ADMINISTRATOR GUIDE

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Poly Edge B Series IP Phones

Getting Help

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

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Contents

Before You Begin

Topics:

- Related Poly and Partner Resources
- Trunks, Endpoints, and Terminals Usage
- ITSP Profiles/SP Services

This guide describes how to administer, configure, and provision Poly Edge B Series IP phones.

This guide focuses on the following Poly devices:

- Poly Edge B10
- Poly Edge B20
- Poly Edge B30

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 using 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

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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.

ITSP Profiles/SP Services

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

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, if abstraction of an ITSP profile isn't sufficient to cover a specified ITSP, 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:

- Phone Overview
- Configuration Options

Poly Edge B Series IP phones with Poly Voice Software Lite (PVOS Lite) are monochromatic desktop phones that support **Home** screen paging, the OPUS codec, Acoustic Fence, five-way audio conferencing, and cloud management.

Manage and configure Poly Edge IP phones from the phone’s local interface, the system web interface, Poly Device Management Service for Service Providers (PDMS-SP), or Poly Lens.

Phone Overview

Poly Edge B Series IP phones support the following features and functionalities.

- Two line keys on Poly Edge B10 and B20 phones and four line keys on Poly Edge B30 phones
- Three additional virtual line key pages, expanding the number of line keys on Poly Edge B10 and B20 phones to eight and Poly Edge B30 phones to sixtee
- 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
- Support for standard SIP-based IP PBX and ITSPs/VSPs
- Six SIP accounts with universal inter- and intraservice two-way call bridging among the six accounts and the PDMS-SPservice
- Cloud management enabled via the PDMS-SP service with both a user portal and an ITSP partner portal
- Recursive digit maps and associated call routing (outbound and inbound)
- VoIP network management for endpoint devices and applications
- High-quality voice encoding using G.711, G.722, G.729, iLBC, and Opus codecs
- Background firmware updates and remote device management
- Fully programmable line keys and softkeys
- Programmable feature keys with preassigned functions and labels

Configuration Options

Configure and manage your Edge B Series 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:

- Press **Home**, then go to **Product Info**.
- Press **Home**, then go to **Settings** > **Network**.
- Dial `***`, then select option 1 to enter the **Voice Admin** menu.

Access the Phone's Local Interface

Access and configure Edge B Series IP phones using the phone's local interface.

Procedure

1. Press the **Home** key, then go to **Settings**.
2. 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

All Poly phones have a system web interface that you can access from a 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, view all the parameters that you can configure and read or set their values.

The phone's system web interface will prompt you to change the admin password on initial boot up or factory reset, if it's not automatically changed via zero touch, PDMS-SP or ITSP provisioning.

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 the username and password.

   The default username is `admin`, and the default password is `admin`.

Access the PDMS-SP Portal

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

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

Register your phones on PDMS-SP, which enables you to access all your phones without having to enter the URL `http://`*`Device-IP-Address`*`/` for each phone. Once you add your devices on PDMS-SP, you can manage them on the PDMS-SP portal. The appearance between the PDMS-SP portal and the system web interface is different, but administrators have identical access to settings.

Procedure

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

Registering the System with Poly Lens

Poly Lens provides cloud-based management and insights for your system.

You can register your system with Poly Lens during system setup or on the Poly Lens registration page. For more information, see Poly Lens Help.

Register Later

If you don't register during setup, you can do so on the Poly Lens registration page.

Procedure

1. Go to https://lens.poly.com/go.
2. Follow the instructions to register your system.

Setting Up the Phone

Topics:

- Power Your Phone
- Configure the Primary Line
- Set the Line Capacity
- Set Service Account Credentials

After you set up and power on your phone, configure its features.

For more information about setting up your phone, see your product's setup documentation at the Poly Online Support Center.

Power Your Phone

Power your phone using the provided AC power adapter or using Power over Ethernet (PoE).

Procedure

» Do one of the following:

- Plug a PoE cable to the phone's **SW** Ethernet switch port.

  Edge B10 doesn't support PoE.
- 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.

Configure the Primary Line

Configure the primary line as the default service used to make calls when users aren't using an explicit access code prefix.

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

Primary Line	Parameter Value
SP1	sp1
SP2	sp2
SP3	sp3
SP4	sp4
SP5	sp5

Primary Line	Parameter Value
SP6	sp6
PDMS-SP	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, select the service you want to set as the primary line.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Set the Line Capacity

Set the line capacity to configure 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** > **SP*N* Service** > **Calling Features**.
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 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

Parameter	Description
`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.

UC Software Configuration on Poly Edge B Series IP Phones

Topics:

- Creating Configuration Files
- Add Poly Edge B Series IP Phones to Your Primary Configuration File
- Supported UC Software Parameters

You can use your UC Software provisioning server and configuration files to provision and configure features on Edge B series IP phones.

Using UC Software configuration files, you can provision and configure your phones, and you can include a mix of UC Software parameters and OBi Edition parameters within the configuration file.

Creating Configuration Files

Create a configuration file that includes a mixture of UC Software and OBi parameters.

You can use a small subset of UC Software parameters to configure your phones, mainly for registering the phone with a SIP service. Then use OBi parameters to configure any additional features you want to include on your phone.

The following is an example configuration with both UC Software parameters in standard XML format and OBi parameters in the same format as UC Software parameters.

```
<?xml version="1.0" encoding="utf-8" standalone="yes"?>
<PHONE_CONFIG>
  <config
  tcpIpApp.sntp.address="ntp.poly.com"
  reg.1.server.1.address="server.poly.com"
  reg.1.address="0123456789"
  reg.1.auth.userId="0123456789"
  reg.1.auth.password="poly"
  reg.1.broadsoft.userId="0123456789@server.polycom.com"
  reg.1.server.1.transport="TCPOnly"
  >
  </config>
<OBiParameterList
    VoiceService.1.VoiceProfile.6.SIP.ProxyServer="i3.voip.polycom.com"
    VoiceService.1.VoiceProfile.6.SIP.ProxyServerPort="5066"
    VoiceService.1.VoiceProfile.6.SIP.ProxyServerTransport="TLS"
    VoiceService.1.VoiceProfile.1.Line.6.X_LineName="i3"
    VoiceService.1.VoiceProfile.1.Line.6.X_ServProvProfile="F"
   >
   </OBiParameterList>
</PHONE_CONFIG>
```

Add Poly Edge B Series IP Phones to Your Primary Configuration File

Add Edge B Series IP phones to your UC Software primary configuration file to fetch the latest firmware and configuration files on your provisioning server.

Add the phone firmware and configuration files to your provisioning server.

Procedure

1. In your primary configuration file (`MAC.cfg` or `000000000000.cfg`), enter the following parameters and values:

```
APP_FILE_PATH_EdgeB10="<Edge B10 firmware file name>.fw"
CONFIG_FILES_EdgeB10="<Name of Edge B10 configuration file>.cfg"
APP_FILE_PATH_EdgeB20="<Edge B20 firmware file name>.fw"
CONFIG_FILES_EdgeB20="<Name of Edge B20 configuration file>.cfg"
APP_FILE_PATH_EdgeB30="<Edge B30 firmware file name>.fw"
CONFIG_FILES_EdgeB30="<Name of Edge B30 configuration file>.cfg"
```

2. Save your configuration file.

Supported UC Software Parameters

Poly Edge B series IP phones support a small list of UC Software parameters.

Note: For the `reg.x.server.y.*` parameters, Poly Edge B series phones only support configuring the system to only one server, so use `reg.1.server.1.` for all related parameters.

`call.directedCallPickupMethod`

Specifies how the phone performs a directed call pick-up from a BLF contact.

legacy - Indicates that the phone uses the method specified in the `call.directedCallPickupString` parameter.

native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header.

`call.directedCallPickupString`

The star code to initiate a directed call pickup.

*97

Note: The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.

`call.parkedCallRetrieveMethod`

The method the phone uses to retrieve a BLF resource’s call that has dialog state confirmed.

legacy (default) - Indicates that the phone uses the method specified in `call.parkedCallRetrieveString`.

native - Indicates that the phone uses a native protocol method (in this case, SIP INVITE with the Replaces header).

`call.parkedCallRetrieveString`

The star code that initiates retrieval of a parked call.

Null

Permitted values are star codes.

`call.parkedCallString`

The star code to initiate the call park.

String

*68

Change causes system to restart or reboot.

`device.dns.altSrvAddress`

Sets the secondary server where the phone directs DNS queries.

Server Address

Change causes system to restart or reboot.

`device.dns.serverAddress`

Sets the primary server where the phone directs DNS queries.

Server Address

Change causes system to restart or reboot.

`device.prov.serverName`

IP address

Domain name string

URL

If you modify this parameter, the phone provisions again. The phone also reboots if the configuration on the provisioning server changes.

`device.sntp.serverName`

Enter the SNTP server where the phone obtains the current time.

IP address

Domain name string

`feature.directedCallPickup.enabled`

0 (default) - Disables the directed call pickup feature.

1 - Enables the directed call pickup feature.

Change causes system to restart or reboot.

`reg.x.address`

The user part (for example, 1002) or the user and the host part (for example, `1002@poly.com`) of the registration SIP URI.

Null (default)

String address

`reg.x.broadsoft.userId`

Enter the BroadSoft user ID to authenticate with the BroadSoft XSP service interface.

Null

string

`reg.x.displayName`

The display name used in SIP signaling and the label that displays on the phone key line.

Null

UTF-8 encoded string

`reg.x.label`

The text label that displays next to the line key for registration *x*.

Null

UTF-8 encoded string

`reg.x.outboundProxy.address`

The IP address or hostname of the SIP server where the phone sends all requests.

Null

IP address or hostname

`reg.x.outboundProxy.port`

The port of the SIP server where the phone sends all requests.

0

1 to 65535

`reg.x.outboundProxy.transport`

The transport method the phone uses to communicate with the SIP server.

DNSnaptr

DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly

`reg.x.server.y.address`

If this parameter is set, it takes precedence even if the DHCP server is available.

IP address or hostname - SIP server that accepts registrations.

`reg.x.server.y.expires`

The phone's requested registration period in seconds. The period negotiated with the server may be different. The phone attempts to reregister at the beginning of the overlap period.

3600 (default)

Positive integer, minimum 10

`reg.x.server.y.port`

Null (default) - The port of the SIP server doesn't specify registrations.

1 to 65535 - The port of the SIP server that specifies registrations.

`reg.x.server.y.transport`

The transport method the phone uses to communicate with the SIP server.

If `reg.x.server.y.address` is an IP address or if you provide a port, then the phone uses UDP.

TCPpreferred

UDPOnly

TLS

TCPOnly

`reg.x.srtp.enable`

1 (default) - The registration accepts SRTP offers.

0 - The registration always declines SRTP offers.

Change causes system to restart or reboot.

`reg.x.srtp.require`

0 (default) - Secure media streams are not required.

1 - The registration is only allowed to use secure media streams.

Change causes system to restart or reboot.

`reg.x.type`

private (default) - Use standard call signaling.

shared - Use augment call signaling with call state subscriptions and notifications and use access control for outgoing calls.

`tcpIpApp.sntp.address`

Specifies the SNTP server address.

NULL (default)

Valid hostname or IP address.

`tcpIpApp.sntp.gmtOffsetcityID`

NULL (Default)

0 to 127

`tcpIpApp.sntp.gmtOffset`

Specifies the offset in seconds of the local time zone from GMT.

0 (Default) - GMT

3600 seconds = 1 hour

-3600 seconds = -1 hour

Positive or negative integer

Configuring Network Settings

Topics:

Control how your Poly phones access the web and your network. You also can register your system with SIP for calling.

PVOS Supported Ports

The following table lists the ports used by Poly PVOS Software.

PVOS Supported Ports

Port Number	Protocol	Outgoing	Incoming	UDP or TCP
53	DNS			UDP
67	DCHP	Server		UDP
68	DHCP		Client	UDP
69	TFTP	Provisioning, Logs		UDP
80	Provisioning, Logs, Poll			TCP
123	NTP	Time Server		UDP
389	LDAP			TCP
443	HTTPS	Provisioning, Logs	Web Interface, Phone XML App	TCP
514	Syslog	Logs		UDP
636	LDAP	LDAP		
16600-16798	RTP	Media Packets	Media Packets	
16600-16798	RTCP	Media Packet Statistics	Media Packet Statistics	

Port Number	Protocol	Outgoing	Incoming	UDP or TCP
	SIP	SIP Signaling	SIP Signaling	TCP and UDP
	SIP over TLS	Secure Signaling	Secure Signaling	TCP

Network Connectivity

Configure the External Ethernet Ports

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
Switch Port > **Speed**	• Auto • Full • Half • Full • Half • Full • Disabled Poly recommends using only the Auto or Disabled settings.
PC Port > **Speed**	• Auto • Full • 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**.
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	Specify up to 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.
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 • PoE Class (Power Draw) .

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

DHCP Options

Configure DHCP Options

The options that the phone tries to extract from the DHCP offer is a comma-separated list of option numbers. The phone doesn't recognize any other option numbers.

Procedure

1. In the system web interface, go to **System Management** > **WAN Settings** > **DHCP Client Settings**.
2. Under **DHCP Client Setting**, clear the check box in the **Default** column for the `ExtraOptions`.
3. In the **Value** column, enter the extra DHCP options in a comma-separated list.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Set the URL for the Configuration File

Set the URL that the system uses for your phone in the system's configuration 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 **Default** column, clear the check box for `ConfigURL`, then enter the URL of the configuration file in the **Value** column.

   The default value for `ConfigURL` is `tftp://$DHCPOPT66/$DM.xml;$DHCPOPT66;`.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

DNS Lookup

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. Configure the phone to control the order the phone queries the server types.

Procedure

1. In the system web interface, go to **System Management** > **WAN Settings**.
2. Under **DNS Control**, clear the check box in the **Default** column for `DNSQueryOrder`.
3. In the **Value** column, select the available DNS servers for the parameter in the order you want them to be queried by the phone.
4. Select **Submit**.
5. 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. Configure the phone 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**.
2. Under **DNS Control**, clear the check box in the **Default** column for `DNSQueryDelay`.
3. In the **Value** column, select a value from the drop-down menu to set a 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.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Define Local DNS Records

Define as many as 32 local DNS records in the phone configuration.

Once you define the DNS records, 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**.
2. Under **Local DNS Records**, clear the check box in the **Default** column for each **Local DNS Record** you want to configure.
3. In the **Value** column, enter a local DNS record for each parameter.
4. Select **Submit**.
5. 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

Configure up to two NTP servers.

Procedure

1. In the system web interface, go to **System Management** > **WAN Settings**.
2. Under **Time Service Settings**, clear the check box in the **Default** column for `NTPServer1` and `NTPServer2`.
3. In the **Value** column, enter the host names or IP addresses of the NTP servers.

4. Select **Submit**.
5. 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. 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**.
2. Under **DHCP Client Settings**, clear the check box in the **Default** column for `ExtraOptions`.
3. In the **Value** column, delete option 42 from the extra options list for `ExtraOptions`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Set the Local Time Zone

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. Configure the local time zone to ensure the phones display the correct time.

Procedure

1. In the system web interface, go to **System Management** > **WAN Settings**.
2. Under **Time Service Settings**, clear the check box in the **Default** column for `LocalTimeZome`.
3. In the **Value** column, select your local time zone from the drop-down menu for the `LocalTimeZone` parameter.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Daylight Saving Time

Enable daylight saving time to enable the phones to automatically adjust when daylight saving time starts and ends.

Procedure

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

Specify Start and End Rules for Daylight Saving Time

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**.
2. Under **Time Service Settings**, clear the check box in the **Default** column for `DaylightSavingTimeStart` and `DaylightSavingTimeEnd`.
3. In the **Value** column, enter the start and end dates for `DaylightSavingTimeStart` and `DaylightSavingTimeEnd`.

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

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

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

Configure Amount of Time to Adjust for Daylight Saving Time

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**.
2. Under **Time Service Settings**, clear the check box in the **Default** column for `DaylightSavingTimeDiff`.
3. In the **Value** column for `DaylightSavingTimeDiff`, enter the amount of time to adjust for when daylight saving time takes effect.

   The format is `h:m:s` or `-h:m:s`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable SIP Signaling to Provide the Time

Enable the phone to use SIP signaling to provide the 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*N*** > **SIP**.
2. Under **Time Service Settings**, clear the check box in the **Default** column for `X_ProcessDateHeader`.
3. In the **Value** column for `X_ProcessDateHeader`, select the check box.
4. Select **Submit**.
5. 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 RegistrarServer

Set up your phones to periodically register to the `RegistrarServer`.

`ProxyServer` is a required parameter that you must configure on your phone. You rarely need the `RegistrarServer` and `RegistrarServerPort` parameters. These parameters are assumed to be the same as the `ProxyServer` and `ProxyServerPort` parameters 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*N*** > **SIP**.
2. In the **Value** column, enter the host name or IP address for the `RegistrarServer` parameter.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configure Third-Party Registration

Configure a third-party registration to register the system for an Address of Record (AOR) that isn't the same as the account user-id.

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 to 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*N*** > **SIP**.
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. Configure your phone to control how the margin to subtract is computed.

Procedure

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

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 NAPTR lookup, 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 of 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*N*** > **SIP**.
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

Configure your phone to use the discovered external IP address and port to replace its private address and port. 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*N*** > **SIP**.
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.

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.

For more information on NAT Traversal considerations, see 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** > **SP*N* Service** > **Calling Features**.
2. Configure the following parameters:

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

<table>
<tr><th>Parameter Group</th><th>Parameter</th><th>Description</th></tr>
<tr><td>SPn Service</td><td><code>X_KeepAliveMsgType</code></td><td>The type of keep-alive messages to send out periodically if keep-alive is enabled. Use one of the following:
keep-alive: 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.
empty: A blank line.
stun: A standard STUN binding request. Your phone uses the binding response to form its contact address for REGISTRATION.
custom: Use the value of X_CustomeKeepAliveMsg.
options: A valid SIP OPTIONS request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled.
notify: A valid SIP NOTIFY request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled.</td></tr>
<tr><td>SPn Service</td><td><code>X_CustomKeepAliveMsg</code></td><td>Defines the custom message to use when X_KeepAliveMsgType is custom. The value has the following format:
<code>mtd=NOTIFY;event={whatever};user={anyone}</code>
Where
NOTIFY may be replaced by any other SIP method, such as PING. The event parameter is optional and is only applicable if the method is NOTIFY.
If the event isn’t specified, the keep-alive event is used with NOTIFY.
The user parameter is optional.
If the user isn’t specified, the request-uri doesn’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 the user is specified, it’s used as the user ID in the Request-URI 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.</td></tr>
</table>

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.

For more information about STUN and ICE, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center].

Configure the STUN Feature

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

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

Note: 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 *N***.
2. In the **Value** column for the `X_ICEEnable` parameter, select the check box.
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Securing the System

Topics:

Download and add custom certificates to your Poly phones.

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.

Load Custom Device Certificates

Load custom device certificates in the system web interface.

Procedure

1. In the system web interface, go to **System Management** > **Device Admin**.
2. Under `Custom Device Certificate`*n*, enter the URL of the certificate to download in the **Value** column for the `DownloadURL` parameter.
3. Under `TLSPlatform Profile`*n*, select the custom device certificate `DeviceCert` in the **Value** column.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Load Custom Platform CA Certificates

Load custom platform CA certificates using the system web interface

Procedure

1. In the system web interface, go to**System Management** > **Device Admin**.
2. Under **Platform CA n**, enter the URL of the root CA certificate to download in the **Value** column for the **DownloadURL** parameter.
3. Under **TLSPlatform Profilen**, select the appropriate value in the **Value** column for **CACertList**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Set the 802.1X Authentication Mode

All Poly phones support 802.1X authentication.

Procedure

1. In the system web interface, go to **System Management** > **WAN Settings** > **Internet Settings**.
2. In the **Default** column, clear the check box for **802_1XMode**, then select one of the following modes from the drop-down menu:
    - Disable
    - MD5
    - TLS
    - TTLS/MSCHAPv2
    - PEAP-MSCHAPv2 (optional for all parameters)
3. Depending on selected mode, configure the following additional authentication parameters:

Parameter	Description	(EAP) MD5	(EAP) TLS (1.0)	TTLS/ MSCHAPv2
`802_1XIdentity`	The username. If you don't need a username, set the value as an empty string.	Required	Required	Required
`802_1XPassword`	The password or passphrase. If you don't need a password or passphrase, set the value as an empty string.	Required	Required	Required
`802_1XAnonymousID`	When empty, the phone doesn't use anonymous identity in authentication.		Required	Required
`802_1XTLSSecurityProfile`	Security profile for the 802.1X authentication.		Required	

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

Set Port Number for the Management Web Server

Set the port number value for the management web server.

Procedure

1. In the system web interface, go to **System Management** > **Device Admin**.
2. Under **Web Server**, 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.

Enable or Disable the PC Port

The PC port can be enabled or disabled to allow or deny another device from connecting to the network through the Poly Edge B's PC port.

Procedure

1. In the system web interface, go to **System Management** > **Device Admin**.
2. Under **External Port Security**, select the `PCPort` parameter and do one of the following:
    - Check (to enable)
    - Uncheck (to disable)
3. Select **Submit**.
4. Reboot your system when you complete your changes.

Configuring Passwords

Change the administrator and user passwords.

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

Controlling User Access to the System Web Interface

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 users 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.

Change the Default Password

The system web interface prompts for a new administrator password when the phone is powered on for the first time or if the phone is factory reset.

Important: You must change the default password before you can access the system web interface.

Procedure

1. Enter `http://`*`<PhoneIPAddress>`* to load the system web interface.
2. Enter the default admin **Username** and **Password** (`admin`/`admin`).

3. Enter the default password (`admin`) for **Old Password**.
4. Enter and re-enter the new password in **New Password** and **Confirm New Password**.
5. Select **Submit**.
6. Select **OK**.

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 admin.
`UserPassword`	User login password. Default is user.

Set the Minimum Password Length

Determine the minimum length for passwords that users and admins can create.

Note: The service provider may specify the minimum length of the password. Check with your service provider for any password requirements.

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.

Lock the System Web Interface

Lock the system web interface for a specified time period after a specified number of failed attempts.

Procedure

1. In the system web interface, go to **System Management Device** > **Admin Web Server**.
2. Select the check box to enable the **Web Server Lockout** feature.
3. Set the period of time, in seconds, that the user is locked out of the web server. Enter a value between 60 and 300.

   ```
   LockOutPeriod="<value>"
   ```
4. Set the maximum number of failed login attempts before the user is locked out. Enter a value between 3 and 20.

   ```
   LockOutInvalidAttempts="<value>"
   ```
5. Set the length of time, in seconds, that the user is locked out. Enter a value between 60 and 300.

   ```
   LockOutInvalidAttemptsDuration="<value>"
   ```
6. Select **Submit**.
7. 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 `ProxyServer` parameters.

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.

`ProxyServer` is a required parameter that you must configure on your phone. You rarely need the `RegistrarServer` and `RegistrarServerPort` parameters. These parameters are assumed to be the same as the `ProxyServer` and `ProxyServerPort` parameters 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*N*** > **SIP**.
2. In the **Default** column, clear the check boxes for `ProxyServer` and `ProxyServerPort`.

3. In the **Value** column, configure the following settings:

Parameter	Description
`ProxyServer`	Enter the host name or IP address of the SIP proxy server.
`ProxyServerPort`	Configure the destination port to connect to the SIP server if the listening port is nonstandard.

Configure the Transport Protocol for SIP Messages

Set the transport protocol for SIP messages using UDP, TCP, and TLS protocols.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. In the **Default** column, clear the check box for `ProxyServerTransport`.
3. In the **Value** column for `ProxyServerTransport`, select the transport protocol to connect to the SIP server.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Local Contact Port for SIP Messages

Configure the local contact port for sending and receiving SIP messages.

The SP*n* services for *n* = 1 to 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** > **SP*N* Service**.
2. In **Default** column, clear the check box for `X_UserAgentPort`.
3. In the **Value** column for `X_UserAgentPort`, 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.

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

SIP Outbound Proxy Server

Configure an outbound proxy server on your phone, so 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 Profile*N*** > **SIP**.
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

Configure the transport protocol for the outbound proxy server, which may use a different transport protocol from the proxy server.

If the `OutboundProxyTransport` protocol is set to 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 outbound proxy 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 Profile*N*** > **SIP**.
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 so that your phone doesn'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*N*** > **SIP**.
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.

Configure the Phone to Use SIP/TLS

When setting `ProxyServerTransport` to TLS, the phone uses `sips` as the SIP scheme when registering to the SIP server. If the SIP server doesn't accept `sips`, the registration fails.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP ProfileN** > **SIP**
2. In the **Value** column, configure the following settings:

Setting	Description
ProxyServer	Enter the host name or IP address of the SIP proxy server.
ProxyServerTransport	Set to **UDP** or leave the default setting if the default value is UDP.
OutboundProxy	Enter the host name or IP address of the outbound proxy server.
OutboundProxyPort	Enter the destination port to connect to the outbound proxy.
X_OutboundProxyTransport	Set to **TLS**.
X_UserAgentContactFollowProxyServerTransport	Select the check box.

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, configure the phone to apply 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*N*** > **SIP**.
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

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

If you disable the `X_ProxyServerRedundancy`, 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*N*** > **SIP**.
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

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*N*** > **SIP**.
2. In the **Value** column for `X_CheckPrimaryFallbackInterval` and `X_CheckSecondaryFallbackInterval`, enter the interval (in seconds) for checking the fallback lists.

   Setting the value to 0 disables this periodic check.
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.

The phone checks the following message headers (all these headers can carry caller's name and number information):

- FROM
- P-Asserted-Identity (PAID)
- Remote-Party-ID (RPID)

For more information on SIP privacy, see 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*N*** > **SIP**.
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

Instruct your phone to block outbound caller ID.

The phone uses `sip:anonymous@localhost` in the FROM header to block outbound caller ID when you configure these settings.

Procedure

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

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.

Edit the Outbound Call Route Configuration

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 `OutboundCallRoute`.
3. In the **Value** column for these parameters, enter the syntax for `OutboundCallRoute`.
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. You can configure the inbound call routes all incoming calls and on a per-line basis.

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** > **SP*N* Service**.
2. In **Default** column, clear the check box for `InboundCallRoute`.
3. In the **Value** column for `InboundCallRoute`, enter the routing rule for inbound calls.
4. Select **Submit**.
5. 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. Configure the digit map to determine the set of number patterns that users can dial.

Use the `(Mname)` syntax to refer to digit maps by name. For example, the default `DigitMap` parameter refers to digit maps defined for SP1, SP2, SP3, SP4, and PDM-SPservices 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 **Default** column, clear the check box for `DigitMap`.
3. In the **Value** column for `DigitMap`, enter a digit map to set the number patterns users can dial.
4. Select **Submit**.
5. 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 you define 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 PDMS-SP 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 **Default** column, clear the check box for `OutboundCallRoute`.
3. In the **Value** column for the `OutboundCallRoute` parameter, enter the routing rule for outbound calls made from the phone.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Customize Service Route Access Codes

If necessary, 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 **Default** column, clear the check boxes for `DigitMap` and `OutboundCallRoute`.
3. In the **Value** column, configure the following parameters:

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

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

Auto Attendant

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

Note: In the phone configuration, the auto attendant 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**.
2. Under **Auto Attendant 1**, in the **Default** column, clear the check box for `Enable`.
3. In the **Value** column, select the check box for `Enable`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure the Auto Attendant Callback Service

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

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** > **SP*N* Service**.
2. In the **Default** column, clear the check box for **X_InboundCallRoute**.
3. In the **Value** column, enter a routing rule for `X_InboundCallRoute` to direct an auto attendant to call back the caller's or another number if the caller hangs up before auto attendant answers.
4. Go to **Voice Services** > **Auto Attendant**.
5. Clear the check box for the following parameters in the **Default** column, then configure them in the **Value** column.
   - `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.
6. Select **Submit**.
7. Reboot your system when you complete your changes.

Customize Service Route Access Codes for the Auto Attendant

Customize the service route access codes, including the outbound call rate and digit map, for calling via the Auto Attendant.

Procedure

1. In the system web interface, go to **Voice Services** > **Auto Attendant**.
2. Under **Auto Attendant 1**, in the **Default** column, clear the check boxes for **DigitMap** and **OutboundCallRoute**.
3. In the **Value** column, enter the rules for the auto attendant for **DigitMap** and **OutboundCallRoute**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Voice Services

Topics:

- ITSP Profiles
- The PDMS-SP 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

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 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 supporting Polydevices.

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 PDMS-SP 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. The service subscriber, known as an SP service, includes the `X_ServProvProfile` parameter that binds the SP service with an ITSP Profile. Every SP service user account requires a username 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

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.

Tip: To avoid a common mistake, set the `X_ServProvProfile` parameter correctly to point to the corresponding ITSP profile.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. In the **Default** column, clear the check box for **X_ServProv_Profile**.
3. In the **Value** column, select a service provider profile from the drop-down menu for **X_ServProv_Profile**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

The PDMS-SP Service

PDMS-SPis a proprietary protocol for communications among Poly Edge B Series IP phones and PDMS-SPdevice management servers.

The protocol is intended for two main purposes:

- Peer-to-peer calling between Poly Edge B Series devices
- Device management by PDMS-SP servers

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

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

Enable the PDMS-SP Service

The PDMS-SP service is enabled by default, unless disabled through Zero Touch customization.

Procedure

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

Limit PDMS-SPCalls

Limit OBiTALK calls to just the OBihai echo server.

A simple way to disable PDMS-SPvoice calls completely is by setting `MaxSessions="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**.
2. In the **Default** column, clear the check box for **DigitMap**.
3. In the **Value** column for **DigitMap**, enter a value of (<ob>222222222 | ob222222222).

   You can change or add more OBi numbers to this digit map by following the same pattern.
4. Select **Submit**.
5. 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 PDMS-PHService

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

Set the trunk capacity to determine the maximum number of simultaneous calls allowed on the trunk.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service** > **Calling Features**.
2. In the **Default** column, clear the check box for **MaxSessions**.
3. In the **Value** column for **MaxSessions**, enter the maximum number of simultaneous sessions that the endpoint can conference.

   The default value is 2 for all services. For the PDMS-SP 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

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
- 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

Reference a trunk group and its associated digit map using the short name TG*n* 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, TG3, 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** > **Trunk Groups**.
2. In the **Default** column, clear the check boxes for the following settings, then configure them in the **Value** column.

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.

Parameter	Value
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 PDMS-SP callers call the gateway first with a normal PDMS-SP call, get the auto attendant, and then dial the target number. For authentication, the auto attendant 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 PDMS-SP number and refer to a gateway and its associated digit map with the short trunk name VG*n* and (Mvgn). You can use VG*n* and (Mvgn) in call routing rules and digit maps just like other real trunks.

For more information on voice gateways, see the *Poly OBi Device Technical Reference* at the Poly Online Support Center.

Configure a Gateway for Direct Dialing

Configure a gateway with one-stage or direct dialing so that the caller can dial the target number directly without going through the auto attendant.

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** > **Trunk Groups**.
2. In the **Default** column, clear the check boxes for **AuthUserID** and **AuthPassword**.
3. In the **Value** column, enter the values for **AuthUserID** and **AuthPassword**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring Call Settings

Topics:

- Conference Calls
- Phone Number Formats
- Configuring Auto-Answer
- Configure Speed Dial Keys
- Caller ID
- Call Forwarding
- Configure Do Not Disturb
- Configuring Do Not Ring
- Configure Message Waiting Indicators
- Configure Multicast Page Groups
- Configure Music On Hold
- PTT Mode
- Busy Lamp Field (BLF)

Determine the call settings that are available to users when they place and answer calls.

Conference Calls

The Edge B Series 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 three-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 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.

Enable an External Conference Bridge

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.

When using an external conference bridge, the bridge limits the conference size. Check with your service provider on the conference size limit.

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** > **SP*N* Service**.
2. Under **Calling Features**, clear the check box for **X_ConferenceBridge**.
3. In the **Value** column, enter the user ID or (SIP) URL of the external conference bridge for **X_ConferenceBridge**.
4. Select **Submit**.
5. Go to **IP Phone** > **Phone Settings**
6. Under **Calling Features**, clear the check box in the **Default** column for **UseExternalConferenceBridge**.
7. In the **Value** column for **UseExternalConferenceBridge**, select the check box.
8. Select **Submit**.
9. Reboot your system when you complete your changes.

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 conference participants 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 conference participant who is connected through the PDMS-SP service, the phone keeps the call with that participant 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 PDMS-SP 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.

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].

Configure Auto-Answer Settings

Choose the signaling method the phone uses to automatically answer incoming calls.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. Under **Feature Configuration**, in the **Default** column, clear the check box for **X_AutoAnswerMethod**.
3. In the **Value** column, select the signaling method from the drop-down menu for `X_AutoAnswerMethod`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Auto-Answer Incoming Calls Based on Inbound Call Routing Rules

Enable 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** > **SP*N* Service**.
2. Clear the check box in the **Default** column for **X_InboundCallRoute**.
3. In the **Value** column, enter a rule that routes the inbound call to `ph(autoans)`.
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.

Procedure

1. In the system web interface, do one of the following:
   - Go to **IP Phone**, then select **Right Line Keys** or **Left Line Keys**.
   - 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.

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

Configure the name that displays when a user places a call.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. Under **Calling Features**, clear the check box for **CallerIDName** in the **Default** column.
3. In the **Value** column, enter a name in **CallerIDName** that displays on the screen during a call.
4. Select **Submit**.
5. 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.

Enable this as 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

Block caller ID for all calls on the phone.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. Clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

Parameter	Description
X_InsertRemotePartyID	Select the check box to insert a SIP Remote-Party-ID header in all outbound INVITE.
X_InsertPrivacyHdr	Select the check box to insert a Privacy:id header in INVITE for anonymous calls.
X_UserAnonymousFROM	Select the check box to use `sip:anonymous@localhost` 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

Set the phones to automatically block the caller ID for all calls on an SP service.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. Under **Calling Features**, clear the check box in the **Default** column for **AnonymousCallEnable**, then select the check box in the **Value** column.
3. Under **Network Provided Services**, clear the check box for **AnonymousCall** in the **Default** column, then select the check box in the **Value** column.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable the Block Caller ID Feature Key

Configure the block caller ID function on a feature key.

Procedure

1. In the system web interface, go to **IP Phone** > **Phone Settings** > **Calling Features**.
2. In the **Default** column, clear the check boxes for **AnonymousCallEnable** and **AnonymousCall**.
3. In the **Value** column, select the check box for **AnonymousCallEnable** and **AnonymousCall**.

   Enabling this option is equivalent to enabling the feature for all services.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Block Caller ID on a Line or Programmable Key

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**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Block Caller ID** from the drop-down menu for **Function**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable the Block Anonymous Calls Feature Key

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**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Block Anonymous Call** from the drop-down menu for **Function**.
4. Select **Submit**.
5. 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 PDMS-SP 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.

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.

Poly Edge B Series 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*N*** > **SIP**.
2. In the **Default** column, clear the check box for **X_Use302ToCallForward**.
3. Select the check box in the **Value** column for **X_Use302ToCallForward**.
4. Select **Submit**.
5. In the system web interface, go to **Voice Services** > **SP*N* Service** > **Calling Features**.
6. In the **Value** column, configure the following parameters.

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

7. Select **Submit**.
8. Reboot your system when you complete your changes.

Configure Call Forward Signaling

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. Select **Submit**.
4. Go to **Service Providers** > **ITSP Profile *X*** > **SIP** .
5. 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.
6. Select **Submit**.

7. Reboot your system when you complete your 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** > **SP*N* Service**.
2. Under **Calling Features**, clear the check box for **DoNotDisturbEnable** in the **Default** column.
3. In the **Value** column, select the check box for **DoNotDisturbEnable**.
4. Under **Network Provided Services**, clear the check box in the **Default** column for **DoNotDisturb**.
5. In the **Value** column, select the check box for **DoNotDisturb**.
6. Select **Submit**.
7. Go to **IP Phone** > **Phone Settings** > **Calling Features** > Phone Settings > Calling Features.
8. In the **Value** column for the `DoNotDisturbEnable` parameter, select the check box.
9. Go to **User Preferences Settings**.
10. In the **Value** column for the `DoNotDisturbFeatureProvider` parameter, select the provider for the DND feature.
11. Select **Submit**.
12. 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 there is 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 to silence the ringer for all incoming calls to the phone.

Procedure

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

Configure a Do Not Ring Feature Key

Define a feature key for the Do Not Ring function.

Procedure

1. In the system web interface, do one of the following:
    - Go to **IP Phone**, then select **Right Line Keys** or **Left Line Keys**.

- Go to **IP Phone** > **Programmable Keys**.

2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column for **Function**, select **Do Not Ring**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Message Waiting Indicators

Configure the visual and audio message waiting indicators 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** > **SP*N* Service**.
2. Under **Calling Features**, clear the check boxes in the **Default** column for the following features, then configure the features in the **Value** column.

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. Select **Submit**.
4. Go to **Service Providers** > **ITSP Profile *N*** > **SIP**.
5. Clear the check boxes in the **Default** column for the following features, then configure the features in the **Value** column.

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

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

Configure Multicast Page Groups

Configure the page groups on your phone. Each phone supports two multicast groups and up to 10 page groups.

Procedure

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

2. Under **Page Group *n***, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

Parameter	Description
GroupName	A user-friendly name to label the group on the phone UI.
MulticastAddress	This must be a valid IPv4 multicast address. The default is `224.1.1.100` for both paging groups.
MulticastPort	The default is 65322 for page group 1 and 65324 for page group 2. **Note:** Each group must use a different port number.
TTL	The TTL value of outgoing (multicast) RTP packets. The default is 2.
ParticipantName	A string value to identify the current user who is paging to the group in outgoing RTCP packets.
AudioCodec	Audio codec to use for outgoing page. Default is G711U.
TxPacketSize	The outgoing RTP packetization in milliseconds. The default is 20.
RTCPTxInterval	The interval, in milliseconds, between sending outgoing RTCP packets when paging. No RTCP packets are sent if the value is 0 (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 is used for outgoing page. The default is false.
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. The default is true.
PreferredAudioDevice	Select the preferred audio device to play incoming page. Enter either System, Headset, or Speaker. The default is System to let the phone picks according to other user preferences.
PageTimeout	Limit of the outbound page duration in number of seconds to avoid accidentally jamming the page group channel. The duration is unlimited if the value is 0 (default).
EnableJoinLeaveGroupUserControl	A Boolean option to control whether to enable you to join or leave a page group by pressing the pg1 through pg10 softkeys. The default is **true**. When set to **false**, users can press the key once to talk and press it again to go back to listen. Although users can’t choose the leave the page group, they can still use the **AutoAnswerIntercom** option from the **User Preferences** menu not to play any of the incoming pages.

Parameter	Description
SpeakerVolume	Select the speaker volume when paging. Use 0 to follow the 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**.
2. Under **Calling Features**, clear the check box in the **Default** column for **MOHServiceNumber**
3. In the **Value** column, enter the music on hold service number for **MOHServiceNumber**.
4. Select **Submit**.
5. 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 group 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**.
2. Under **Page Group *n***, clear the check box in the **Default** column for **PushToTalk**.
3. In the **Value** column, select the check box for **PushToTalk**.
4. Select **Submit**.
5. 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**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Number** in the **Default** column.
3. In the **Value** column, enter the `ptt` flag for **Number** 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 and 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 star codes in the number aren’t processed, and the result may not be desirable.

You can include the star codes to be processed by the phone by enclosing them in a pair of brackets [...] 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.

Busy Lamp Field (BLF)

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 service provider on your phone, 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. Assign as many BLF keys as there are feature keys available.

Assign BLF to a Feature Key

When a specific SP service on the phone supports BLF, 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**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Busy Lamp Field** from the drop-down menu for **Function**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure the Speed Dial Number for BLF

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**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under the key that you defined with the **Busy Lamp Field** function, clear the check box for **Number** in the **Default** column.
3. In the **Value** column, enter the number, extension, or userid attribute associated with the assigned function for **Number**.
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*N*** > **SIP**.
2. Under **Feature Configuration**, clear the check box for **X_DirectedCallPickupMethod** in the **Default** column.

3. In the **Value** column, select one of the following methods for **X_DirectedCallPickupMethod**.

Parameter	Description
Feature Code	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.
INVITE+Replaces	Your phone sends an INVITE with a Replaces header that identifies to the softswitch the ringing call to pick up from the monitored extension. Used with MetaSwitch.

4. Select **Submit**.
5. 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*N*** > **SIP**.
2. Under **Feature Codes**, clear the check box in the **Default** column for **BargeIn**.
3. In the **Value** column, enter a feature code for **BargeIn**.

   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.
4. Select **Submit**.
5. 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*N*** > **SIP**.
2. Under **Feature Codes**, clear the check box for **CallPickup** in the **Default** column.
3. In the **Value** column, enter a feature code for **CallPickup** for 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.

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

Configure Floating BLF Key Assignment

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

Procedure

1. In the system web interface, do one of the following:
   - Go to **IP Phone**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under the key that you defined with the **Busy Lamp Field** function, clear the check box in the **Default** column for **Number**.
3. In the **Value** column for **Number**, 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.

   `Number = sales-team/`
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure BLF with the PDMS-SP Service

Configure BLF to work with the PDMS-SP service.

The phone monitors 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 applies the following operations to the monitored OBi number:

- Call (Call the monitored PDMS-SP 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**.
2. Under **Calling Features**, clear the check box in the **Default** column for **CallStatusAggregator**.

3. In the **Value** column, enter an OBi number for **CallStatusAggregator** that aggregates the call status for OBi devices.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring System Settings

Topics:

- Ringtones
- Configuring Star Codes
- Customizing Softkey Sets
- Feature Keys
- Set Audio Codec Priority

Customize system settings for your phones using applications and various keysets.

Ringtones

Ringtones provide audible call progress indicators to the user.

Set a Custom Tone Pattern

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. 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* at the Poly Online Support Center.

Procedure

1. In the system web interface, go to **Tone Settings** > **Tone Profile *X*** .
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* at the Poly Online Support Center.
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** > **SP*N* Service** > **Calling Features**.
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** > **SP*N* Service**.
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.

For more information on star codes, see the *Poly OBi Device Technical Reference* at the [Poly Online Support Center].

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.

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

Program star codes for any enabled feature in addition to the default star codes available.

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. 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.

For more information on softkey specifications, see the *Poly OBi Device Technical Reference* at the Poly Online Support Center.

Assign Softkeys

Assign softkeys to the configurable soft key sets.

Procedure

1. In the system web interface, go to **IP Phone** > **Soft Keys**.
2. In the **Default** column, clear the check boxes then assign softkeys in the **Value** column for the following settings.

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 on hold.	`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 conference participant.	`end`
ConfRinging	Second conference participant ringing.	`end,,conf.now`
ConfConnected	Connected with second conference participant.	`end,hold,resume,conf.now`
ConfBridgeControl	Connected with an external conference bridge.	`end,deaf,mute`
Conferencing	Merged conference call item.	`end,split,newcall`

Parameter	Description	Softkey Examples
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.

Feature Keys

Configure feature keys on your phone to perform predefined functions.

Configure virtual line keys and programmable keys.

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 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.

Preassigned Feature Keys (PFKs)

Poly Edge B Series 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	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.

Name	Default Function	Description
Hold	Hold	Hold a call.

Assign Feature Keys

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**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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

Parameter	Description	Notes
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 `Number` 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.	

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

Configure Feature Keys

Configure each feature key to perform various functions.

Procedure

1. In the system web interface, do one of the following:
   - Go to **IP Phone**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

<table>
<tr><th>Feature Key Function</th><th>Description</th></tr>
<tr><td>Call Appearance</td><td>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:
• <code>Service</code>: Optional. The service or line with which to bind the key
• <code>MaxCalls</code>: Number of calls managed by this call key</td></tr>
<tr><td>Line Monitor</td><td>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:
• <code>Service</code>: Required. The service or line to monitor</td></tr>
</table>

Feature Key Function	Description
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
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

Feature Key Function	Description
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@examplecompany.com`, you may specify `abcd` for the **Number** field to monitor this buddy, provided there's no other JID in the buddy list that starts with abcd.

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 with`Number`: 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}: 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: 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: 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. 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 can't 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 can't 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 can't 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 is there is at least one call in the Holding state, or turned off otherwise. This function can’t be bound to any specific voice service. Parameters: None
Join/Leave Page Group *N*	Join/Leave Page Group 1 to 10. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. Users can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise they're just listening). If you enable this feature, users can also "Clamp On" the feature key to talk continuously without needing to hold down the key. This function can’t 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 only sends 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.

Feature Key Function	Description
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.
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 be bound to any specific voice service. Parameters: None **Note:** You can disable the multiple Line Key Tabs feature 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

Feature Key Function	Description
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 times out as user enters the target number.
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

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

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, do one of the following:
    - Go to **IP Phone**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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.

Parameter	Description
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.

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

Set Audio Codec Priority

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.

Assign each of the SP and PDMS-SP 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 web page has a higher priority.

Procedure

1. In the system web interface, go to **Codecs** > **Codec Profile *X***.
2. Under the codec you want to set the priority for, clear the check box in the **Default** column for **Priority**.
3. In the **Value** column, enter a priority (1 is the highest) in the **Value** column.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring BroadSoft Server Features

Topics:

Configure Poly Edge B Series IP 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

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 APIs that correspond to an SP service on the phone.

Find these features under **Network Services per SP** service on the phone, and find 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

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**.
2. Under **GUI Menu**, clear the check box in the **Default** column for **NetServicesMenu1**.
3. In the **Value** column, enter a comma-separated list of user features for **NetServicesMenu1**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Call Recording Controls

Enable the call recording feature on a per SP service basis. 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**.
2. Under **Network Provided Services**, clear the check box in the **Default** column for **CallRecording**.
3. In the **Value** column, select the check box for **CallRecording**.
4. Select **Submit**.
5. 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

Enable disposition codes to enable call agents to classify calls during or after a call.

An agent can enter a disposition code for the current call that is still ongoing or for the last call that has just ended. The option only applies to calls on the same SP service. 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 it enters and submits the code.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. In the **Default** column, clear the check box for **DispositionCode**.

3. In the **Value** column, select the check box for **DispositionCode**.
4. Select **Submit**.
5. 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

Enable phones to use the call trace function for calls on the same SP service.

Procedure

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

Define Speed Dial Function Key for Call Trace

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**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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

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

Enable the Escalation Feature

Enable the escalation feature for calls on the same SP service.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.

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

Enable the Call Center Feature

Enable the call center feature to 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 displays 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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **CallCenter** in the **Default** column.
3. In the **Value** column, select the check box for **CallCenter**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

BroadSoft Guest Login/Logout (Hoteling)

Set up the phone for a guest to use it temporarily, such as a visiting employee or temp worker in a hot desk environment.

Consult BroadSoft documentation on how to administer this feature on the server

Enable the Hoteling Feature

After you enable hoteling and the phone achieves 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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **Hoteling** in the **Default** column.
3. In the **Value** column, select the check box for **Hoteling**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Define a Hoteling Function Key

Define a feature key with the **Hoteling** function but with only 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**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Hoteling** in the drop-down menu for **Function**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure the Expires Value of the Subscription

Configure the expires value of this subscription for each ITSP profile.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. Under **Feature Configuration**, clear the check box for **X_BWHotelingSubscribeExpires** in the **Default** column.
3. In the **Value** column, enter the interval (in seconds) in the **X_BWHotelingSubscribeExpires** field for when the renewal subscription expires.
4. Select **Submit**.
5. 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.

Activate BroadSoft AS-Features

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

Activating the BroadSoft AS-Feature lets you configure 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** > **SP*N* 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, so set the `expires` value of the subscription dialog to initiate the subscription renewal for each SP service feature enabled.

When a setting is changed, the server updates the Edge B Series IP phones with a NOTIFY that specifies the latest settings of the affected features.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **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.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Call Forwarding for All Calls

Activate this feature to forward all incoming calls.

The functionality provided by **Call Forward All** is similar to that of the **CallForwardUnconditional** function provided natively by the phone on a per-line basis. 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 **Voice Services** > **SP*N* Service**.
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**.

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

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

Enable Call Forwarding for Busy Lines

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** > **SP*N* Service**.

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

Enable Call Forwarding on No Answer

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **CallForwardNoAnswer** in the **Default** column.
3. In the **Value** column, select the check box for **CallForwardNoAnswer**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Do Not Disturb

Enable Do Not Disturb (DND) on your third-party provider.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* 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**.
5. Reboot your system when you complete your changes.

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.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **ACDAgent** in the **Default** column.
3. In the **Value** column for **ACDAgent**, select the check box.
4. Select **Submit**.
5. 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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box in the **Default** column for **SecurityClass**.
3. In the **Value** column, select the check box for **SecurityClass**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Executive Call Filter

Make the **Executive Call Filter** option viewable and changeable from the phone.

Enable the Executive Call Filter

Enable users to view and change the **Executive Call Filter** option from the phone.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **Executive** in the **Default** column.
3. In the **Value** column, select the check box for **Executive**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Define an Executive Call Filter Feature Key

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

Note: 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**, then select **Right Line Keys** or **Left Line Keys**.
   - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Exec Filter On/Off** in the drop-down menu for **Function**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable the Executive Assistant Function

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **ExecutiveAssistant** in the **Default** column
3. In the **Value** column, select the check box for **ExecutiveAssistant**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Define an Executive Assistant Feature Key

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**, then select **Right Line Keys** or **Left Line Keys**.
    - Go to **IP Phone** > **Programmable Keys**.
2. Under **Key *N***, clear the check box for **Function** in the **Default** column .
3. In the **Value** column, select **Exec Assistant** in the drop-down menu for **Function**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Call Recording

Enable call recording to allow the phone to 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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **CallRecording** in the **Default** column.
3. In the **Value** column, select the check box for **CallRecording**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

BroadSoft XSI Features

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

XSI features are available per SP or SIP service, so you can configure independent sets of XSI services per system and one per SP service.

Enable BroadSoft XSI Features

Enter the SIP credentials for the BroadSoft account to enable XSI features.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **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:
    a. For `X_XsiServer`, enter the hostname or IP address.
    b. For `X_XsiServerPort`, enter the server port.
    c. For `X_XsiServerScheme`, select **HTTP** or **HTTPS** from the drop-down menu.
4. Select **Submit**.
5. In the system web interface, go to **Voice Services** > **SP*N* 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:
    a. For `X_XsiUserName`, enter the user name to authenticate the XSI server.
    b. For `X_XsiPassword`, enter the password to authenticate the XSI server.
8. Select **Submit**.
9. Reboot your system when you complete your changes.

Enable Anonymous Call

Enable users to place calls anonymously from the phone.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **AnonymousCall** in the **Default** column.
3. In the **Value** column for **AnonymousCall**, select the check box.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Simultaneous Ring

After you enable Simultaneous Ring, users can access the feature from the **Net Services** application.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **SimultaneousRing** in the **Default** column.
3. In the **Value** column for **SimultaneousRing**, select the check box.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Enable Remote Office

After you enable Remote Office, users can access the feature from the **Net Services** application.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **RemoteOffice** in the **Default** column.
3. In the **Value** column for the `RemoteOffice` parameter, select the check box.
4. Select **Submit**.
5. 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 BroadWorks Anywhere

When you enable BroadWorks Anywhere, users can access the feature from the **Net Services** to launch the feature on the phone.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **BroadWorksAnywhere** in the **Default** column.
3. In the **Value** column, select the check box for **BroadWorksAnywhere**.
4. Select **Submit**.
5. 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.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **Directory** in the **Default** column.
3. In the **Value** column, select the check box for **Directory**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configure Voice Service Network Directories Service

Use the `VoiceService` parameter to control which SP service's network directories service the phone launches.

Procedure

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

Enable NetDir on the Phone's Menu

Enable the phone to 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**.
2. Under **Network Directory**, clear the check box for **Enable** in the **Default**.
3. In the **Value** column, select the check box for **Enable**.
4. Go to **Voice Services** > **SPn Service**.
5. Under **Network Provided Services**, clear the check box for **Directory** in the **Default**.
6. In the **Value** column, select the check box for **Directory**.
7. Select **Submit**.
8. Reboot your system when you complete your changes.

Disable the Search Option

Disable the **Search** option for the network directory.

This configuration 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** > **SP*N* Service**.
2. Under **Network Directory Setup**, clear the check box for **EnableSearch** in the **Default** column.
3. In the **Value** column, clear the check box for **EnableSearch**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring Other PBX Platforms

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

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 doesn'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`

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.

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 on the phones to enable users to search LDAP directories.

Procedure

1. In the system web interface, go to **IP Phone** > **LDAP** > **Search Parameters**.
2. Under **Server**, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

Parameter	Description
Host	Enter the host name, which can be an IP address or domain name, with an optional `ldap://` or `ldaps://` scheme prefix. For example, `192.168.15.186, ldap.forums.com` or `ldap://ldap.testathon.net` are acceptable host name formats. **Note:** If you don't specify the scheme, the phone uses `ldap://`.
Port	Enter the LDAP server listening (TCP) port. The standard port is 389 for `ldap://` and 636 for `ldaps://`. **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

Configure LDAP search settings.

Procedure

1. In the system web interface, go to **IP Phone** > **LDAP** > **Search Parameters**.
2. Under **Search Parameters**, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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 `admin@ldap.example.com`. For this example, the corresponding BindDN is: `CN=admin,OU=users,DC=example,DC=com`. If none is specified, the query is regarded as anonymous, which may or may not be acceptable to the server.
SearchBase	Use this parameter 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 `Host` 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 never or demand. The default value is never, which means the client doesn'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

cConfigure Simple Authentication and Security Layer (SASL) to send 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** > **Search Parameters**.
2. Under **LDAP SASL Authentication Parameters**, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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, configure how your phones display LDAP search options.

Specify LDAP Search Fields

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. Under **Search Parameters**, clear the check box for **QueryFields** in the **Default** column.
3. In the **Value** column, , enter a comma-separated list of user input LDAP attributes for **QueryFields**.

```
givenName/First Name, sn/
Last Name=s*,telephoneNumber/Tel,mobile/Mobile,homePhone/Home
```

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

Configure the LDAP Attributes for Search Results

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. Under **Search Parameters**, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

Parameter	Description
NameFieldPreference	Enter a comma-separated list of LDAP attributes to be used as the Name to display on the screen, ordered by preference.
NumberFieldPreference	Enter a comma-separated list of LDAP attributes to be used as the Number to display on the screen, ordered by preference.

Parameter	Description
PhotoFieldPreference	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

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. Under **Search Parameters**, clear the check box for **ResultFields** in the **Default** column.
3. 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.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Configuring LDAP Application Launch Options

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

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**.
2. Under **GUI Menus**, clear the check box for **MainMenu1** in the **Default** column.
3. In the **Value** column, enter one of the following value strings for **MainMenu1**:
    - directories
    - calls
    - netdir
    - call-histories
    - all-preferences
    - settings
    - prod-info
    - messages
    - buddy
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Set Up LDAP Application Launch Using the Network Directory Service

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 set **MainMenu1** to netdir. See Set Up LDAP Application Launch in the Directories List on page 106.

Procedure

1. In the system web interface, go to **IP Phone** > **Phone Settings**.
2. Under **Network Directory**, clear the check boxes in the **Default** column for **Enable** and **VoiceService**.
3. In the **Value** column, configure the following parameters:
    - `Enable`: Select the check box.
    - `VoiceService`: Select **LDAP** .
4. Select **Submit**.
5. 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**.
2. Under **Soft Key Sets**, clear the check box for softkey set you want to modify.
3. 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`.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Customizing the Local Interface

Topics:

- Define an LED Pattern
- Customize the Local Interface Menu
- Select User Preferences Settings from Internal Data Storage Paths

There are a number of ways to customize the appearance of Poly Edge B Series IP phones, such as customizing LED patterns or creating background images.

Define an LED Pattern

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

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 you don't specifiy `display-text` for an item, the phone uses the default displayed text.

Procedure

1. In the system web interface, go to **IP Phone** > **Phone Settings**.
2. Under **GUI Menus**, clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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*	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*	Enter menu items to display on each of the **Settings** menus. For example: `net;Network,flash;Storage,ringfile;Ringtones`
ProductInfoMenu*N*	Enter menu items to display on each of the Product menus. For example: `ipaddr,model,obnumber;OBi Number,mac;MAC Address,wfmac;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,rmoff,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`
DeviceAdministration Menu1	Enter menu items to display on the **Device Administrator** menu. For example: `admin.port,admin.adminPassword,admin.userPassword,admin.syslogServer,prov.itsp.Method,prov.itsp.Inteval,prov.itsp.ConfigURL,fwup.Method,fwup.Interval,fwup.FirmwareURL`
FactoryResetRequired AdminPassword	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 user interface messages with custom messages.

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

Select User Preferences Settings from Internal Data Storage Paths

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, each with its own dedicated internal storage areas:

- OBi Built-in
- ITSP (or Administrator) Customized
- User Customized

Procedure

1. In the system web interface, go to **User Settings** > **User Preferences**.
2. Clear the check boxes in the **Default** column for the following settings, then configure the settings in the **Value** column.

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: `/scratch/phone/dict/` • ITSP customized: `/scratch/itsp/dict/` • Phone built-in: `/data/dict/` **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.
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.

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

System Maintenance

Topics:

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 phones.

Reporting

Poly phones provide 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

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*N*** > **SIP**.
2. In the **Default** column, clear the check box for **X_InsertRTPStats**.
3. In the **Value** column, select the check box for **X_InsertRTPStats**.
4. Select **Submit**.
5. 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.

For example: `X-RTP-Stat:PS=1234,OS=34560,PR=1236,OR=24720,JI=1,DU=1230,PL=0,EN=G711U, DE=G711U`

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

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 are defined in RFC 3611, are also supported. RTCP-XR transmits VQ reports for a call.

Enable RTCP and RTCP-XR

Configure RTCP and RTCP-XR to enable the phone to send VQ reports 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 in the Background

Update your phone's firmware in the background.

Once enabled, the phone handles the firmware update, 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.

Use the Background Firmware Update Feature

After you enable the phone to update the firmware in the background, the device only uses 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**.
2. Under **Auto Firmware Update**, clear the check box for **FirmwareURL** in the **Default** column.
3. In the **Value** column, enter the URL of the firmware package for **FirmwareURL**.

   The following is an example of a correct URL: `http://192.168.1.1/Skyline-6-3-1-25001-devel-2018-12-13.fw`
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Capture Your Device's Current Screen

From the system web interface, 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

Reset all configuration parameters to factory default values or to the customized default values.

Factory Reset Your Phone in the Local Interface

Reset all phone settings to factory default values from the phone's local interface.

Procedure

1. On the phone, go to **Settings**.
2. Press the **Factory Reset** softkey.
3. Press the **OK** softkey.

Factory Reset Your Phone in the System Web Interface

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:

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

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.

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.

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** > **SP*N* Service**.
2. Under **Network Provided Services**, clear the check box for **CallLogs** in the **Default**.
3. In the **Value** column, select the check box for **CallLogs**.
4. Select **Submit**.
5. Reboot your system when you complete your changes.

Activate Syslog Messaging

Enable your Poly Edge B Series IP system to send syslog messages for troubleshooting.

Procedure

1. In the system web interface, go to **System Management** > **Device Admin** > **Syslog**.
2. In the **Default** column, clear the check boxes for **Server**, **Port**, and any **Level** settings you want to modify.
3. In the **Value** column, enter the hostname, FQDN, or IP address for your syslog server for `Server`.
4. Set the `Port` to 514.

5. Optional: For the **Level*X*** settings, choose options from the drop-down menus for the reporting levels you require.
6. Optional: For the **Reporting*X*** settings, configure the options to enable the system to periodically upload buffered syslog files to a web server.
7. Select **Submit**.
8. Reboot your system when you complete your changes.

Include Detailed SIP Messages in Syslog Messaging

Include detailed SIP messages in your syslog messages.

Procedure

1. In the system web interface, go to **Voice Services** > **SP*N* Service**.
2. Under **Debug Options**, set the `X_SipDebugOption` parameter to the reporting level you require.
3. 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 SP*N* Service Status Messages

View the current state of your configured voice services for troubleshooting issues with SIP-based services.

If a problem exists with the registration or authentication of your system with a prescribed service, a SIP 4xx error message displays.

Procedure

» In the system web interface, go to **Status** > **System Status** > **SP*N* Service Status**.

The status for this service displays, including any error messages.

SP*N* Service Status Error Messages

The following table lists some of the SP*N* 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.

Error Message	Description
406 Not Acceptable	The request can't be processed due to a requirement in the request message.
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.

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. 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 Profile*N*** > **SIP**.
2. Clear the check box for **X_RegisterIncludeInstance** in the **Default**.
3. In the **Value** column, clear the check box **X_RegisterIncludeInstance**.
4. Select **Submit**.
5. 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** > **SP*N* Service**.
2. Clear the check box for **X_UserAgentPorts** in the **Default** column.
3. In the **Value** column, enter as many as 10 alternative SIP UDP ports for **X_UserAgentPorts**.

   You must separate each SIP user agent port by a comma.
4. Select **Submit**.
5. 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 redirects 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

Configure REGISTER rules on the actual error code.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. Clear the check box for **X_RegisterRetryResponseCodes** in the **Default** column.
3. 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.
4. Select **Submit**.

5. Reboot your system when you complete your changes.

Configure REGISTER Retry Value

Configure the REGISTER retry value with the `RegisterRetryInterval` parameter.

Procedure

1. In the system web interface, go to **Service Providers** > **ITSP Profile*N*** > **SIP**.
2. Clear the check box for **RegisterRetryInterval** in the **Default** column.
3. In the **Value** column, enter the number of seconds for **RegisterRetryInterval** 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.
4. Select **Submit**.
5. 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,you 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.