also serves as speed dial to that buddy. Parameters: • Service: Required. The service that provides the XMPP service for this function • Number: Required. The JID of the buddy in the buddy list to monitor. Only the user id portion of the JID is needed. You may specify at the end such that a partial match of the JID is enough. For example, if the JID of the buddy is: abcd-12345@gmail.com, you may specify <code>abcd</code> for the Number field to monitor this buddy, provided there's no other JID in the buddy list that starts with <code>abcd</code>.

Feature Key Function	Description
Speed dial	If a number has not been assigned, VLKW shows no textual information. Otherwise it shows the speed dial number configured and, if the speed dial is bound to a service, the name of the service that it's bound to. If the speed dial has a display name configured, the VLIW shows the display name (such as John Seymour) instead of the assigned number. Parameters: Service: Optional. The suggested service the make the call withNumber: Required. The number to call. It may include the following attributes: • ptt (with no value): make the key a Push-To-Talk key. • bx - (with no value). Enable one-touch blind transfer behavior • send={digit-codes} to automatically dial some digits after the call, where {digit-codes} is a sequence of the following case-sensitive codes: ◦ 0-9,*,#,a,b,c,d - The DTMF digit to send to the peer. Each digit is sent with 100 ms on and 100 ms off. ◦ S - Pause for 3 seconds. ◦ s - Pause for 1 second. ◦ U"{prompt}" - Prompt the user to enter one more digit manually on the phone with the given {prompt} shown on the screen, then press OK to continue. ◦ A - Wait for the called party to answer before continuing. Note that the phones starting executing the first code in {digits} when the call receives early media or when the call is answered, whichever happens first. For example: send=Ass1234U"Enter Passcode"5678 It's acceptable to have the send attribute specified in a speed dial without a number (for example, Number=;send=1234#). In this case, the speed dial can be used to send out the list of digits on a connected call. Note that if a voice service is specified for the speed dial either in the Service Field or the given number is a full number (one that includes voice service information), the phone doesn't apply digit map before calling the speed dial. Hence any *codes in the number aren't processed and the result may not be desirable. You can include the *codes that the phone needs to process by enclosing them in a pair of [...] at the beginning of the number field. For example: you may set the number field to [*96]SP1(2113);ptt. The phone then interprets *96 locally (to request auto-answer on the called party) and makes the call to 2113 using SP1 service.

Feature Key Function	Description
Speed dial (continued)	Using Speed Dial to store a Feature Access Code prefix In many applications, the Service Provider can perform a function by calling a feature access code prefix followed by a target number. For example, the prefix *48 followed by the target extension such as 1002 may invoke from the Service Provider the call-monitoring function on extension 1002. You can just dial *481002 directly. You can also store *48 in a speed dial and label it as Monitor on the system web interface. To mark the speed dial as a prefix, you append 2 or more dots to the number, such as *48... When user presses the Monitor speed dial, the phone shows a dial box on screen with *48 entered. Then you can continue to dial the target extension, or press another speed dial or BLF that is the target extension to monitor.
Do Not Disturb	Turn the Do Not Disturb feature on or off. If the function is bound to a specific voice service, it's applied to incoming calls on that service only. Otherwise, it's applied system wide to all incoming calls no matter which service the calls come from. Parameters: Service: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls the phone
Do Not Ring	Turn the Do Not Ring feature on or off. When the feature is enabled, incoming calls come through like normally but phone doesn't play audible ring (call waiting tone is played during call waiting). This function can't bound to any specific voice service. It's applied system-wide to all incoming calls no matter which service the calls come from. Parameters: None
Block Anonymous Callers	Turn the Block Anonymous Caller feature on or off. If the function is bound to a specific service, it's applied to incoming calls on that service only. Otherwise, it's applied system wide to all incoming calls no matter which service the calls come from. If this feature is enabled, the phone rejects all incoming calls with Caller ID (name/number) hidden (blocked). Parameters: Service: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.
Block Outgoing Caller ID	Turn the Block Caller ID feature on or off. If the function is bound to a specific service, it's applied to outgoing calls on that service only. Otherwise, it's applied system wide to all outgoing calls no matter which service is used for the calls. If this feature is turned on, the phone attempts to hide user's caller ID on outbound calls so the called party can't see who's calling. Parameters: Service: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.

Feature Key Function	Description
Call Forward All	Turn the Call Forward All Calls feature on or off. If the function is bound to a specific service, it's applied to incoming calls on that service only. Otherwise, it's applied system wide to all incoming calls no matter which service the calls come from. Parameters: • Service: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.
Auto Answer	Turn the Auto Answer (Intercom Calls) feature on or off. Normally this feature is turned on so that incoming intercom calls are answered automatically by the phone on the speakerphone or headset. If this feature is turned off, incoming intercom calls are treated as regular calls and the phone rings normally. This function cannot be bound to any specific voice service. It's applied system-wide to all incoming calls, no matter which service the calls come from. Parameters: None
Call Waiting	Turn the Call Waiting feature on or off. Normally, this feature is enabled and you can accept more incoming calls while already on a call. If this feature is turned off, all incoming calls are rejected as busy when you're on a call. This function cannot be bound to any specific voice service. Parameters: None
Message Status (Monitor Voicemail Status)	Monitor the number of messages in a mailbox. The function must be bound to an SP service that has the MWI (Message Waiting Indication) feature enabled. The VLKW shows if and how many new messages are available in the mailbox. Parameters: • Service: Required. The SP service that provides the voicemail service. • Number: Optional. The number to call (to check voicemail). It may include the following attributes: ◦ mbox={mailbox-id} where {mailbox-id} is third-party mailbox to monitor, if it isn't the same as the main mailbox for the given SP Service. Don't specify this attribute if you're monitoring the main mailbox for the given SP Service
Hold	Hold all calls that are in the Connected State. VLKW shows how many calls are currently in Holding State. The LED turns red if there's a least one call in the Holding State, or turned off otherwise. This function cannot be bound to any specific voice service. Parameters: None

Feature Key Function	Description
Add to Conference	Add all calls that are in the Holding State to the current conversation. VLKW shows how many calls are in the Holding State. The LED turns red if there is at least one call in the Holding state, or turned off otherwise. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 1	Join/Leave Page Group 1. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 2	Join/Leave Page Group 2. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 3	Join/Leave Page Group 3. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 4	Join/Leave Page Group 4. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None

Feature Key Function	Description
Join/Leave Page Group 5	Join/Leave Page Group 5. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 6	Join/Leave Page Group 6. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 7	Join/Leave Page Group 7. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 8	Join/Leave Page Group 8. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Join/Leave Page Group 9	Join/Leave Page Group 9. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None

Feature Key Function	Description
Join/Leave Page Group 10	Join/Leave Page Group 10. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function cannot be bound to any specific voice service. Parameters: None
Change ACD Agent State	Change/monitor an ACD (or Call-Center) Agent State to one of the following values: • Available (to take new calls) • Unavailable (to take new calls) • Signed Off • Wrapping Up (the last call) ACD stands for Automated Call Distribution and is the primary way a call-center distributes calls among a number of agents working for the call-center. The ACD controller should only send incoming call to an agent whose state is "Available". Note that an agent can't change state to "Signed Off" or "Wrapping Up" directly by pressing the feature key. To change to these states, an agent must use the corresponding feature key menu item from the UI (invoked by pressing and holding down the feature key), or some other means provided by the softswitch, such as a web portal. This function must be bound to a specific voice service. The ACD agent handles calls on the bound service only with respect to the underlying Call Center. The Call Center isn't aware of calls the agent makes or receives with other voice services installed on the phone. Parameters: • Service: Required. The service that provides the ACD Agent function.
Guest User Login/Logout	This feature is also known as Hoteling on some softswitches. The phone may be set up to be used temporarily by a guest, such as a visiting employee or temp worker. The guest can press the key and enter a user-id and password to log in and start using the guest phone with a personal extension temporarily (until the guest logs out or the server logs the user out remotely). This function must be bound to a specific voice service (that supports this feature). Parameters: • Service: Required. The service that provides the Guest Login function.

Feature Key Function	Description
Enter Disposition Code for the last call.	Enter a Disposition Code for the last call. This is used by a Call Center agent to enter a disposition code for the last customer call. This function must be bound to a specific voice service (that supports this feature). Parameters: Service: Required. The service that provides the Disposition Code function
Next Tab	Press this key to switch to the next VLK Tab or cycle back to VLK Tab 1. This function can't bound to any specific voice service. Parameters: None Note that the multiple Line Key Tabs feature can be disabled also by disabling the option User Preferences::LineKeyTabs. In that case the "Next Tab" function does nothing.
Transfer	Invoke the call transfer function on the currently highlighted call on the screen when the Calls App is at the top of the display stack. The call must be in a transferable state, that is, Holding or Connected State. Parameters: Service: Optional. The suggested service to use to make the call to the transfer target, if the Number parameter is also specified Number: Optional. The number of the transfer target to call. User aren't prompted to enter the transfer target number if this parameter is specified
Blind Transfer	Invoke the blind call transfer function on the currently highlighted call on the screen when the Calls App is at the top of the display stack. The call must be in a transferable state, that is, Holding or Connected State. Parameters: Number: Optional. The number of the transfer target to transfer the call to. Users aren't prompted to enter the transfer target number if this parameter is specified
Blind Transfer 2	Same as the Transfer function, collecting the target number from the user, but a blind transfer is performed after the target number is collected. In contrast to Blind Transfer, Blind Transfer 2 plays dial tone, applies digit map, and would time out as user enters the target number.

Feature Key Function	Description
Action URL	Launch an OBiPhoneXML application at the given URL Parameters: Number: Required. The URL of the OBiPhoneXML application to run Name: A name to identify the application on the phone screen

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Related Links

[Shared Line and Shared Call Appearances](#) on page 71

Preassigned Feature Keys (PFKs)

VVX business IP phones have no dedicated programmable keys. Instead, they have hard keys wired to the most commonly used features with function-specific icons printed on the physical keys.

Theoretically, though not recommended, you can reprogram these hard keys to provide other functions, like the programmable keys on the phones.

The default (and recommended) functions of these hard keys are summarized in the following table.

Name	Default Function	Description
Transfer (all phones)	Transfer	Two valid alternatives to this behavior are to change Function to Blind Transfer or Blind Transfer 2 when pressing the key would provide corresponding behavior, respectively.
Messages (VVX 350 and VVX 450 only)	Message Status	Set up the Number and Service parameters of this key to access voicemail.
Hold (all phones)	Hold	Hold a call.

Configure Call Keys

Configure a call key to a specific voice service account installed on the phone or unbound to any service.

You can configure a bound call key to handle calls on the bound service account only, while an unbound call key can handle calls on any service account.

Procedure

1. In the system web interface, go to **IP Phone > X Line Keys** (where X = Left or Right).
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Value** column, configure the following parameters:

Parameter	Description
Function	Select a function to assign to this key.
Service	Select a voice service to apply to the assigned function.
Number	Enter the number, extension, or user ID of the entity associated with the assigned function.
Name	Enter a display name for the entity associated with the assigned function.
Group	Enter the ID of the Custom Line Key Group to use to format the contents of the line key window.
Style	Enter the ID of the Custom Line Key Style to use to format the contents of the line key window.
MaxCalls	Select the maximum number of calls this key should handle.

4. Select **Submit**.
5. Reboot your system when you complete your changes.

Disable Line Key Tabs

By default, the line keys area to the right of the screen has four tabs of virtual line keys (VLKs), and only one tab is visible at any time.

You can disable this feature using the option **LineKeyTabs**.

When disabled, the phone does the following:

- Displays no line key tabs in the status bar
- Displays only one page of VLKs for supported phones:
 - None on the VVX 150
 - VLK 1 to 4 on the VVX 250
 - VLK 1 to 6 on the VVX 350 and VVX 450
- Displays status icons in the right corner

Procedure

1. In the system web interface, go to **User Settings > User Preferences**.
2. In the **Value** column for the `LineKeyTabs` parameter, clear the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Disable the Calls App

Normally, calls display together on the **Calls App** screen.

You can disable this option with the `PackCallsOnDisplay` parameter, which causes the phone to display only the calls belonging to the same call key.

Procedure

1. In the system web interface, go to **User Settings > User Preferences**.
2. In the **Value** column for the `PackCallsOnDisplay` parameter, clear the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Audio Codec Priority

You can select a codec as the preferred codec for a profile by setting the priority of that codec to be highest among all the enabled codecs in the profile.

You can assign each of the SP and OBiTALK services to a codec profile in its corresponding configuration. For codecs with the same priority setting, the codec that appears first on the codec profile webpage has a higher priority.

Procedure

1. In the system web interface, go to **Codecs > Codec Profile A or Codec Profile B**.
2. Select the codec you want to set the priority for.
3. In the **Value** column for the `Priority` parameter, enter a priority (1 is the highest).
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Real-Time Voice Transcription

VVX business IP phones, OBi Edition, can stream live phone audio to your server-based natural language processing (NLP) service.

Depending on your NLP service, you can save phone recordings for training or quality control, produce live conversation analytics data, and monitor live conversation transcriptions.

Set Up Real-Time Voice Transcription

Before you can enable real-time voice transcription, you must first set it up in your environment.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Transcription Service**.
2. Configure the following parameters:

Parameter	Description
URL	Transcription service URL.
Key	Transcription service authentication key. Your transcription service provider provides the authentication key.
Secret	Secret for authentication. Your transcription service provides the secret.
Config	Configuration string.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Real-Time Voice Transcription

Once you set up real-time voice transcription in your environment, you can enable it for all available lines and OBiTALK.

Procedure

1. In the system web interface, go to **Voice Services**.
2. Select the service provider lines (SP 1 to 6 and OBiTALK Service) that you want to use with real-time voice transcription.
3. For each service provider you want to use with real-time voice transcription, go to **Calling Features** and select the check box for the `X_TranscriptionService` parameter.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring BroadSoft Server Features

Topics:

- [Built-In BroadSoft Phone Applications](#)
- [Enable Call Recording Controls](#)
- [BroadSoft Call Center Features](#)
- [BroadSoft Guest Login/Logout \(Hoteling\)](#)
- [BroadSoft AS-Feature-Event](#)
- [Configure BroadSoft XSI Features](#)

You can configure VVX business IP phones, OBi edition systems with BroadSoft server options.

Built-In BroadSoft Phone Applications

The **Net Services** and the **Net Directory** built-in phone apps rely on the BroadSoft/BroadWorks switch.

Built-In BroadSoft Phone Applications(continued)

Display Caption (Default)	Action URL	Summary
Net Services	Phone://netsrv	A collection of settings of features the service provider offers. The settings organize on a per SP Service basis. This App is only applicable with a BroadSoft softswitch.
Net Dir	Phone://netdir	A directory service offered by a service provider, either BroadSoft directory or LDAP.

Configure the Net Services Menu

BroadSoft offers a set of user features that you can access through a few APIs that correspond to an SP service on the phone.

You can find these features under **Network Services per SP** service on the phone, and the settings under the **Net Services** app on the **Main Menu**.

The top level of the app shows a list of the SP services that have network features enabled and the status of the enabled features:

- DND On or Off
- Call Forward All On or Off
- ACD Agent State
- Anonymous Call On or Off

You can configure the softkey set on this screen with multiple options:

- BCI

- DND
- CFA
- ACD

Select the SP service to view or change all the available settings for the SP service.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > GUI Menu**.
2. In the **Default** column for the `NetServicesMenu1` parameter, clear the check box.
3. In the **Value** column for the `NetServicesMenu1` parameter, enter a comma-separated list of user features.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Call Recording Controls

You can enable the call recording feature on a per SP service basis using the `CallRecording` parameter.

Your phone supports the call recording functions available with a BroadSoft application server. During a call, your phone provides the controls for call recording.

The softkey options for recording controls options are available only when the call is in the Connected or Holding state.

Procedure

1. In the system web interface, go to **Voice Services > SP n Service > Network Provided Services**.
2. In the **Value** column for the `CallRecording` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

BroadSoft Call Center Features

This suite of features supports call center applications with a BroadSoft softswitch.

Enable Disposition Code

An agent can enter a disposition code for the current call that is still ongoing or for the last call that has just ended.

For the first case, the agent selects the **Dispose Code** softkey that is available when the call is in the connected state. The agent then enters the code and submits it while talking to the caller.

For the latter case, the agent presses the feature key that has been assigned the **Disposition Code** function right after the call; then, enters and submits the code.

To use this feature on the phone, you must enable the option via the `DispositionCode` parameter. The option only applies to calls on the same SP service.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. In the **Value** column for the `DispositionCode` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Customer Originated Call Trace

The agent can start a call trace during the call by pressing the **Trace** softkey when the call is in the Connected state.

The actual call trace function completes on the softswitch after the user launches it from the phone.

Enable Customer Originated Call Trace

To use the call trace function on the phone, you must enable the option `CallTrace` parameter.

The option only applies to calls on the same SP service.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `CallTrace` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define Speed Dial Function Key for Call Trace

You can define a speed dial function key for call trace with the following parameters.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where n =Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function`, `Number`, and `Service` parameters, clear the check boxes.
4. In the **Value** column, configure the following parameters:

Parameter	Description
Function	Select Speed Dial.
Number	Enter customer-originated-trace.
Service	Enter the Voice Service to invoke the Call Trace operation, such as SP3.

5. Select **Submit**.
6. Reboot your system when you complete your changes.

Enable the Escalation Feature

To use the escalation feature on the phone, you must enable the option via the `Escalation` parameter.

The option only applies to calls on the same SP service.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `Escalation` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable the Call Center Feature

When the softswitch sends an incoming call to the phone, it may include some basic information about the call center where the call is coming from.

The phone displays the information, if available, with the Ring Alert message for the incoming call. You can't configure this behavior.

The phone can display the following information:

- Call Center name
- Call Center user ID
- Average waiting time
- Number of calls in the queue

The phone subscribes to the status of the call center that an SP service belongs to. Users can view the call center status information at any time on the phone via the **Net Services** app from the phone's main menu.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. In the **Value** column for the `CallCenter` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

BroadSoft Guest Login/Logout (Hoteling)

(SIP only.)

This feature is also known as hoteling (consult BroadSoft documentation on how to administer this feature on the server). The phone may be set up to be used temporarily by a guest, such as a visiting employee or temp worker in a hot desk environment.

Enable the Hoteling Feature

To use hoteling on the phone, you must first enable the feature.

After the first successful registration with the SIP proxy server, the phone starts a subscription to the **x-broadworks-hoteling** event package in the context of an SP service.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. In the **Value** column for the `Hoteling` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define a Hoteling Function Key

You can define a feature key with the function **Hoteling** (but no more than one hoteling feature key per SP service).

The LED and the icon of the key reflect the current guest login state.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where n =Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function` parameter, clear the check boxes.
4. In the **Value** column for the `Function` parameter, select **Hoteling**.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Configure the Expires Value of the Subscription

You can configure the expires value of this subscription using the `X_BWHotelingSubscribeExpires` parameter.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP > Feature Configuration** (where X =the ITSP profile for the service provider).
2. In the **Value** column for the `X_BWHotelingSubscribeExpires` parameter, enter the interval (in seconds) for when the renewal subscription expires.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

BroadSoft AS-Feature-Event

The AS-Feature is a collection of features available on a BroadSoft application server.

For more information on parameters, see the appropriate data model for your phone:

- [VVX 150 Data Model](#)
- [VVX 250 Data Model](#)
- [VVX 350 Data Model](#)
- [VVX 450 Data Model](#)

Activate BroadSoft AS-Features

You can activate the BroadSoft AS-Feature set to access features available from your third-party server.

Activating the BroadSoft AS-Feature lets you use the following network-provided services:

- Call Park
- Call Forward Always
- Call Forward Busy
- Call Forward No Answer
- Network Directory
- Do Not Disturb

Note: You can configure the BroadSoft AS-Feature set on a per-line basis only.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. Under **Calling Features**, clear the check box in the **Default** column for `X_ASFeatureEventSubscribe`.
3. Select the check box in the **Value** column for `X_ASFeatureEventSubscribe`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Set the Subscription Renewal Intervals

The AS-Feature is based on the SIP subscribe and notify framework, and you can set the `expires` value of the subscription dialog to initiate the subscription renewal for each SP service feature enabled.

Procedure

1. In the system web interface, go to **Service Providers > ITSP ProfileN > SIP**.
2. Under **Feature Configuration**, in the **Default** column, clear the check box for `X_ASFeatureEventSubscribeExpires`.
3. In the **Value** column, enter the value in seconds.
The default is 3600 seconds.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Call Forwarding for All Calls

The functionality provided by **Call Forward All** is similar to that of the **CallForwardUnconditional** function provided natively by the handset (per line).

Poly recommends that you disable the native version when using the network-provided version to avoid ambiguity.

Procedure

1. In the system web interface, go to **Service Providers > ITSP ProfileN > SIP**.

2. Under **Network Provided Services**, clear the check box in the **Default** column for `CallForwardAlways`.
3. Select the check box in the **Value** column for `CallForwardAlways`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Note: When you specify the number to forward all incoming calls to, these settings are submitted to and stored on the server.

Enable Call Forwarding for Busy Lines

You can activate call forwarding for calls when the line is busy on your third-party provider.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Default** column, clear the check box for `CallForwardBusy`.
3. In the **Value** column, select the check box for `CallForwardBusy`.
4. Select **Submit**.

Enable Call Forwarding on No Answer

You can activate call forwarding for calls that no one answers on your third-party provider.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Default** column, clear the check box for `CallForwardNoAnswer`.
3. In the **Value** column, select the check box for `CallForwardNoAnswer`.
4. Select **Submit**.

Enable Do Not Disturb

You can enable Do Not Disturb (DND) on your third-party provider.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. Under **Network Provided Services**, in the **Default** column, clear the check box for `DoNotDisturb`.
3. In the **Value** column, select the check box for `DoNotDisturb`.
4. Select **Submit**.

Enable ACD Agent State

ACD (Automated Call Distribution) is the primary way a call center distributes calls among a number of agents.

Through as-event subscription, your phone enables an agent to sign on, sign off, or change their availability states from the phone's local interface. To make this feature available on the phone, you must enable the `ACDAgent` parameter.

For more information on ACD Agent state, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center](#).

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `ACDAgent` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Security Classification

To enable users to view or set the security levels for the extension from the phone, you must enable the `SecurityClass` option.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `SecurityClass` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Executive Call Filter

You can make the **Executive Call Filter** option viewable and changeable from the phone.

Enable the Executive Call Filter

To enable users to view and change the **Executive Call Filter** option from the phone, you must enable the `Executive` parameter.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `Executive` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define an Executive Call Filter Feature Key

You can define a feature key with the function **Exec Filter On/Off** as a shortcut to turn this setting on or off.

The LED color also reflects the current on or off status.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where n =Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function` parameter, clear the check boxes.
4. In the **Value** column for the `Function` parameter, select **Exec Filter On/Off**.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Enable the Executive Assistant Function

You can enable the executive assistant feature on the phone.

By enabling the `ExecutiveAssistant` parameter, the phone makes the following settings of this feature available to the user:

- From a list of executives currently associated with the assistant, turn the call filtering feature on or off for any of the executives.
- Enable or disable the **Divert** option and set or modify the phone number to divert to.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `ExecutiveAssistant` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Define an Executive Assistant Feature Key

You can define a feature key with the function **Exec Assistant** as a shortcut to launch this option.

The LED color also reflects the current **Divert** on or off status.

Procedure

1. In the system web interface, do one of the following:
 - Go to **IP Phone > n Line Keys** (where n =Left or Right).
 - Go to **IP Phone > Programmable Keys**.
2. Select a key (for example, **Key 1** or **Key 2**).
3. In the **Default** column for the `Function` parameter, clear the check boxes.
4. In the **Value** column for the `Function` parameter, select **Exec Assistant**.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Enable Call Recording

The phone can extract the call recording settings from the as-event notifications to help determine which call recording controls to present to the user during a call.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `CallRecording` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure BroadSoft XSI Features

XSI features is a collection of features provided with a BroadSoft XSI application server.

Procedure

1. In the system web interface, go to **Service Providers > ITSP ProfileN > SIP**.
2. In the **Default** column, clear the check box for the following parameters:
 - `X_XsiServer`
 - `X_XsiServerPort`
 - `X_XsiServerScheme`
3. In the **Value** column, do the following:
 - For `X_XsiServer`, enter the hostname or IP address.
 - For `X_XsiServerPort`, enter the server port.
 - For `X_XsiServerScheme`, select **HTTP** or **HTTPS** from the drop down.
4. Select **Submit**.
5. In the system web interface, go to **Voice Services > SPN Service**.
6. Under **SIP Credentials**, in the **Default** column, clear the check box for the following parameters:
 - `X_XsiUserName`
 - `X_XsiPassword`
7. In the **Value** column, do the following:
 - For `X_XsiUserName`, enter the user name to authenticate the XSI server.
 - For `X_XsiPassword`, enter the password to authenticate the XSI server.
8. Select **Submit**.
9. Reboot your system when you complete your changes.

Network Directories

Network Directories are directories hosted by a server somewhere in the network.

The network directory results screen provides four softkeys:

- **Call**: Call the highlighted entry.
- **Search**: Enter one or more search fields and press **OK** or **Enter** to start the search.
- **Prev Page**: Display the previous page of results.
- **Next Page**: Display the next page of results.

Enable BroadSoft-Hosted PBX Platform Services

To access this service from the phone, you must enable the `Directory` parameter of the corresponding SP service.

You can access the network directories on other SP services through the **Net Services** app.

With the BroadSoft BroadWorks platform, the phone supports the following types of network directories:

- Group
- Group Common
- Enterprise
- Enterprise Common
- Personal

Consult your BroadSoft documentation on how to set up and manage these directories on the server side.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `Directory` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Voice Service Network Directories Service

The `VoiceService` parameter controls which SP service's network directories service the phone launches.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Network Directory**.
2. In the **Value** column for the `VoiceService` parameter, select the SP service that provides this feature.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable NetDir on the Phone's Menu

You can launch the network directory service of a specific SP service from the phone using the **Net Dir** app from the **Main Menu** of the phone's local interface.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Network Directory**.
2. In the **Value** column for the `Enable` parameter, select the check box.
3. Go to **Voice Services > SPn Service > Network Provided Services**.
4. In the **Value** column for the `Directory` parameter, select the check box.
5. Select **Submit**.
6. Reboot your system when you complete your changes.

Disable the Search Option

You can disable the **Search** option by setting the `EnableSearch` parameter to **false**.

This replaces the **Search** softkey with the **Refresh** softkey, which reloads the results from the URL.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Directory Setup**.
2. In the **Value** column for the `EnableSearch` parameter, clear the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Other PBX Platforms

A service provider can host a group directory similar to the format used in BroadWorks on the network.

You can configure the URL to access this directory with the `URL` parameter. For example:

```
URL = https://mypbx.com/group-dir.php?user=jsmith
```

In addition, you can append the following optional URL parameters to the URL:

- `results`: Number of results to return by the server. This is merely a suggestion to the server, which may return a different number of results. The phone won't reject the result if the number of results returned is different from the request. It makes a best effort to display all the available results.
- `start`: The 1-based index of the first result to return by the server. The implied start index is 1 if not specified.

Example: `https://mypbx.com/group-dir.php?user=jsmith&start=30&results=20`

You may also include user name and password in the URL for authentication to the user, such as:

```
https://jsmith:!4xuIKKl2y@mypbx.com/group-dir.php?user=jsmith
```

Enable BroadWorks Anywhere

To make this function available on the phone, you must enable the `BroadWorksAnywhere` parameter.

There's no specialized app, feature key function, or softkey option to launch this function. You can only launch this function by going through the **Net Services** app.

You can view and change the following settings of this feature from the phone's local interface:

- Turn the **Alert all locations for Click-To-Dial Calls** option on or off
- Turn the **Alert all locations for Group Paging Calls** option on or off
- Enable or disable a location
- Add a location
- Remove a location (you can't remove the last location via XSI)
- Edit a location's **Number** or **Description** attributes

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `BroadWorksAnywhere` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Remote Office

To make this function available on the phone, you must enable the `RemoteOffice` parameter.

There's no specialized app, feature key function, or softkey option to launch this function. You can only launch this function by going through the **Net Services** app.

You can view and change the following settings of this feature from the phone local interface:

- Enable or disable this feature
- Change the Remote Phone Number (you can't remove the last location via XSI)

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
 2. In the **Value** column for the `Remote Office` parameter, select the check box.
 3. Select **Submit**.
 4. Reboot your system when you complete your changes.
- Change the **Phone Number** attribute of a location.
 - Turn the **Answer confirmation required** option on or off for each location.

Enable Simultaneous Ring

To make this function available on the phone, you must enable the `SimultaneousRing` parameter.

There's no specialized app, feature key function, or softkey option to launch this function. You can only launch this function by going through the **Net Services** app.

You can view and change the following settings of this feature from the phone's local interface:

- Enable or disable this feature by turning the **Active** option on or off
- Turn the **Do not ring my Simultaneous Ring Numbers if I'm already on a call** option on or off
- Add a location
- Remove a location (you can't remove the last location via XSI)

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `SimultaneousRing` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Enable Anonymous Call

This feature is similar to the **AnonymousCall** feature that is available natively on the phone (per service).

To use the version provided by the softswitch, disable the corresponding service on the phone.

Use the **Net Services** app to access the feature and define a feature key with the function Block Caller ID. Set the Service parameter of the key to the Service Provider service that provides this feature.

Users can enable or disable this feature from the phone's local interface, and the server stores the feature settings.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `AnonymousCall` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Setting Up a Directory

Topics:

- [Configure the LDAP Server](#)
- [Configure LDAP Search](#)
- [Configure LDAP SASL Authentication](#)
- [Configuring LDAP Search Display Options](#)
- [Configuring LDAP Application Launch Options](#)

Your phone supports a directory search function with an external server using LDAP. To use this function, you must configure an LDAP service on the phone.

You can point to the **Network Directory** option on the main menu of the phone to an LDAP service using LDAP parameters.

Configure the LDAP Server

Configure the LDAP server using the following parameters.

Procedure

1. In the system web interface, go to **IP Phone > LDAP > Server**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
Host	Enter the host name, which can be an IP address or domain name, with an optional <code>ldap://</code> or <code>ldaps://</code> scheme prefix. For example, <code>192.168.15.186</code> , <code>ldap.forums.com</code> or <code>ldap://ldap.testathon.net</code> are acceptable host name formats. Note: If you don't specify the scheme, the phone uses <code>ldap://</code> .
Port	Enter the LDAP server listening (TCP) port. The standard port is 389 for <code>ldap://</code> and 636 for <code>ldaps://</code> . Note: If the port value is 0 or blank, the phone uses the corresponding standard port.
Password	Enter the bind password for simple or SASL authentication. Note: This parameter is case-sensitive.
TLSSecurityProfile	Enter the security profile for 802.1X authentication.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure LDAP Search

You can configure LDAP search settings.

Procedure

1. In the system web interface, go to **IP Phone > LDAP > Search Parameters**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
BindDN	Enter a distinguished name (DN) that is authorized to use the LDAP service. The BindDN value is usually derived from a username that looks like an email address, such as <code>admin@ldap.example.com</code>. For this example, the corresponding BindDN would be: <code>CN=admin,OU=users,DC=example,DC=com</code>. If <code>none</code> is specified, the query is regarded as anonymous, which may or may not be acceptable to the server.
SearchBase	Use SearchBase as the starting point of the LDAP search. Enter a comma-separated list of { object }={ value } pairs, where { object } can be any of the following values: • CN (Common Name) • OU (Organization Unit) • (Organization) • c (Country) • DC (Domain) If the value isn't specified, the phone by default derives the search starting point from the value of the LDAP server <code>Host</code> parameter. It's a common convention to use just the last two parts of the service domain as a search base, but it isn't specified.
ProtocolVersion	Select either 3 or 2. 3 is the default.
TLS_ReqCert	Select either <code>never</code> or <code>demand</code> . The default value is <code>never</code> , which means the client won't check or verify the server's certificate.

Parameter	Description
ResultsPerPage	Select from the following values to specify how many results to display on the screen, per page: • 20 • 40 • 60 • 80 • 100 The Default value is 20.
DefaultSearchFilter	Enter a default search filter to append to each search. The default value is (objectclass=*). This value must be specified as a complete and valid LDAP search filter, for example: ((objectclass=contact)(objectclass=person)) .
QueryFields	Enter a comma-separated list of user input LDAP attributes/Display-Name to form the query filter. Each item has three attributes separated by two slashes (/) : {ldap-attr}/{display-name}/{type}where • {ldap-attr} is the standard ldap-attribute-name, such as sn, givenName, telephoneNumber, cn, ... This is the only required attribute in each field. • {caption} is optional. This is the caption to display on the screen for the input box. If not specified, the {ldap-attr} value is used in its place. • {type} is either A or N, for alphanumeric or number type respectively and is case insensitive. If not specified, it's assumed to be A. The default value is givenName/First Name,sn/Last Name,telephoneNumber/Tel,mobile/Mobile,homePhone/Home.
ResultFields	Enter a comma-separated list of LDAP fields to display for each entry of the search result. Each field has three attributes separated by two slashes (/) : {ldap-attr} / {caption} / {type} where • {ldap-attr} is the standard ldap-attribute-name, such as sn, givenName, telephoneNumber, cn, ... This is the only required attribute in each field. • {caption} is optional. This is the caption to display on the screen for the input box. If not specified, the {ldap-attr} value is used in its place. • {type} is either s, c, or N, for String, Callable Number, or Picture, respectively. It's case insensitive. If not specified, it's assumed to be s. The default value is cn/Common_Name,sn/Last_Name,givenName/First_Name,telephoneNumber/Tel/y,mobile/Mobile/y,homePhone/Home/y.
NameFieldPreference	Enter a comma-separated list of LDAP attributes to be used as the Caller ID Name to display on the screen, ordered by preference. The first nonempty value in the list is used. The default value is cn, givenName sn.

Parameter	Description
NumberFieldPreference	Enter a comma-separated list of LDAP attributes to be used as the Number to display on the screen and to call by default, ordered by preference. The first nonempty value in the list is used. The default value is telephoneNumber,mobile,homePhone.
PhotoFieldPreference	Enter a comma-separated list of LDAP attributes to be used as the photo to display on the screen, ordered by preference. The first nonempty value is used. The default value is thumbnailPhoto.
SortByAttribute	Enter the LDAP attribute to use for sorting the search results by the server. The default value is cn.
Attributes	Enter a comma-separated list of attributes to request from the server.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure LDAP SASL Authentication

Simple Authentication and Security Layer (SASL) involves sending the LDAP server the FQDN of the client and the corresponding password in clear-text.

This method has security issues unless you use `ldaps://` or TLS.

Note: LDAP v2 supports `ldap://` and `ldaps://` with simple authentication only. LDAP v3 adds support for TLS and SASL authentication.

For more information on SASL, go to <http://www.openldap.org>.

Procedure

1. In the system web interface, go to **IP Phone > LDAP > LDAP SASL Authentication Parameters**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
SASL_AuthMethod	Select the method to use for SASL authentication using any of the following parameters: • Disabled (Default) • Plain • MD5
SASL_AuthCID	Enter the authentication ID for SASL authentication. The format of this ID depends on the actual SASL mechanism used.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring LDAP Search Display Options

After you set up the LDAP server, you can configure how your phones display LDAP search options.

Specify LDAP Search Fields

You can specify which LDAP search fields your phone displays.

By default, your phone displays the following search fields:

- Last Name (`sn`)
- First Name (`givenName`)
- Tel Num (`telephoneNumber`)
- Mobile Num (`mobile`)
- Home Num (`homePhone`)

Procedure

1. In the system web interface, go to **IP Phone > LDAP > Search Parameters**.
2. In the **Value** column for the `QueryFields` parameter, enter a comma-separated list of user input LDAP attributes.

```
givenName/First Name, sn/
Last Name=s*,telephoneNumber/Tel,mobile/Mobile,homePhone/Home
```

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure the LDAP Attributes for Search Results

You can control the LDAP attributes used for the **Name**, **Number**, and **Picture** display by setting the corresponding parameters.

Each of these parameters is a comma-separated list of LDAP attributes arranged in order of preference.

Procedure

1. In the system web interface, go to **IP Phone > LDAP Search Parameters**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
<code>NameFieldPreference</code>	Enter a comma-separated list of LDAP attributes to be used as the Name to display on the screen, ordered by preference.
<code>NumberFieldPreference</code>	Enter a comma-separated list of LDAP attributes to be used as the Number to display on the screen, ordered by preference.
<code>PhotoFieldPreference</code>	Enter a comma-separated list of LDAP attributes to be used as the caller photo to display on the screen, ordered by preference.

3. Select **Submit**.

4. Reboot your system when you complete your changes.

Customize the LDAP Attribute Display

You can customize the attributes to display on the **LDAP Result Details** screen with the `ResultFields` parameter.

Procedure

1. In the system web interface, go to **IP Phone > LDAP > Search Parameters**.
2. In the **Value** column for the `ResultFields` parameter, enter a comma-separated list of LDAP attributes to display for each entry of the search result.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring LDAP Application Launch Options

You can configure your phones to launch the LDAP application from the **Main Menu** or with a softkey.

Set Up LDAP Application Launch in the Directories List

You can configure your phones to launch the LDAP search application from **Main Menu > Directories**.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > GUI Menus**.
2. In the **Value** column for the `MainMenu1` parameter, enter a value string:
 - directories
 - calls
 - netdir
 - call-histories
 - all-preferences
 - settings
 - prod-info
 - messages
 - buddy
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Up LDAP Application Launch Using the Network Directory Service

You can configure your phones to launch the LDAP application using the network directory service.

To set up a network directory service on the **Main Menu**, you must first include `netdir` in the `MainMenu1` parameter in **IP Phone > Phone Settings > GUI Menus**.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > Network Directory**.
2. In the **Value** column, configure the following parameters:
 - **Enable:** Select the check box.
 - **VoiceService:** Select **LDAP**.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set Up LDAP Application Launch Using a Softkey

To use a softkey to launch the LDAP application, add the **LDAP** softkey to any of the configurable softkey sets.

Procedure

1. In the system web interface, go to **IP Phone > Soft Keys > Soft Key Sets**.
2. In the **Value** column, configure the LDAP value for any configurable softkey set.
For the **Home** softkey set, configure the following value: `redial, cfwd, dnd, ldap`
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Customizing the Local Interface

Topics:

- [LED Patterns](#)
- [Customize the Local Interface Menu](#)
- [Phone Customization Data](#)
- [Select User Preferences Settings from Internal Data Storage Paths](#)

There are a number of ways you can customize the appearance of VVX business IP phones, such as customizing LED patterns or creating background images.

LED Patterns

VVX phones have a number of states that support customizable LED patterns.

For example, you can configure an LED pattern of one long green light with two short red lights to represent a specific phone state.

Define an LED Pattern

You can customize the LED patterns for each phone state by configuring the **LED Settings**.

Each customizable pattern has its own parameter as a comma-separated list of {Color} [{Duration}] pairs, where {Color} = R (for red), G (for green), or X (for off). The optional {Duration} is the number of milliseconds to show the given color. If you don't specify a {Duration}, the LED stays at the given color indefinitely. The entire pattern plays repeatedly from left to right indefinitely until the state changes.

The following are some examples of LED patterns:

```
X - Steady off
```

```
R - Steady red
```

```
R500,X500 - 500 ms red followed by 500 ms off
```

```
G50,X50,G50,X1000 - 50 ms green, 50 ms off, 50 ms green, 1s off
```

Procedure

1. Open the configuration file in an editor.
2. Go to the **LED Settings** section of the file.
3. Make the desired changes and save the configuration file.

Customize the Local Interface Menu

You can customize some of the menus on your phone's local interface.

Each menu parameter is a comma-separated list of menu items, where each menu item is specified with an item id followed by an optional semicolon and an item display text separated by a semicolon.

For example:

```
menu
= item, item, ..., item
item
= item-id;item-display-text
```

When you specify multiple instances of the same menu parameters, the items are concatenated internally into a single item list. Items are displayed in the specified order. If `display-text` isn't specified for an item, the default displayed text is used.

Procedure

1. In the system web interface, go to **IP Phone > Phone Settings > GUI Menus**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
MainMenu1	Enter the menu items you want to display on the phone's Main menu. For example:directories,calls,call-histories,all-preferences,settings,prod-info,messages,buddy
PreferencesMenu n (where $n=1-6$)	Enter menu items you want to display on each of the Preferences menus. For example:language;Language,timeFormat;Time Format,dateFormat;Date Format
SettingsMenu n (where $n=1-2$)	Enter menu items to display on each of the Settings menus. For example:net;Network,wifi;WiFi,flash;Storage,ringfile;Ringtones
ProductInfoMenu n (where $n=1-2$)	Enter menu items to display on each of the Product menus. For example:ipaddr,model,obnumber;OBi Number,mac;MAC Address,wfmac;WiFi MAC Address,serial;Serial Number
NetServicesMenu1	Enter menu items to display on the NetServices menu. For example:regst,ringtone,acd,bac,bci,bwanw,buddy,ccs,cfa,cfb,cfna,rec,dnd,dispcode,exec,xass,hotel,clog,dir,rmoft,secClass,simring
NetDirectoryMenu1	Enter menu items to display on the NetDirectory menu. For example:name,firstName,lastName,ext,number,email,groupID,department,mobile,userID,ImpID,presence
DeviceAdministrationMenu1	Enter menu items to display on the Device Administrator menu.admin.port,admin.adminPassword,admin.userPassword,admin.syslogServer,prov.itsp.Method,prov.itsp.Interval,prov.itsp.ConfigURL,fwup.Method,fwup.Interval,fwup.FirmwareURL

Parameter	Description
FactoryResetRequiredAdminPassword	Select the check box to require user to enter a Web Administrator password to factory reset from the phone.
CustomDictionary	Specify an XML dictionary to replace built-in GUI messages with custom messages.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Phone Customization Data

You can customize some of the data that drives the local interface.

These data are collectively called phone customization data, which includes the data types in the following table:

Phone Customization Data Type

Customization Data Type	Description	Internal Data Storage Path
Background pictures (wallpaper)	The pictures that you can choose to show as wall paper during normal phone operation. The following formats are supported: JPEG, PNG, BMP, or GIF.	/backgnd/
Splash screen	Logo . raw file used as the phone's splash screen.	/backgnd/
Dictionary	Dictionary used to enable a localized user interface.	/dict/
Fonts	The true-type font files for rendering text to display on the screen.	/fonts/
Ringtones	Ringtones in 16-bit mono linear PCM wave format sampled at 16 kHz.	/ringtones/
Contacts	Phone book contact pictures.	/people/

Create the Phone Customization Data Package

You can create the phone customization data package.

You must consolidate customizable data into one single `.tar` data package to upload it onto the phone. The size of the tar file must be no larger than 30 MB.

Procedure

1. Create a directory on a Linux machine (for example, `/home/test/obiphone-data`) that contains the same folder structure as the phone's internal storage:
 - `/backgnd`
 - `/dict`
 - `/fonts`
 - `/ringtones`
 - `/people`
2. Copy all the data files you want to upload to the phone in the corresponding folders in this directory and issue the following command lines to create the target `tar` file:

```
% cd /home/test/obiphone-data
% tar -zcvf ../obiphone-data.tar.gz *
```

Note: You must issue the `tar` command inside of the working directory `obiphone-data`, as the phone can't strip the first level in the `.tar` file.

3. Optional: Use the following command line to generate the MD5 checksum of the target `.tar` file after you create it:

```
% md5sum obiphone-data.tar.gz
```

The MD5 checksum is used as the version of the target customization data package. You must configure it on the phones to trigger the phones to download a different version of the package if the currently installed package has a different MD5 checksum from the configured MD5 checksum.

Upload the Customization Data Package to the Phone

After you create the `.tar` file of the customization data package, you must place it on a server so the phones can upload it on request.

Procedure

1. In the system web interface, go to **System Management > Auto Provisioning > Phone Customization Data Package**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
Method	Select one of the following operational methods: • Disabled=Do not execute DownloadURL • System Start=Execute DownloadURL just once on system start • Periodically=Execute DownloadURL on system start and then periodically at the interval specified in the interval parameter Note: First download on system start occurs after firmware update and configuration provisioning are complete.
Interval	Enter the interval in number of seconds between execution of DownloadURL. This parameter is valid when Method is set to Periodically. Note: If the value is 0, the phone executes DownloadURL just once on system start.
TimeOfDay	Enter the time of day in hh:mm[+rr] format. This option is valid when the Method parameter is set to Time of Day.
DownloadURL	Enter the URL to download the Phone Customization Data Package, such as http://abcd.com/phone/cfg/datapkg.tar. The full syntax is a script similar to what you can specify in the FirmwareURL parameter for firmware update. For a description of all the syntaxes of this script, refer to the <i>Poly OBi Edition Deployment Guide</i> at the Poly Online Support Center.
MD5Checksum	Enter the standard MD5 checksum of the Customization Data Package (a 32-character string of hexadecimal digits). You must provide this value to the phone. It serves as the version of the target data package. The phone executes DownloadURL only if the currently installed data package has a different MD5 checksum from the value of this parameter.
Incremental	Select the check box to enable this Boolean option. This will cause the phone to install the customization data files without erasing the old data files first. Otherwise, the old data files are erased before saving the downloaded data files.
DNSLookupType	Select one of the following options: • A Record Only • SRV Record Only • Try Both
DNSSrvPrefix	Select one of the following options: • No Prefix • With Prefix • Try Both
Username	Select the check box to enable username. (Optional) Username for authentication if DownloadURL uses the scheme http:// or https:// .

Parameter	Description
Password	Select the check box to enable password. (Optional) Password for authentication if DownloadURL uses the scheme <code>http://</code> or <code>https://</code> .

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Select User Preferences Settings from Internal Data Storage Paths

You can directly or indirectly select one of several internally stored data files for the phone to perform certain tasks, such as set the background picture or enable a ringtone.

The data files are organized into three levels:

- OBi Built-in
- ITSP (or Administrator) Customized
- User Customized

Each level has its own dedicated internal storage areas.

Procedure

1. In the system web interface, go to **User Settings > User Preferences**.
2. In the **Value** column, configure the following parameters:

Parameter	Description
Language	Select one of the available languages. The parameter value must match the language attribute of the root element of one of the dictionary XML files found under one of the following internal data storage folders: • User customized -- <code>/scratch/phone/dict/</code> • ITSP customized -- <code>/scratch/itsp/dict/</code> • Phone built-in -- <code>/data/dict/</code> Note: When there is a conflict, the user version has the highest priority; then, the ITSP; then, your phone's version. On each bootup, folders are scanned to create a list of available languages from the lang attributes that users can select.

Parameter	Description
BackgroundPicture	Enter full internal path name of a picture file stored in one of the following folders: • User customized -- /scratch/phone/backgnd/ • ITSP customized -- /scratch/itsp/backgnd/ • Phone built-in -- /data/themes/backgnd/ If you specify a URL in the value, it must start with http:// or https:// and the phone downloads and caches the data internally until it's power cycled. Note: The filenames must have one of the following extensions: PNG, JPG, JPEG, GIF, or BMP.
DefaultRingtone	Enter full internal path name of a wave file stored in one of the following folders: • User customized -- /scratch/phone/ringtones/ • ITSP -- /scratch/itsp/ringtones/ • Phone built-in -- / data/ringtones/ If you specify a URL in the value, it must start with http:// or https://. The phone downloads and caches the data internally until it's power cycled. All available wave files under these folders are listed under a) the Default Ringtone option of the Preferences menu for users to select as their default ring tone, and b) the Ringtone field of the built-in Phone Book application for users to choose a ring tone for individual contact in phone book.
DefaultFont	Enter one of the available font family names found in the font.map files in the following folders: • User customized -- /scratch/phone/ringtones/ • ITSP customized -- /scratch/itsp/ringtones/ • Phone built-in -- /data/fonts/ When there is a conflict, the versions have the following priorities: • User version • ITSP customized version • The phone's version If a parameter value is to specify the filename of an internally stored data file, the full internal path must be specified. For example: User Settings > User Preferences::BackgroundPicture = /scratch/phone/backgnd/CherryBlossom.png User Settings > User Preferences::DefaultRingtone = /data/ringtones/Office A.wav

3. Select **Submit**.
4. Reboot your system when you complete your changes.

System Maintenance

Topics:

- [Reporting](#)
- [Updating the Firmware](#)
- [Capture Your Device's Current Screen](#)
- [Factory Resetting Your Phone](#)

You can perform system management and maintenance tasks through the system web interface. You can also use different methods to upgrade the firmware and software on your VVX business IP phone, OBi Edition.

Reporting

Your VVX business IP phones, OBi Edition, provides reports on multiple statistics and functions. Configure the phone to enable or disable reports in the system web interface.

Enable the X-RTP-Stat Feature

You can enable the X-RTP-Stat feature to enable your phone to collect the RTP statistics for calls.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP** (where X=the ITSP profile for the service provider).
2. In the **Value** column for the `X_InsertRTPStats` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

RTP Statistics - the X-RTP-Stat Header

When ending an established call, your device can include a summary of the RTP statistics collected during the call in the SIP BYE request or the 200 response to the SIP BYE request sent by the peer device.

The summary is carried in an X-RTP-Stat header in the form of a comma-separated list of fields.

The following table lists the reported fields.

Field	Description
PS	Number of Packets Sent
PR	Number of Packets Received
OS	Number of bytes sent

Field	Description
OR	Number of bytes received
PL	Number of packets lost
JI	Jitter in milliseconds
LA	Decode latency or jitter buffer size in milliseconds
DU	Call duration in seconds
EN	Last Encoder Used
DE	Last Decoder Used

For example:

```
X-RTP-Stat:PS=1234,OS=34560,PR=1236,OR=24720,JI=1,DU=1230,PL=0,EN=G711U,
DE=G711U
```

RTCP Reports

Your phone supports Real-time Transport Control Protocol (RTCP), which is defined in RFC 3550.

RTCP sends control packets to participants in a call.

Real-time Transport Control Protocol Extended Reports (RTCP-XR), which is defined in RFC 3611, is also supported. RTCP-XR transmits VQ reports for a call.

Enable RTCP and RTCP-XR

You can configure RTCP and RTCP-XR to enable VQ reports to be sent to the proxy server.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > RTP** (where X=the ITSP profile for the service provider).
2. In the **Value** column, select the check box for the following parameters:
 - Enable
 - X_VqPublishEnable
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Updating the Firmware

You may need to update the firmware on your phone to receive new features, enhancements, and bug fixes.

Update the Firmware Using Interactive Voice Response

You can update your phone's firmware by calling into the interactive voice response (IVR) menu.

Procedure

1. Dial ***.
2. Press **6**.
3. If a firmware update is available, press **1** to continue.
4. Hang up.

The download and update process begins.

Update the Firmware from the Local Interface

You can update your phone's firmware from the local interface.

Procedure

1. Launch the **Settings** app from the main menu.
2. Press the **F/W Update** softkey.
A message displays to describe whether new firmware is available.
3. If available, press the **Update Now** softkey.
Don't power off the phone until the update completes and the phone reboots.

Updating the Firmware in the Background

You can update the firmware in the background on VVX 250, VVX 350, and VVX 450 business IP phones.

Once enabled, the phone handles the firmware update without going through the Polyupdater, and the update process occurs in the background during normal operation. Users can still use the phone normally.

After the new firmware copies to the flash memory (and the phone isn't in a call), the phone auto-reboots to use the new firmware version.

Note: Phones with software version 6.3.1 or later, background firmware update is enabled by default. You must always update the VVX 150 software manually.

Enable Background Firmware Update

Enabling background firmware update requires a one-time format transition process.

This process takes 4 to 5 minutes to complete, and no one can use the phone during this period.

Note: Poly recommends that you DO NOT disconnect the power during the process.

Procedure

1. In the system web interface, go to **System Management > Auto Provisioning > Auto Firmware Update**.
2. In the **Value** column for the `FirmwareURL` parameter, enter the URL of the firmware package.
Append the `bgu-conv=enable` keyword, such as:

```
http://192.168.1.1/backgroundupdate/,bgu-conv=enable.
```

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Once the transition process completes, the phone auto-reboots and factory resets. All existing configuration, call logs, and phone book are lost. The `BackgroundUpdate` parameter also changes to **Enabled**.

Note: If the procedure is interrupted or fails, you must use the Polyupdater to recover the device. It's sometimes necessary to reinstall a `sip.ld` formatted image into the device.

Use the Background Firmware Update Feature

After the feature is enabled, the device only takes a `.fw` formatted image.

When using auto firmware update during provisioning, you must provide a complete URL, including the firmware file name, in the `FirmwareURL` parameter. The phone does NOT append `<partnumber>.sip.ld` to the path when downloading the image. Failure to include the `.fw` firmware file name results in a 404 or File not found error.

When installing a `.fw` firmware from the system web interface, the phone doesn't take a config XML file as firmware file anymore, as is the case of a `sip.ld` file. You must provide the path to the firmware file directly. If the given file isn't a valid `.fw` firmware file, the firmware update procedure fails.

Procedure

1. In the system web interface, go to **System Management > Auto Provisioning > Auto Firmware Update**.
2. In the **Value** column for the `FirmwareURL` parameter, enter the URL of the firmware package.
The following is an example of a correct URL:

```
http://192.168.1.1/Skyline-6-3-1-25001-devel-2018-12-13.fw
```

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Revert Firmware Updates to sip.ld

Reverting back to use sip.ld firmware is equivalent to disabling background firmware update (to a .fw firmware). This requires a one-time format transition process on the device that has background firmware update enabled.

Procedure

1. In the system web interface, go to **System Management > Auto Provisioning > Auto Firmware Update**.
2. In the **Value** column for the `FirmwareURL` parameter, enter a complete URL.

This URL must include the firmware file name of a corresponding sip.ld formatted image corresponding to the phone model in use in the `FirmwareURL` parameter.

You must append the parameter with the special keyword `bgu--conv=disable`. For example:

```
http://192.168.1.1/ucs/3111-48840-001.sip.ld,bgu-conv=disable
```

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Capture Your Device's Current Screen

From the system web interface, you can take a snapshot of the current phone screen and save it as a .bmp file on your host computer.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Web Server**.
2. In the **Value** column for the `LCDScreenShot` parameter, select the check box.
3. Go to **System Management > Device Update**.
4. Press **Snap**.

The .bmp file of your phone's current screen downloads to your host computer.

Factory Resetting Your Phone

You can reset all configuration parameters to factory default values or to the customized default values.

Factory Reset Your Phone in the Local Interface

You can reset all phone settings to factory default values from the phone's local interface.

Procedure

1. Go to **Settings**.
2. Press the **Factory Reset** softkey.
3. Press the **OK** softkey.

Factory Reset Your Phone in the System Web Interface

You can reset the phone user data and voice configuration settings to factory default values in the system web interface.

Procedure

1. In the system web interface, go to **System Management > Device Update > Reset Configuration**.
2. Select **User Data**, **Voice Configuration**, **Networking**, or all three.
3. Select **Reset**.

Troubleshooting

Topics:

- [System Logs](#)
- [Contacting Support](#)
- [TCP/TLS Connection with ProxyServer Closed](#)
- [List Alternative UDP Ports if Upstream Router Blocks REGISTER Response](#)
- [Phone Encounters Registration Error](#)
- [Limitations of Transfer by Internal Bridging](#)

Refer to the following topics to help you diagnose and fix issues with your system.

System Logs

Logs contain information about system activities and configurations to help you troubleshoot issues.

Enable Network Call Logs

The network call logs consist of four logs: All, Missed, Received, and Outgoing.

The server stores log data and downloads it to the phone when you invoke this function. Consult BroadSoft on how to manage these call logs on the server side.

To make the network call logs function available on the phone, you must enable the option `CallLogs` parameter. There's no specialized app, feature key function, or softkey option to launch network call logs. You can only invoke this function by going through the Net Services app.

If you enable the Buddy List and it's available under the same SP service, the phone displays the presence icon in the network directory.

Procedure

1. In the system web interface, go to **Voice Services > SPn Service > Network Provided Services**.
2. In the **Value** column for the `CallLogs` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Activate Syslog Messaging

You can enable your VVX IP business phones to send syslog messages for troubleshooting.

Procedure

1. In the system web interface, go to **System Management > Device Admin > Syslog**.
2. In the **Value** column, enter the hostname, FQDN, or IP address for your syslog server for `Server`.
3. Set the `Port` to 514.

- Optional: For the **LevelX** settings, choose options from the drop-down menus for the reporting levels you require.

Include Detailed SIP Messages in Syslog Messaging

You can include detailed SIP messages in your VVX IP business phones syslog messages.

Procedure

- In the system web interface, go to **Voice Services > SPn Service > Debug Options**.
- Set the `X_SipDebugEnabled` parameter to the reporting level you require.
- In the `X_SipDebugExclusion` parameter, enter a list of SIP methods (requests and responses) to exclude from the log.

For troubleshooting a call flow, you can exclude methods such as `OPTIONS` that are used for keep-alive purpose in most cases.

View SPN Service Status Messages

You can view the current state of the voice services you have configured for the system.

If a problem exists with the registration or authentication of your system with a prescribed service, a `SIP 4xx` error message displays. You can use this information for troubleshooting issues with SIP-based services.

Procedure

- » In the system web interface, go to **Status > System Status > SPn Service Status**.
The status for this service displays, including any error messages.

SPN Service Status Error Messages

The following table lists some of the SPn Service Status error messages you might encounter when a firmware upgrade fails.

SPn Service Status Error Messages

Error Message	Description
400 Bad Request	The server can't understand the request.
401 Unauthorized	The request must perform authentication.
402 Payment Required	Indicates that payment is required for further processing of request.
403 Forbidden	Sent when the server understands the request and found the request to be formulated correctly, but isn't servicing the request.
404 Not Found	The server hasn't found the SIP URI indicated by the user.
405 Method Not Allowed	The request contains a list of methods that aren't allowed.
406 Not Acceptable	The request can't be processed due to a requirement in the request message.

Error Message	Description
407 Proxy Authentication Required	Indicates that the UAC first has to authenticate itself with the proxy before the request can be processed.
408 Request Timeout	The specified time period in the Expires header field of INVITE request has passed.
423 Interval Too Brief	Returned by a registrar that is rejecting a registration request because the requested expiration time on one or more Contacts is too brief.
480 Temporarily Unavailable	Indicates that the request has reached the correct destination, but the called party isn't available for some reason.
481 Dialog/Transaction Does Not Exist	Indicates that a response referencing an existing call or transaction has been received for which the server has no record or state information.
483 Too Many Hops	Indicates that the request has been forwarded the maximum number of times as set by the Max-Forwards header.
486 Busy Here	Indicates that the user agent is busy and can't accept the call.
487 Request Terminated	Sent by a User Agent that has received a CANCEL request for a pending INVITE request.

Contacting Support

Contact obi.spsupport@poly.com to learn more about PDMS-SP and how to sign up to receive access.

If you need to talk to Poly technical support, have the following information handy:

- Phone models
- Software versions
- Partner platforms

TCP/TLS Connection with ProxyServer Closed

When using TCP/TLS, the phones initiate a TCP/TLS connection only with the `ProxyServer` parameter.

All subsequent SIP messages are exchanged between the phones, and the servers must use the same connection.

If for any reason the TCP/TLS connection is closed, your phone attempts to re-establish the connection following an exponential back-off retry pattern.

Note: Dynamic address binding through periodic registration isn't necessary if the local IP address of your phone doesn't change. You can statically configure your phone's contact address on the registration server. Also, your network administrator may be able to reconfigure the duration of your DHCP lease so that your DHCP address effectively doesn't change.

Disable the +SIP-Instance Parameter

Your phone includes the `+sip-instance` parameter in the contact header that specifies your phone's MAC address in the UUID.

You can suppress this parameter by disabling the `X_RegisterIncludeInstance` option.

Here is a typical REGISTER request generated by your phone:

```
REGISTER sip:as.xyz.broadworks.net:5060 SIP/2.0
Call-ID: 7107d244@192.168.15.207
Content-Length: 0
CSeq: 10722 REGISTER
From: <sip:3134445567@as.xyz.broadworks.net>;tag=SP337b73f3bf7a504c3
Max-Forwards: 70
To: <sip:2404982564@as.xyz.broadworks.net>
Via: SIP/2.0/UDP 192.168.15.207:5062;branch=z9hG4bK-f9e9e56c;rport
User-Agent: OBIHAI/OBi1062-5.0.0.1542
Contact: <sip:2404982564@192.168.15.207:5062>;expires=60;
    +sip.instance="urn:uuid:00000000-0000-0000-0000-9abcde700065>"
Allow: ACK,BYE,CANCEL,INFO,INVITE,NOTIFY,OPTIONS,PRACK,REFER,UPDATE
Supported: replaces, eventlist, record-aware
```

In this example, your phone doesn't use the Expires header in REGISTER requests. Instead, the Expires value (in seconds) is encoded as a parameter in the Contact header. The two methods are equivalent in this usage per RFC3261.

Procedure

1. In the system web interface, go to **Service Providers > ITSP ProfileX > SIP**.
2. In the **Value** column, clear the check box for the `X_RegisterIncludeInstance` parameter.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

List Alternative UDP Ports if Upstream Router Blocks REGISTER Response

In some cases, your phone may not receive any response to its REGISTER from the server if an upstream router blocks the outgoing message sent by it.

Configure your phone to try other SIP user agent ports for sending and receiving SIP packets.

Procedure

1. In the system web interface, go to **Voice Services > SPN Service**.
2. In the **Value** column for the `X_UserAgentPorts` parameter, enter as many as 10 alternative SIP UDP ports.
You must separate each SIP user agent port by a comma.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Your phone cycles through these ports to retry REGISTER until it receives a response from the server.

Phone Encounters Registration Error

When registration encounters an error, you can schedule retries based on the type of error. Each recognizable error type uses a three-digit code.

For 3xx class responses with a valid Contact header, your phone follows the given contact to retry registration quickly. If a valid contact isn't found, or if the number of consecutive redirections reaches 5, your phone considers the 300 response an error and performs standard error handling.

For 401 and 407 responses with a valid Proxy-Authenticate or WWW-Authenticate header, your phone retries registration quickly and includes the properly computed Proxy-Authorization or Authorization header. However, if the error response contains no valid Proxy-Authenticate or WWW-Authenticate header, and if the number of consecutive 401/407 responses received has reached 2, your phone considers the 401/407 a true error and performs standard error handling.

For 423 responses with a valid Min-Expires header value, your phone retries registration quickly with a new Expires value that conforms to the Min-Expires value from the server. However, if the Min-Expires header isn't present in the response or the value isn't larger than the current Expires value sent by your phone, your phone considers it an error and performs standard error handling.

For 5xx - 6xx responses with a Retry-After header, your phone schedules a retry after the specified value.

The standard way the phone handles a REGISTER final non-2XX response is by waiting for a certain number of seconds before trying to register again.

Three-Digit Code/Range	Description
300-699	Error response codes returned by the server.
900-999	Indicate other errors.
900	Timeout waiting for a response from the server.
901	Can't resolve the server name or host isn't reachable.

Configure REGISTER Rules on Error Code

You can configure REGISTER rules on the actual error code with the `X_RegisterRetryResponseCodes` parameter.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP**.
2. In the **Value** column for the `X_RegisterRetryResponseCodes` parameter, enter the error codes you want your phone to retry.

The format of this parameter is the same as a digit map:

```
(<40[17]:w120>|<40[34]:w120>|<99[01]:w120-200>|[4-9]xx)
```

Each rule is a substitution where a certain error code or error code pattern is mapped to the number of seconds to wait. With this example, your phone waits for 120 seconds for 401 and 407

error codes, 120 seconds for 403 and 404 error codes, randomly between 120 and 200 seconds for 990 and 991 error codes, and a fixed default value for all other error codes.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure REGISTER Retry Value

You can configure the REGISTER retry value with the `RegisterRetryInterval` parameter.

Procedure

1. In the system web interface, go to **Service Providers > ITSP Profile X > SIP**.
2. In the **Value** column for the `RegisterRetryInterval` parameter, enter the number of seconds before your phone retries registration.

The syntax `w{a}-{b}` specifies a random range of between {a} seconds and {b} seconds. Error codes not covered by these rules cause your phone not to retry registration after the error.

3. Select **Submit**.
4. Reboot your system when you complete your changes.

Limitations of Transfer by Internal Bridging

The phone acts as a proxy of RTP packets sent by each peer of the bridge, without any transcoding.

While the phone tries to negotiate the codec to use with each call peer that is acceptable by the other peer, the administrator must understand this limitation and configure the codec profiles accordingly. For example, if you use SIP service 1, then a call transfer involving a SIP service 1 call leg must make sure that the other call leg supports the G711U codec.