



**ADMINISTRATOR GUIDE**

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# **Poly VVX Business IP Phones, OBi Edition**

## **VVX 150, VVX 250, VVX 350, VVX 450**

### **GETTING HELP**

For more information about installing, configuring, and administering Poly products or services, go to [Poly Support](#).

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# Before You Begin

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

## Audience, Purpose, and Required Skills

This guide is for a technical audience. You must be familiar with the following concepts:

- Current telecommunications practices, protocols, and principles
- Telecommunication basics, video teleconferencing, and voice or data equipment
- OpenSIP networks and VoIP endpoint environments

## Related Poly and Partner Resources

See the following sites for information related to this release.

- The [Polycom Support Site](#) is the entry point to online product, service, and solution support information including Licensing & Product Registration, Self-Service, Account Management, Product-Related Legal Notices, and Documents & Software downloads.
- The [Polycom Document Library](#) includes active product, service, and solution content in responsive HTML5 format, so that you can easily access and view installation, configuration, or administration content from any type of online device.
- The [Polycom Community](#) provides access to the latest developer and support information and forums. Participate in discussion forums to share ideas and solve problems with your colleagues. When logged in, you can access Poly support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.
- The [Polycom Partner Network](#) are industry leaders who natively integrate the Poly standards-based RealPresence Platform with its customers' current UC infrastructures making it easy for you to communicate face to face with the applications and devices you use every day.
- The [Polycom Collaboration Services](#) help your business succeed and be more collaborative—and get the most out of your investment through the benefits of collaboration.
- You can find Request for Comments (RFC) documents by entering the RFC number at <http://www.ietf.org/rfc.html>.
- For information on IP PBX and softswitch vendors, see Polycom [Desktop Phone Compatibility](#). If you're using the Poly Trio solution, see [Poly Trio and SoundStation IP Platform Compatibility](#).

## Notational Conventions

This guide provides device configuration parameters and their values in the following formats:

- Canonical fashion
- Literal fashion

Both notational conventions point to the same parameters, but their appearances are different.

The canonical fashion simplifies locating parameters on the phone's system web interface or on OBiTALK.com.

You must use the literal fashion when provisioning or writing OBiPhoneXML apps.

### Canonical Fashion

The following example shows the format of the canonical fashion.

- **Parameter Group Name**::ParameterName = Parameter Value

The **Parameter Group Name** is the heading of the parameter group on the left side panel of the device local interface or the OBiTALK configuration web page. This string may contain spaces. When a group heading has more than one level, separate each level with a –, such as:

- **Services Providers – ITSP Profile A – SIP:**

The `ParameterName` is the name of the parameter as shown on the web page and **MUST NOT CONTAIN ANY SPACES**. Separate the **Parameter Group Name** and `ParameterName` by two colons (::).

The `Parameter Value` is the literal value to assign to the named parameter and may contain spaces. You can omit **Parameter Group Name** or its top-level headings when the context is clear. For example:

- **SP1 Service**::AuthUserName = 4082224312
- **ITSP Profile A - SIP**::ProxyServer = sip.myprovider.com
- ProxyServerPort = 5082

### Literal Fashion

The following examples show the format of the literal fashion. Use the literal fashion when provisioning or writing OBiPhoneXML apps.

- **ParameterGroupName**.ParameterName.Parameter Value
- **Parameter.Group.Name**.ParameterGroupName.ParameterName.Parameter Value

The **ParameterGroupName** is the name of the first parameter group in literal fashion. This string **MUST NOT CONTAIN ANY SPACES**, and always ends with a period. You can use more than one **ParameterGroupName**. The **ParameterGroupName** is case-sensitive.

The `ParameterName` is the name of the parameter, and always ends with a period. This string **MUST NOT CONTAIN ANY SPACES**. The `ParameterName` is case-sensitive.

The `Parameter Value` is the literal value assigned to the named parameter and may contain spaces. The `Parameter Value` isn't case-sensitive, but it **MUST EXACTLY MATCH** the value when one or more choices are available.

When using the literal fashion in your XML, you must exactly match the text string for ***ParameterGroupName***.ParameterName.Parameter Value. However, the system removes any text formatting (such as bold or italics) when it processes your script or app.

## ***Boolean Values***

Parameters that take a Boolean value have a check box next to the parameter name on the system web interface. Throughout the document, we refer to a Boolean value as *enable or disable* or *yes or no*, but you must use `true/false` or `True/False` (case-sensitive) values for Boolean parameters in a phone configuration file. This is equivalent to selecting or clearing the check box in the system web interface.

## ***Multiple Choice Values***

If your device uses parameter values contained in a drop-down menu, you must provision those parameters with string values that match exactly one of those choices. Otherwise, the device uses the default choice.

Matching the provisioned value against valid strings is case-sensitive and doesn't allow extra spaces.

## ***Parameter Values***

When entering a parameter value from the system web interface or via provisioning, avoid adding extra white spaces before or after the parameter value. If the value is a comma-separated list of strings or contains attributes after a comma or semicolon, avoid adding extra white spaces before and after the delimiter.

For example: `CertainParameter = 1,2,3,4;a;b;c`

If a parameter value can include white spaces, such as `X_DisplayLabel`, use just a single space and no extra space before and after the value.

For example: `X_DisplayLabel = My New Service`

## ***XML Usage***

When you write or edit XML for your VVX business IP phone, use an XML editor that automatically checks your syntax.

# Getting Started

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The VVX business IP phones, OBi Edition, support Polycom HD Voice technology. You can manage the phone's local interface and the network interaction on VVX business IP phones, OBi Edition, directly from your phone or through the system web interface.

## Product Overview

VVX business IP phones, OBi Edition, implement the following features and functionalities:

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

## Supported Phones and Accessories

The following table lists the product names and features for VVX business IP phones, OBi Edition.

### VVX OBi Edition Products and Accessories

Product Name	Features
VVX 150 business IP phone	Compact monochrome display
VVX 250 business IP phone VVX 350 business IP phone VVX 450 business IP phone	High-resolution color active-matrix TFT LCD display with a customizable UI
5 VDC power adapter	Powers your phone when Power over Ethernet (PoE) isn't available

**VVX OBi Edition Products and Accessories (continued)**

Product Name	Features
Polycom® WiFi wireless network adapter	Connects your phone to a Wi-Fi network (VVX 250, VVX 350, and VVX 450 only)
USB headset	(VVX 250, VVX 350, and VVX 450 only)
Polycom VVX EM50 Expansion Module	(VVX 450 only)

## Working with VVX Business IP Phones, OBi Edition

There are several ways to configure and manage VVX business IP phones, OBi Edition. Use the methods that best suit your deployment scenario.

### Input Methods

You can input information directly into your phone using different options in your phone's keypad.

### Phone Keypad

You can input digits or text directly from your phone's keypad. There are three input modes:

- **Numeric (123)**: Each key press inputs the corresponding digit 0 to 9, \*, and #.
- **Alphanumeric (abc)**: Each key press provides a set of ASCII characters to input.
- **Capitalized Alphanumeric (Abc)**: Same as alphanumeric mode, except the ASCII character options defaults to a capital letter.
- **IPv4 Address (IPv4)**: Same as numeric, but \* maps to a dot (.) and # maps to a colon (:) (to enter an IPv4 address).

You can change between input modes by pressing the **Mode** softkey (by default, the label is **Switch Mode**).

### Logging In to the Phone

You can access the OBi Edition VVX business IP phones and web portals using default account credentials:

- For user access, the default user name and password are `user` and `user`.
- For administrator access, the default user name and password are `admin` and `admin`.

If you don't have access to the phone or web portals using the default credentials, contact your service provider for the user name and password. Make sure that you have access to the correct administrator or user password before you attempt to log in to the OBi Device Management web page.



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.



## Phone Native Web Portal

Each phone has its own built-in portal site at `http://{ip-address}` where `{ip-address}` is the IP address of the phone. The native web portal offers configurable options and status information organized into a number of web pages. The default page is the **System Status** page.

Default user name and passwords are `user` and `user` for the user level, and `admin` and `admin` for the administrator level.

Administrator level has full access to all configuration parameters. The administrator can decide which parameters are hidden, read-only, or read-writable at the user level.

## Find the Phone IP Address

Each phone must have a valid IPv4 address to connect to the network and communicate with other devices or cloud-based services. A valid IP address is a value other than `0.0.0.0`.

To find your phone's IP address:

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

## Local Device Configuration

OBI Edition VVX business IP phones have an integrated device management web portal that you can access from any standard web browser. The device management web page is organized into sections to allow for a manageable and compartmentalized approach to configuring parameters available on your phone.

## Access the Device Management Web Portal

You can access and sign into the device management web page as a user or administrator.

### To access the Device Management web page:

- 1 Connect your phone to the LAN.
- 2 From the phone **Main** menu, select **Settings**.
- 3 Under **Settings**, the first item, **Network**, shows the IP address of the phone.
- 4 Enter the phone **IP Address** as the URL of the web site you want to visit in your web browser.
- 5 When prompted, enter your user name and password.

## Navigating the Device Management Web Portal

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

Each web page is divided into three frames:

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

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



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. However, you can restart the phone just once after you have made and submitted all the necessary changes on all the pages.

The following table lists the available device configuration web pages:

#### Device Configuration Web Pages

Configuration Page	Available Settings
Status	System Status Call Status Call History SP Services Stats
OBiWiFi Configuration	Wi-Fi Settings Wi-Fi Access Points
System Management	WAN Settings Auto Provisioning Device Admin Device Update
Service Providers	ITSP Profile A (repeated for ITSP Profile B, C, D, E, and F) <ul style="list-style-type: none"> <li>• General</li> <li>• SIP</li> <li>• RTP</li> </ul>
Voice Services	SP1 Service SP2 Service SP3 Service SP4 Service SP5 Service SP6 Service OBiTALK Service Auto Attendant Gateways and Trunk Groups

**Device Configuration Web Pages (continued)**

Configuration Page	Available Settings
IP Phone	Phone Settings Left Line Keys Right Line Keys — on right side of phone screen (available only on VVX 450, OBi Edition) Programmable Keys Side Car 1 Side Car 2 LED Settings Soft Keys LDAP Line Key Customization Screen Item Customization
Codec Profiles	Codec Profile A Codec Profile B
Tone Settings	Tone Profile A Tone Profile B
Ring Settings	Ring Profile A Ring Profile B
Star Codes	Star Code Profile A Star Code Profile B
User Settings	User Preferences Speed Dials User Defined Digit Maps

## ***OBiTALK Portals***

OBiTALK.com is a device management portal website that serves Polycom customers and service providers deploying OBi Edition devices. OBiTALK.com uses remote provisioning to manage OBi Edition devices. It stores or dynamically generates on demand a configuration file for each managed device and periodically checks in with the OBiTALK server for configuration updates.

There are two levels of management portals at OBiTALK.com: User and ITSP.

### **User Portal**

Users can add and manage one or more OBi Edition devices from an OBiTALK account. The portal has setup wizards that help users configure voice services on any of their devices. Users can also see the detailed status and current parameter values of their devices and change settings on multiple devices from within the portal. The user portal has an upper limit of 20 devices per portal instance.

The user portal also enables members to add people and associated VVX business IP phones, OBi Edition, to *circles of trust*. These circles of trust enable authorized users to share additional functionality.

## Device Management Platform Portal – for ITSPs

Service providers, system integrators, and value-added resellers deploying OBi Universal Adapters and IP Phones can use the Device Management Platform portal to manage all their OBi endpoints remotely, enabling fast and easy device deployment and maintenance.

Service providers and system integrators can add OBi Edition devices to the OBiTALK portal for secure and remote provisioning, real-time call status reporting and troubleshooting, device UI management, and more.

### ***Record Version Information***

Contact [partner@obihai.com](mailto:partner@obihai.com) to learn more about the OBiTALK DMP and how to sign up to receive access.

In case you need to contact Polycom technical support, record the following information for future reference:

- Phone models
- Software version
- Partner Platform

# Supported Network Configurations

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VVX business IP phones, OBi Edition offer two interfaces for networking: Ethernet (indicated as **WAN** in the local and system web interfaces) and Wi-Fi. You can use both network options at the same time, but Ethernet takes precedence over Wi-Fi.

## Configuring the External Ethernet Ports

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

### To configure the external Ethernet ports:

- 1 Access the system web interface and log in with admin credentials.
- 2 Go to **System Management > WAN Settings**.
- 3 Configure the following parameters:

#### External Ethernet Port Configuration

Parameter	Values
<i>WAN Settings – Switch Port::Speed</i> <i>WAN Settings – PC Port::Speed</i>	<ul style="list-style-type: none"><li>• Auto</li><li>• 100 Full</li><li>• 100 Half</li><li>• 10 Full</li><li>• 10 Half</li><li>• 1000 Full</li><li>• Disabled</li></ul> <p>Poly recommends using only the <code>Auto</code> or <code>Disabled</code> settings.</p>

## Configure the WAN Interface

The WAN interface on the phone refers to the internal Ethernet switch port connected directly to the phone processor. You can configure the following setting groups on the phone.

**WAN Interface Setting Groups**

Setting Group	Description
VLAN	The phones support VLAN tagging in compliance with 802.1p/q. If you enable <b>WAN Settings – Internet Settings::VLANEnable</b> , the phone tags outbound traffic according to the <code>VLANID</code> and <code>VLANPriority</code> 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. Enable this feature using the <b>WAN Settings – Internet Settings::LLDP-MED</b> parameter.
IP Address Assignment	The phones support different methods of acquiring an IP address assigned to its WAN interface. Configure the method using the <b>WAN Settings – Internet Settings::AddressingType</b> parameter, which can have one of the following values: <ul style="list-style-type: none"> <li>• <b>DHCP</b>: Request address assignment from a DHCP server.</li> <li>• <b>Static</b>: Use the statically assigned IP address, subnet mask, and default gateway from the <b>WAN Settings – Internet Settings::IPAddress</b>, <code>SubnetMask</code> and <code>DefaultGateway</code> parameters, respectively.</li> </ul>
DNS Servers	You can specify as many as two DNS servers to use with the WAN interface using <b>WAN Settings – Internet Settings::DNSServer1</b> and <code>DNSServer2</code> . Note that 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, and PoE Class (Power Draw).

## Configure the Wi-Fi Interface

The VVX 250, VVX 350, and VVX 450 business IP phones support Wi-Fi via the Polycom Wi-Fi wireless network adapter. Connect the adapter to one of the USB ports on the phone.

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

### Wi-Fi Interface Setting Groups

Setting Group	Description
IP Address Assignment	<p>The phones support different methods of acquiring an IP address assigned to its Wi-Fi interface. Configure the method using the <b>WiFi Settings – Basic Settings::AddressingType</b> parameter, which can have one of the following values:</p> <ul style="list-style-type: none"> <li>• <b>DHCP:</b> Request address assignment from a DHCP server.</li> <li>• <b>Static:</b> Use the statically assigned IP address, subnet mask, and default gateway from the <b>WAN Settings – Internet Settings::IPAddress</b>, <b>SubnetMask</b> and <b>DefaultGateway</b> parameters, respectively.</li> </ul>
DNS Servers	<p>You can specify up to two DNS servers to use with the Wi-Fi interface using <b>WiFi Settings – Internet Settings::DNSServer1</b> and <b>DNSServer2</b>. Note that 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.</p>

## DHCP Options

The phones support the following DHCP network options for both networking interfaces:

- 42
- 66
- 150
- 159
- 160
- 161

The options that the phone tries to extract from the DHCP offer is a comma-separated list of option numbers specified in the **WAN Settings – DHCP Client Settings::ExtraOptions** parameter. The phone doesn't recognize any other option numbers.

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

- \$DHCP OPT66
- \$DHCP OPT150
- \$DHCP OPT159
- \$DHCP OPT160
- \$DHCP OPT161

For example, the default value of **Auto Provisioning – ITSP Provisioning::ConfigURL** is `tftp://$DHCP OPT66/$DM.xml`.

## DNS Lookup

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

## Lookup Order

When there are multiple DNS servers available, the phone queries as many DNS servers as necessary to resolve a domain name. Use the **WAN Settings – DNS Control::DNSQueryDelay** parameter to insert a short delay between each query to stop once the phone receives a positive response. If you set the delay to 0, the phone queries all the DNS servers at the same time.

When there are DNS servers obtained from both DHCP and statically configured values, the phone queries all the servers of one type before moving to the other type. Use the **WAN Settings – DNS Control::DNSQueryOrder** parameter to control the order the phone queries the server types.

## Locally Configured DNS Lookup Table

You can define as many as 30 local DNS records in the phone configuration. Once you define the DNS records, you can enable the phone to search through these records before hitting the external DNS services when attempting to resolve a domain name. These records can be A or SRV records.

This feature is particularly useful when you want to enable proxy redundancy without using any DNS servers. Note that the only way to provide a list of redundant servers to the phone is through the use of DNS A or DNS SRV records.

## NTP Servers and Local Time

The phone keeps track of current time by querying NTP servers (using SNTP). You can configure as many as two NTP servers using the `NTPServer1` and `NTPServer2` parameters.

By default, the phone discovers the SNTP server using DHCP option 42, and discovered servers take precedence. To disable SNTP discovery in DHCP and use the configured SNTP servers, remove 42 from the DHCP extra option list.

The phone determines the local time by the local time zone set in the `LocalTimeZone` parameter. The phone queries the NTP servers once per hour to update the current time, and the phone uses its own local clock in between NTP refreshes.

You can enable Daylight Saving Time (DST) by enabling the `DaylightSavingTimeEnable` parameter. The phone automatically adjusts DST based on the start and end rules specified in `DaylightSavingTimeStart` and `DaylightSavingTimeEnd` parameters. You can configure the amount of time to adjust when daylight saving time is in effect using the `DaylightSavingTimeDiff` parameter.

You can also enable the phone to use SIP Signaling to tell time. When the phone renews registration with a SIP proxy server, the server may include a Date header in the response to the phone that indicates the current GMT time. The phone processes this time value the same way it does with the result from the NTP servers, if you enable **ITSP Profile X – SIP::X\_ProcessDateHeader**.



# Manually Configuring Phones

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There are several ways to configure and manage your phone. Use the methods that best suit your deployment scenario.

## Telephone-IVR-Based Local Configuration

VVX business IP phones, OBi Edition, have a built-in interactive voice response (IVR) for checking and setting a small but essential subset of configuration parameters. Configuration via the IVR is a legacy configuration method inherited from older OBi products that don't have a display.

The IVR is an instance of an automated attendant (AA). VVX business IP phones, OBi Edition, offer two instances of AA, referred to as AA1 and AA2. The IVR for configuration purposes is AA2, which we'll just refer to as the IVR to avoid confusion with AA1, which is the AA used to handle phone calls.

To invoke the IVR, pick up the phone, dial `***`, and follow the announced instructions. In order for the `***` number to work, make sure the digit map pattern `***` is included in the **Phone Settings::DigitMap** parameter, and the rule, `{***:aa2}` is included in the **Phone Settings::OutboundCallRoute** parameter. The standard (non-customized) default values of these parameters are, 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))
and
{ ([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 }
```

For the meaning of these values, see [User-Defined Digit Maps](#) and [Call Routing](#).

Some parameter changes require a reboot to take effect. Changes made from the OBiTALK Configuration web page trigger the phone to reboot automatically (after any current calls end).

Key Ahead: By pressing the appropriate button sequence on the telephone, you can barge into the next menu of the IVR or invoke a command without waiting for the previous announcement to end.

## Main Menu

The Main menu after starting the IVR is a list of operations that users can select by entering the corresponding 1-digit option number (as listed below):

### Integrated Voice Response Operations

Selection	Announcement	What Can You Do?
1	Basic Network Status Reads the IP address and DHCP status.	Press 0 to repeat the information.
2	Advanced Network Status Reads the primary and back-up DNS server, primary and back-up NTP server.	Press 0 to repeat the information.
3	DHCP Current Value Reads the current value and you have the option to change the value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information.
4	IP Address Current Value Reads the current value and you have the option to change the value. If you elect to enter a new value (static IP address), DHCP is disabled.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information.
5	Password Current Value Reads the current IVR password value and you have the option to change the value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information.
6	Please Wait (while OBi is checking for software update)... One of the following messages plays: <ul style="list-style-type: none"> <li>• <b>Software Update Available. Press 1 to update software</b></li> <li>• Software Update Not Available.</li> </ul>	If an update is available, press 1 to proceed with the update. The software update process starts as soon as you hang up the phone.  Warning: Once the software upgrade process starts, the device's power LED blinks rapidly. Please make sure the power and network cable stay connected to the unit until the process is complete.
8	Restore Factory Default	Press 1 to confirm device restore to factory default settings. Press # to return to device configuration menu. Press ## to exit IVR.
9	Reboot OBi Device	Press 1 to confirm device reboot. Press # to return to device configuration menu. Press ## or hang up to exit IVR.
0	Additional Options Access other configuration options of your phone.	Enter option followed by the # key.

## Additional Integrated Voice Response Operations Options

There are many additional options beyond the top-level Integrated Voice Response Operations options 1 – 9. Unlike the top-level options, however, the list of available additional options in Menu 0 isn't announced.

**To access the additional IVR options, do the following:**

- 1 Enter **\*\*\* 0 #**
- 2 Enter your desired option number, followed by **#**
- 3 Follow any additional prompts, as desired.

You must enter the corresponding option number (followed by a **#** key) to select the particular option. The following tables list the available additional options (grouped by function):

### Syslog Operation Options

Selection (Always Press “#” After Entering Selection)	Announcement	What Can You Do?
1	Firmware Version Reads the current value of the firmware version.	Press 0 to repeat the information. Press # to enter another configuration selection.
2	IVR Password Reads the current value of the IVR password.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
3	Debug Level Reads the current value of the debug level.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
4	Syslog Server IP Address Reads the current IP address of the syslog server.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
5	Syslog Server Port Reads the current value of the syslog server port.	Press 1 to enter a new value. Press 2 to set the default value of 514. Press 0 to repeat the information. Press # to enter another configuration selection.

**WAN Settings Options**

<b>Selection (Always Press “#” After Entering Selection)</b>	<b>Announcement</b>	<b>What Can You Do?</b>
20	DHCP Configuration Reads the current value of the DHCP configuration.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
21	IP Address Reads the current value of the IP address.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
22	Default Gateway Reads the current value of the default Internet gateway.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
23	Subnet Mask Reads the current value of the subnet mask.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
24	DNS Server (Primary) Reads the current value of the primary DNS server.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
26	NTP Server (Primary) Reads the current value of the primary NTP server.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

## SP1 Options

Selection (Always Press “#” After Entering Selection)	Announcement	What Can You Do?
100	Enable Service Provider One (SP1) Reads the current value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
101	Registration State of SP1 Reads the current value.	Press 0 to repeat the information. Press # to enter another configuration selection.
102	SP1 User ID Reads the current value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
167	SP1 Block Caller ID Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
168	SP1 Block Anonymous Call Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
172	SP1 Call Forward ALL – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
173	SP1 Call Forward ALL Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
174	SP1 Call Forward on Busy – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
175	SP1 Call Forward on Busy Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

**SP1 Options (continued)**

<b>Selection (Always Press “#” After Entering Selection)</b>	<b>Announcement</b>	<b>What Can You Do?</b>
176	SP1 Call Forward on No Answer – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
177	SP1 Call Forward on No Answer Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

**SP2 Options**

<b>Selection (Always Press “#” After Entering Selection)</b>	<b>Announcement</b>	<b>What Can You Do?</b>
200	Enable Service Provider Two (SP2) Reads the current value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
201	Registration State of SP2 Reads the current value.	Press 0 to repeat the information. Press # to enter another configuration selection.
202	SP2 User ID Reads the current value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
267	SP2 Block Caller ID Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
268	SP2 Block Anonymous Call Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

## SP2 Options (continued)

Selection (Always Press “#” After Entering Selection)	Announcement	What Can You Do?
272	SP2 Call Forward ALL – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
273	SP2 Call Forward ALL Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
274	SP2 Call Forward on Busy – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
275	SP2 Call Forward on Busy Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
276	SP2 Call Forward on No Answer – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
277	SP2 Call Forward on No Answer Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

**OBiTALK Service Options**

<b>Selection (Always Press “#” After Entering Selection)</b>	<b>Announcement</b>	<b>What Can You Do?</b>
900	Enable OBiTALK Service Reads the current value.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
901	Registration State of OBiTALK Reads the current value.	Press 0 to repeat the information. Press # to enter another configuration selection.
967	OBiTALK Block Caller ID Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
968	OBiTALK Block Anonymous Call Enable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
972	OBiTALK Call Forward ALL – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
973	OBiTALK Call Forward ALL Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
974	OBiTALK Call Forward on Busy – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
975	OBiTALK Call Forward on Busy Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.



**OBiTALK Service Options (continued)**

Selection (Always Press “#” After Entering Selection)	Announcement	What Can You Do?
976	OBiTALK Call Forward on No Answer – Enable / Disable	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.
977	OBiTALK Call Forward on No Answer Number	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

**Auto Attendant Options**

Selection (Always Press “#” After Entering Selection)	Announcement	What Can You Do?
80	Enable / Disable Auto Attendant.	Press 1 to enter a new value. Press 2 to set the default value. Press 0 to repeat the information. Press # to enter another configuration selection.

# Provisioning Phones

The OBiTALK Device Management Platform portal provides access to the configurations of the multiple endpoints in batch mode by defining XML base profiles that are applied to multiple units. When a base profile changes, all units using this base profile are updated automatically with the changes.

## Remote Provisioning

The phone can download a configuration file from a server located in the cloud or in the same enterprise. The configuration file may contain all the necessary parameter values for the phone to function normally. The file may also tell the device to download an additional configuration file from a different URL or to download a different firmware to replace the current one. The configuration file format and parameter naming conventions are proprietary to Poly but are common across all OBi Edition products.

There are currently two configuration file formats supported: full XML format and short XML format with the XML tags substituted with single-letter abbreviations. The XML structure and parameter naming convention closely follows TR-104. For a full description of the configuration file and parameter names, refer to the *Polycom VVX Business IP Phones, OBi Edition Deployment Guide*.

Similar to parameter groupings in device configuration web portal, parameters in the configuration file are grouped into a number of configuration objects for remote provisioning. In fact, you'll find a near one-to-one correspondence between these objects and their location within the configuration web pages.

To illustrate this, consider the web page SP1 Service, the SIP Credentials section:

### SIP credentials

#### SIP Credentials

Parameter Name	Value	Default	
AuthUserName	<input type="text" value="join.j.smith@gmail.com"/>	<input type="checkbox"/>	<a href="#">?</a>
AuthPassword	<input type="password" value="*****"/>	<input type="checkbox"/>	<a href="#">?</a>
URI	<input type="text"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_MyExtension	<input type="text" value="16188"/>	<input type="checkbox"/>	<a href="#">?</a>
X_XsiUserName	<input type="text"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_XsiPassword	<input type="password"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_XmppDomain	<input type="text"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_XmppUserName	<input type="text"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_XmppPassword	<input type="password"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_ContactUserID	<input type="text"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>
X_EnforceRequestUserID	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<a href="#">?</a>

The corresponding configuration object in a phone configuration XML file is **VoiceService.1.VoiceProfile.1.Line.1.SIP**, as shown below:

```
<Object>
  <Name>VoiceService.1.VoiceProfile.1.Line.1.SIP.</Name>
  <ParameterValueStruct>
    <Name>AuthUserName</Name>
    <Value>john.j.smith@gmail.com</Value>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>AuthPassword</Name>
    <Value>zYz123#$12</Value>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>URI</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_MyExtension</Name>
    <Value>16188</Value>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_XsiUserName</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_XsiPassword</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_XmppDomain</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_XmppUserName</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_ContactUserID</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
  <ParameterValueStruct>
    <Name>X_EnforceRequestUserID</Name>
    <Value X_UseDefault="Yes"/>
  </ParameterValueStruct>
</Object>
```

Note that the dot (.) at the end of the object name is part of the name that cannot be omitted in the XML file. You must use the correct object name to create a valid configuration file for the phone.

## 3CX Plug-and-Play Provisioning

3CX plug-and-play provisioning lets the phone broadcast a SIP Subscribe to the Event: ua-profile on each reboot. With 3CX enabled, the phone also listens to SIP Notify of the same Event to extract from it the provisioning URL to request configuration, but only if the current **System Management – Auto Provisioning – ITSP Provisioning::ConfigURL** parameter is still set to the default value. To enable 3CX plug-and-play provisioning, change the **System Management – Auto Provisioning – System Info::X\_Enable3CXPNP** parameter to true on the device web page.

## Zero-Touch Provisioning

Zero-Touch or ZT provisioning is a system level approach to deploying and maintaining thousands or millions of devices with high security and control at the device level down to the individual parameter provisioned on each device.

To enable ZT provisioning, enable the **System Management – Auto Provisioning – OBiTalk Provisioning::ZeroTouch** parameter. Then customize the **System Management – Auto Provisioning – OBiTalk Provisioning::ConfigURL** parameter, which tells the phone where to download a configuration file. With these parameters configured, the first time a new phone is powered on and connected to the network, it can automatically contact the designated URL to get the initial configuration file.

For more information on using ZT provisioning, contact your Poly sales representative.



Zero-Touch devices must contact OBiTALK.com one time to get the customized values before they can start normal operation. Make sure that the device can access the internet before first use.

# Configuring Security Options

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

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

## 802.1X Authentication

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

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

You can set the authentication mode using the **WAN Settings – Internet Settings::802\_1XMode** parameter. Depending on the selected mode, you have to configure the additional authentication parameters listed in the following table.

### 802.1X Authentication

Parameter	Description	(EAP) MD5	(EAP) TLS (1.0)	TTLS/MSCHAPv2
<b>802_1XIdentity</b>	A user name. If the value isn't needed, set the value as an empty string.	Required	Required	Required
<b>802_1XPassword</b>	A password or passphrase. If a password or passphrase isn't needed, set the value as an empty string.	Required	Required	Required
<b>802_1XAnonymousID</b>	When empty, anonymous identity isn't used in authentication.		Required	Required
<b>802_1XTLSecurityProfile</b>	Security profile for the 802.1X authentication.		Required	

## HTTPS Support

VVX OBi Edition software 6.4.0 and later supports *https* in a web browser. Under **System Management > WAN Settings > Web Server**, set the value for Port to 443.

## Minimum Password Length Support

The service provider can now specify the minimum length of the password. You can mention the number and select the check box under **System Management > WAN Settings > Web Server**.

If you use the default password, a warning message *Default password is used* appears at the top right corner of the page. Change this password and record it in a safe place. If you lose this password, you may need to return your phone for service.

# Phone Display and Appearances

The phone has built-in skins that dictate the look and feel of the local interface. You can download a custom skin onto the target phone. You can request an additional document with local interface customization details from Poly.

## LED Pattern Customization

Feature Key LED patterns for each well-defined state can be customized using the LED Settings in the configuration. Each customizable pattern is configured in 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} part is the number of milliseconds to show the given color. If {Duration} is not specified, the LED stays at the given color indefinitely. The entire pattern is played repeatedly from left to right indefinitely until the state changes. Here are some examples:

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

## LED Settings Parameters

VVX business IP phones, OBi Edition, provide these LED setting parameters.

### LED Settings Parameters

Parameter Group	Parameter	Description
<b>Call State</b> For a bound or unbound call key, the LED pattern reflects the local call state.		
<i>IP Phone – LED Settings – Call State</i>	Idle	No call is active on the phone.
<i>IP Phone – LED Settings – Call State</i>	Dialing	The phone is dialing a number.
<i>IP Phone – LED Settings – Call State</i>	Trying	The phone is trying an outgoing call before the called party rings.
<i>IP Phone – LED Settings – Call State</i>	PeerRinging	The called party is ringing.

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – Call State</i>	<b>Connected</b>	The called party is connected.
<i>IP Phone – LED Settings – Call State</i>	<b>Ringing</b>	An incoming call is ringing.
<i>IP Phone – LED Settings – Call State</i>	<b>Holding</b>	The phone is holding the call.
<i>IP Phone – LED Settings – Call State</i>	<b>Error</b>	The call failed.
<i>IP Phone – LED Settings – Call State</i>	<b>CallParked</b>	A call is parked on the bounded extension.
<i>IP Phone – LED Settings – Call State</i>	<b>ServiceDown</b>	The service is down.

**SCA State**

For a call key that is bound to a shared line, the LED pattern reflects the Shared Call Appearance (SCA) state when the user of the SCA is on another phone.

<i>IP Phone – LED Settings – SCA State</i>	<b>Idle</b>	SCA is not used by anyone.
<i>IP Phone – LED Settings – SCA State</i>	<b>Seized</b>	SCA has been seized by another user.
<i>IP Phone – LED Settings – SCA State</i>	<b>Trying</b>	Another user is trying an outgoing call on the SCA.
<i>IP Phone – LED Settings – SCA State</i>	<b>PeerRinging</b>	The call peer is ringing.
<i>IP Phone – LED Settings – SCA State</i>	<b>Connected</b>	The call is connected.
<i>IP Phone – LED Settings – SCA State</i>	<b>Held</b>	The call is on hold.
<i>IP Phone – LED Settings – SCA State</i>	<b>PrivateHeld</b>	Another user is holding the call as private.
<i>IP Phone – LED Settings – SCA State</i>	<b>ServiceDown</b>	The SCA service is down.

**BLF State**

For a feature key that is assigned the **Busy Lamp Field** function, the LED reflects the state of the monitored entity.

<i>IP Phone – LED Settings – BLF State</i>	<b>Idle</b>	The monitored extension is idle.
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## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – BLF State</i>	<b>CallParked</b>	A call is parked against the monitored extension.
<i>IP Phone – LED Settings – BLF State</i>	<b>Ringing</b>	The monitored extension is ringing.
<i>IP Phone – LED Settings – BLF State</i>	<b>Busy</b>	The monitored extension is on a call.
<i>IP Phone – LED Settings – BLF State</i>	<b>Holding</b>	The monitored extension is holding a call.
<i>IP Phone – LED Settings – BLF State</i>	<b>ServiceDown</b>	BLF service is down.

**Service State**

For a feature key that is assigned the **Line Monitor** function, the LED pattern reflects the state of the bounding voice service.

<i>IP Phone – LED Settings – Service State</i>	<b>Idle</b>	No calls.
<i>IP Phone – LED Settings – Service State</i>	<b>InUse</b>	One or more calls is connected or trying.
<i>IP Phone – LED Settings – Service State</i>	<b>Ringing</b>	One or more incoming calls is ringing.
<i>IP Phone – LED Settings – Service State</i>	<b>Holding</b>	The monitored extension is holding a call.
<i>IP Phone – LED Settings – Service State</i>	<b>ServiceDown</b>	The service is down.

**ACD Agent State**

For a feature key that is assigned the **ACD Sign On or Off** function, the LED reflects the current ACD Agent state.

<i>IP Phone – LED Settings – ACD Agent State</i>	<b>LoggedOff</b>	The agent has logged off.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>Available</b>	The agent is logged on and available.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>Unavailable</b>	The agent is logged on but not available.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>WrappingUp</b>	The agent is logged on but is wrapping up the last call.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>ServiceDown</b>	The service is down.

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – ACD Agent State</i>	<i>LoggedOn</i>	The agent just logged on.
<b>Presence State</b>		
For a feature key that is assigned the <b>Presence Monitor</b> function, the LED reflects the presence of the monitored entity.		
<i>IP Phone – LED Settings – Presence State</i>	<b>Offline</b>	Invisible.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>Online</b>	Online or available.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>Busy</b>	Do Not Disturb (DND).
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>Away</b>	Away.
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>ExtendedAway</b>	Extended Away (XA).
<i>IP Phone – LED Settings – ACD Agent State</i>	<b>ServiceDown</b>	The service is down.
<b>Feature Key State</b>		
Miscellaneous feature key functions not covered in the previous fields.		
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AnonymousCallEnabled</b>	For a feature key assigned to the <b>Block Caller ID</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AnonymousCallDisabled</b>	For a feature key assigned to the <b>Block Caller ID</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AnonymousCallServiceDown</b>	For a feature key assigned to the <b>Block Caller ID</b> function, this LED pattern is displayed when the service that provides the feature is down, if the feature is provided by the ITSP.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AnonymousCallBlockEnabled</b>	For a feature key assigned to the <b>Block Anonymous Call</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AnonymousCallBlockDisabled</b>	For a feature key assigned to the <b>Block Anonymous Call</b> function, this LED pattern is displayed when the feature is disabled.

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AutoAnswerIntercomEnabled</b>	For a feature key assigned to the <b>Auto Answer Intercom</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>AutoAnswerIntercomDisabled</b>	For a feature key assigned to the <b>Auto Answer Intercom</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallForwardEnabled</b>	For a feature key assigned to the <b>Call Forward</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallForwardDisabled</b>	For a feature key assigned to the <b>Call Forward</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallForwardServiceDown</b>	For a feature key assigned to the <b>Call Forward</b> function, this LED pattern is displayed when the service providing this feature is down.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallParkYes</b>	For a feature key assigned to the <b>Call Park Monitor</b> function, this LED pattern is displayed when there is a call parked on that park orbit.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallParkNo</b>	For a feature key assigned to the <b>Call Park Monitor</b> function, this LED pattern is displayed when there is no call parked on that park orbit.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallParkMonitorServiceDown</b>	For a feature key assigned to the <b>Call Park Monitor</b> function, this LED pattern is displayed when the underlying call park monitoring service is down
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallWaitingEnabled</b>	For a feature key assigned to the <b>Call Waiting</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>CallWaitingDisabled</b>	For a feature key assigned to the <b>Call Waiting</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>DoNotDisturbEnabled</b>	For a feature key assigned to the <b>Do Not Disturb</b> function, this LED pattern is displayed when the feature is enabled.

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – Feature Key State</i>	<b>DoNotDisturbDisabled</b>	For a feature key assigned to the <b>Do Not Disturb</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>DoNotRingEnabled</b>	For a feature key assigned to the <b>Do Not Ring</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>DoNotRingDisabled</b>	For a feature key assigned to the <b>Do Not Ring</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecFilterEnabled</b>	For a feature key assigned to the <b>Exec Filter on or Off</b> function, this LED pattern is displayed when the feature is enabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecFilterDisabled</b>	For a feature key assigned to the <b>Exec Filter on or Off</b> function, this LED pattern is displayed when the feature is disabled.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecFilterServiceDown</b>	For a feature key assigned to the <b>Exec Filter on or Off</b> function, this LED pattern is displayed when the service providing this feature is down.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecAssistEnabled</b>	For a feature key assigned to the <b>Exec Assistant</b> function, this LED pattern is displayed when the assistant is not filtering calls for any executives (the executive list for this assistant is empty).
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecAssistDisabled</b>	For a feature key assigned to the <b>Exec Assistant</b> function, this LED pattern is displayed when the assistant is filtering calls for at least one executive.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecAssistDivertOn</b>	For a feature key assigned to the <b>Exec Assistant</b> function, this LED pattern is displayed when the assistant has turned on the Divert option.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>ExecAssistServiceDown</b>	For a feature key assigned to the <b>Exec Assistant</b> function, this LED pattern is displayed when the service providing the Exec Assistant feature is down.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>HotelingGuestLoggedIn</b>	For a feature key assigned to the <b>Hoteling</b> function, this LED pattern is displayed when a guest has logged on.

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – Feature Key State</i>	<b>HotelingGuestLoggedOff</b>	For a feature key assigned to the <b>Hoteling</b> function, this LED pattern is displayed when no guest has logged on.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>HotelingServiceDown</b>	For a feature key assigned to the <b>Hoteling</b> function, this LED pattern is displayed when the service providing this feature is down.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>NewMessagesWaitingYes</b>	For a feature key assigned to the <b>Message Status</b> function, this LED pattern is displayed when there are new messages in that mailbox.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>NewMessagesWaitingNo</b>	For a feature key assigned to the <b>Message Status</b> function, this LED pattern is displayed when there are no new messages in that mailbox.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>MWIServiceDown</b>	For a feature key assigned to the <b>Message Status</b> function, this LED pattern is displayed when the MWI service for that mailbox is down.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>PageGroupJoined</b>	For a feature key assigned to the <b>Page Group 1 to Page Group 10</b> function, this LED pattern is displayed when the phone has joined the group.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>PageGroupLeft</b>	For a feature key assigned to the <b>Page Group 1 to Page Group 10</b> function, this LED pattern is displayed when the phone has left the group.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>PageGroupMeTalking</b>	For a feature key assigned to the <b>Page Group 1 to Page Group 10</b> function, this LED pattern is displayed when you are talking to the group.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>PageGroupThemTalking</b>	For a feature key assigned to the <b>Page Group 1 to Page Group 10</b> function, this LED pattern is displayed when you are listening and someone else in the group is talking.
<i>IP Phone – LED Settings – Feature Key State</i>	<b>SecurityClassServiceDown</b>	For a feature key assigned to the <b>Security Class</b> function, this LED pattern when the service providing this feature is down.

## VMWI Lamp

## LED Settings Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LED Settings – VMWI Lamp</i>	<b>NewMessagesWaitingYes</b>	New messages are available for this phone from all mailboxes.
<i>IP Phone – LED Settings – VMWI Lamp</i>	<b>NewMessagesWaitingNo</b>	No new messages are available for this phone from all mailboxes.
<i>IP Phone – LED Settings – VMWI Lamp</i>	<b>Disabled</b>	VMWI is disabled.
<i>IP Phone – LED Settings – VMWI Lamp</i>	<b>Ringing</b>	Describes whether the LED should blink with the given pattern when the phone is ringing. Note that if the pattern is blank, ringing won't affect the VMWI Lamp.

## Local Interface Menu Customization

The parameters for customizing some local interface menus are summarized in the following table. 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 multiple instances of the same menu parameters are specified, the items are concatenated internally into a single item list. Items are displayed in the order they are specified. If `display-text` is not specified for an item, the default displayed text is used.

### UI Menu Customization

Parameter Group	Parameter	Description
<b>Main Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<b>MainMenu1</b>	For example: directories,calls,call-histories,all-p references,settings,prod-info,messages ,buddy
<b>Preferences Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<b>PreferencesMenu1</b>	For example: language;Language,timeFormat;Time Format,dateFormat;Date Format
<i>IP Phone – Phone Settings – GUI Menus</i>	<b>PreferencesMenu2</b>	
<i>IP Phone – Phone Settings – GUI Menus</i>	<b>PreferencesMenu3</b>	
<i>IP Phone – Phone Settings – GUI Menus</i>	<b>PreferencesMenu4</b>	

## UI Menu Customization (continued)

Parameter Group	Parameter	Description
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>PreferencesMenu5</i>	
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>PreferencesMenu6</i>	
<b>Settings Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>SettingsMenu1</i>	For example: net;Network,wifi;WiFi,flash;Storage,ringfile;Ringtones
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>SettingsMenu2</i>	For example: voice;Voice Services,sd99,clr-cache,admin;Device Admin,login
<b>Product Information Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>ProductInfoMenu1</i>	For example: ipaddr,model;Model,obinumber;OBi Number,mac;MAC Address,wfmac;WiFi MAC Address,serial;Serial Number
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>ProductInfoMenu2</i>	
<b>Net Services Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>NetServicesMenu1</i>	For example: regst,ringtone,acd,bac,bci,bwanw,buddy,ccs,cfa,cfb,cfna,rec,dnd,dispcode,exec,xass,hotel,clog,dir,rloff,secClass,simring
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>NetDirectoryMenu1</i>	For example: name,firstName,lastName,ext,number,email,groupId,department,mobile,userId,impId,presence
<b>Device Administration Menu</b>		
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>DeviceAdministrationMenu1</i>	For example: admin.port,admin.adminPassword,admin.userPassword,admin.syslogServer,prov.it.sp.Method,prov.it.sp.Interval,prov.it.sp.ConfigURL,fwup.Method,fwup.Interval,fwup.FirmwareURL

## UI Menu Customization (continued)

Parameter Group	Parameter	Description
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>FactoryResetRequireAdminPassword</i>	This parameter requires the user to enter the Web Administrator password to issue a factory reset from the phone. The default is True.
<i>IP Phone – Phone Settings – GUI Menus</i>	<i>CustomDictionary</i>	This parameter enables you to specify an XML dictionary to replace built-in GUI messages with custom messages.

## Main Menu Item IDs

Main Menu items include the basic set of menu choices on the Home screen, as well as custom proprietary items that service providers can provision on your phone.

## Main Menu Items

Main Menu Item ID	Default Caption	Description
phone-book	Contacts	Local phone book.
calls	Current Calls	List all the currently ongoing calls.
call-history	Call History	Locally stored call logs (for all calls on all services).
settings	Settings	List all the system level settings.
preferences	Preferences	Local phone feature preferences.
prod-info	Product Info	A list of data related to the product such as model and serial numbers, software and hardware versions.
netsrv	Net Services	Access the settings of many network provided features for each SP $n$ service, $n = 1 - 6$ .
netdir	Net Dir	Access the network directory associated with a specific SPI service or the LDAP service.
buddy	Buddy List	Access to the buddy list associated with a specific SP $n$ service.
directories	Directories	A container that lists the following options: <ul style="list-style-type: none"> <li>Local phone-book (same as Contacts)</li> <li>Network directory for each SP<math>n</math> service, when available, <math>n = 1 - 6</math></li> <li>LDAP Search (if configured)</li> </ul> When only the local phone-book is available, this item behaves the same as the phone-book item.
messages	Messages	This option shows a list where each entry is the message or mailbox status of a SP $n$ service, for $n = 1 - 6$ , if enabled.



**Main Menu Items (continued)**

Main Menu Item ID	Default Caption	Description
all-preferences	Preferences	A container that lists the following options: <ul style="list-style-type: none"> <li>Phone (same as preferences item)</li> <li>SP<math>n</math> service (or its label), for <math>n = 1 - 6</math>, if enabled</li> </ul> If none of SP $n$ preferences are enabled, then the item behaves exactly as the legacy preferences item.
call-histories	Call Histories	A container to access the locally stored call histories and the call history associated with each SP $n$ service, $n = 1 - 6$ .

**Net Services Menu Item IDs**

This table lists the menu items on the Net Services menu of the VVX business IP phones, OBi Edition. Item presence depends on subscribed network service.

**Net Services Menu Items**

Net Services Menu Item ID	Default Caption	Description
acd	ACD Agent Sign On/Off	
bci	Anonymous Call	Turn Anonymous Call feature on or off.
bwanw	BroadWorks Anywhere	
buddy	Buddy List	
ccs	Call Center Status	
cfa	Call Forward Always	
cfb	Call Forward Busy	
cfna	Call Forward No Answer	
rec	Call Recording Mode	
dnd	Do Not Disturb	
dispcode	Enter Call Disposition Code	
exec	Executive Call Filter	
xass	Executive Assistance	
hotel	Hoteling	
clog	Network Call History	
dir	Network Directory	
rmoff	Remote Office	

**Net Services Menu Items (continued)**

Net Services Menu Item ID	Default Caption	Description
secClass	Security Class	
simring	Simultaneous Ring	

**Preferences Menu Item IDs**

This table lists the menu items on the Preferences menu of the VVX business IP phones, OBi Edition. Menu items may vary depending on your phone.

**Preferences Menu Items**

Preferences Menu Item ID	Default Caption	Description
sidecar	SideCar Background Picture	Select a background picture for your side car (VVX 450 only)
language	Language	Select the Language of the displayed text on the phone screen.
timeFormat	Time Format	Select the format of the time display on the phone screen.
dateFormat	Date Format	Select the format of the date display on the phone screen.
hpTime	Auto Home Page	Select the amount of idle time before automatically returning the phone screen to the Home screen.
skin	Skin	Select the built-in UI scheme ("skin") to use.
bgpic	Background Picture	Select an available background picture to use as wall paper.
dring	Default Ringtone	Select an available ring tone as the default ring tone.
dfont	Font	Select an available font as the default font for the displayed text.
appCols	Home App Columns	Select the number of columns to display in the UI Main menu.
packcalls	Pack Calls On Display	Enable or disable packing calls on different services on the same screen of the display.
sdim	Dim Screen	Enable or disable dimming the screen (to the least settable brightness level) after the set interval of no-key-presses-and-no-calls.
sdimDelay	Dim Screen Delay	Interval of no-key-presses-and-no-calls in seconds before dimming the screen, if enabled.

## Preferences Menu Items (continued)

Preferences Menu Item ID	Default Caption	Description
ssvr	Screen Saver	Enable or disable the screen saver feature.
ssvrDelay	Screen Saver Delay	Amount of idle time in seconds before starting the screen saver, if enabled.
ssvrType	Screen Saver Type	Select a screen saver type.
ssvrLock	Require Passcode On Wake Up	Select whether a passcode is required to exit from the screen saver.
ssvrPass	Wake Up Passcode	Passcode to exit from screen saver.
	Screen Saver Show Custom Contents	
brightness	Screen Brightness	Set the brightness of the LCD screen.
audioDevice	Preferred Audio Device	Select whether to use speakerphone or a headset as the talk device when a headset is connected.
headsetDevice	Preferred Headset Device	Select the preferred headset device when more than one is connected.
ehs	Electronic Hook Switch	
dnd	Do Not Disturb	Enable or disable Do Not Disturb.
dnr	Do Not Ring	Enable or disable Do Not Ring.
cfa.enable	Call Forward	Enable or disable Call Forwarding (all calls, unconditionally).
cwa	Call Waiting	Enable or disable Call Waiting.
bac	Block Anonymous Call	Enable or disable Block Anonymous Caller.
bci	Anonymous Call	Enable or disable Block Caller ID (also known as Anonymous Call).
aans	Auto Answer Page	Enable or disable Auto-Answering incoming page.
pg1	Join Page Group 1	Join Page Group 1.
pg2	Join Page Group 2	Join Page Group 2.
pg3	Join Page Group 3	Join Page Group 3.
pg4	Join Page Group 4	Join Page Group 4.
pg5	Join Page Group 5	Join Page Group 5.
pg6	Join Page Group 6	Join Page Group 6.
pg7	Join Page Group 7	Join Page Group 7.
pg8	Join Page Group 8	Join Page Group 8.

**Preferences Menu Items (continued)**

Preferences Menu Item ID	Default Caption	Description
pg9	Join Page Group 9	Join Page Group 9.
pg10	Join Page Group 10	Join Page Group 10.
ringerVol	Ringer Volume	Ringer Volume.
speakerVol	Speakerphone Volume	Speakerphone Audio Volume.
micGain	Speakerphone Mic Gain	Speakerphone Mic Input Gain.
handsetVol	Handset volume	Handset Audio Volume.
handsetGain	Handset Mic Gain	Handset Mic Input Gain.
headsetRJ9Vol	RJ9 Headset Volume	Audio Volume for the RJ9 headset device.
headsetRJ9Gain	RJ9 Headset Mic Gain	Mic Input Gain for the RJ9 headset device.
EqEnable	Equalizer	Enable the Equalizer.
AecEnable	Acoustic Echo Cancellation	Enable the Acoustic Echo Canceler.
edialer	Enhanced Dialer	Enhanced Dialer (show matching list of recently dialed numbers while dialing).
dlkt	Default Line Key Tab	Default Line Key Tab (the Line Key Tab to fallback to after a configurable duration of idle time).

## Settings Menu Item IDs

This table lists the menu items on the Settings menu of the VVX business IP phones, OBi Edition. Some items require admin login.

**Settings Menu Items**

Settings Menu Item ID	Default Caption	Description
net	Network	Show current IP address and launch Network Settings menu.
wifi	WiFi	Show Wi-Fi connection status and launch Wi-Fi Settings menu.
flash	Storage	View, read, or copy contents in the internal user data storage or external USB flash drives.
ringfile	Ringtones	Set up the default ring tone per SP $n$ service.
pkeys	Programmable Keys	Launch Programmable Keys Settings menu.
lkeys	Line Keys	Launch Line Keys Settings menu.
voice	Voice Services	Launch Voice Services Settings menu.

**Settings Menu Items (continued)**

Settings Menu Item ID	Default Caption	Description
Sd99	Speed Dials 99	Set up 99 speed dial numbers.
clrcache	Clear Data Cache	Clear the downloaded data cache.
admin	Device Administration	Show Administrative Settings menu.
login	Unlock Admin Settings	Log in or out of administrative mode.

## Product Info Menu Item IDs

This table lists the menu items on the Product Info menu of the VVX business IP phones, OBi Edition. Some items require admin login.

**Product Info Menu Items**

Product Info Menu Item ID	Default Caption	Description
ipaddress	IP Address	Current IP address for this phone
model	Model	Model Number
obinumber	OBi Number	OBi Number
mac	MAC Address	(Ethernet) MAC Address
wfmac	WiFi MAC Address	Wi-Fi MAC Address
serial	Serial Number	Serial Number
swver	Software Version	Software Version
hwver	Hardware Version	Hardware Version
ztinfo	Customization Status	ZT Customization Status
uptime	Up Time	UP time since last bootup

## Cache Control of Downloaded (Temporary) Data

You can download the following data items that the phone caches at run time (in DRAM):

- Ring tone file downloaded from the URL specified in **User Preferences::DefaultRingtone**
- Picture file downloaded from the URL specified in **User Preferences::BackgroundPicture**
- Caller ID Picture files downloaded from URL extracted from Call-Info header of SIP Messages
- Picture files downloaded from URL in `src` attribute of `<img>` elements in rendered `<ScreenItem>`
- Var-Tree XML files downloaded from URL in `value` attribute of `<setvar>` elements in rendered `<ScreenItem>`
- Icon, Wave or MP3 files downloaded when rendering OBiPhoneXML apps

The expiration time of each individually cached data file is taken from the HTTP/Cache-Control header in the 200 response when the phone originally receives the file from the server. If the Cache-Control header is present in the 200 response with the no-cache flag specified, the phone doesn't cache the downloaded data file. If the Cache-Control header is present with the max-age attribute specified, phone caches the data file for the number of seconds as specified in the max-age attribute. Otherwise, the phone caches the data file indefinitely until one of the following happens that clears ALL the cached data:

- Phone power cycles
- Phone fully reboots (for example, after a firmware update)
- User selects **Settings > Clear Data Cache** on the local interface

## Phone Customization Data

You can customize some of the data that drives the local interface. These data are collectively called phone customization data, which include the following data types:

### Phone Customization Data Type

Customization Data Type	Description	Internal Data Storage Path
Background Pictures (Wall Paper)	The pictures that you can choose to show as wall paper during normal phone operation. The data files can be JPEG, PNG, BMP, or GIF.	/backgnd/
Dictionary	The translation to be used for various items shown on the phone screen.	/dict/
Text Fonts	The true-type font files for rendering text to display on the screen.	/fonts/
Ring Tones	Ring tone wave files. It must be in 16-bit mono linear PCM wave format sampled at 16 kHz.	/ringtones/

## Background Pictures

The phone stores custom background pictures in the internal folder `/backgnd/`.

Your VVX business IP phone, OBi Edition, can accept JPEG, PNG, BMP, and GIF file formats. You can choose any of the pictures stored in this folder as background pictures on the phone screen.

You can browse the **Background Picture** entries under the phone's **Preferences** menu and select one of the available background pictures. Each of the built-in "Skins" has its own default background pictures.

## Use a USB Flash Drive to Add a Personal Photo


Add personal photos to internal storage using a USB flash drive on VVX 250, VVX 350, and VVX 450 business IP phones, OBi edition.

For VVX 450 phones with connected VVX EM50 expansion modules, you can connect a USB flash drive to the side USB port on the expansion module.



The maximum resolution for images is 1024 × 768 pixels.

### To use a USB flash drive to add a personal photo

- 1 Attach a USB flash drive to the USB port on your phone.
- 2 Go to **Settings > Storage**.
- 3 Press the **>** softkey to select the USB flash drive.
- 4 Press the **Explore** softkey.
- 5 Do one of the following:
  - Highlight the file, and press the **Select** softkey.
  - Press the **Select All** softkey.
- 6 Press the **Copy** softkey.  
The phone returns to the internal destination folder.
- 7 Go to the phone's background folder.
- 8 Press the **Select**  key.
- 9 Press the **Paste** softkey.

## Fonts Customization

The phone stores fonts in the internal folder `/fonts/`. You can install custom font files on the phone only using the data package upload method. Acceptable font file formats are TTF and OTF.

To choose an installed font, you must also add a font-name-to-font-file for each installed font in the `font.map` file that is also stored in the same folder.

`font.map` maps the font name to the actual font file. The set of font names listed in `font.map` is also presented to you as a list of selections under **Preferences/Font** in the phone UI. On the web page the `DefaultFont` parameter is just a string and should match one of the font names (not file names) listed in `font.map`.

The `font.map` file lists all the factory-installed fonts that come with the phone.

```
#
# font.map
#
# The left column is a font name that a user can select from
# the Preferences menu. The right column is the corresponding
# physical font file installed on the phone.
# A valid font file must either a ttf or otf file.
```

```
#
# You can use any number of white spaces as delimiters between the 2 columns.
#
aleo                Aleo-Light.otf
aleo-bold           Aleo-Bold.otf
aleo-italic         Aleo-LightItalic.otf
aleo-bold-italic    Aleo-BoldItalic.otf
banksia            Banksia-Regular.otf
banksia-bold        Banksia-Bold.otf
droidsans           DroidSans.ttf
droidsans-bold      DroidSans-Bold.ttf
libre-caslon        LibreCaslonText-Regular.ttf
libre-caslon-bold   LibreCaslonText-Bold.ttf
libre-caslon-italic LibreCaslonText-Italic.ttf
opensans            OpenSans-Regular.ttf
opensans-bold       OpenSans-Bold.ttf
opensans-italic     OpenSans-Italic.ttf
opensans-bold-italic OpenSans-BoldItalic.ttf
quattrocento        Quattrocento-Regular.ttf
quattrocento-bold   Quattrocento-Bold.ttf
terminal-dosis      TerminalDosis-Regular.ttf
terminal-dosis-bold TerminalDosis-Bold.ttf
```

In addition, there are four more built-in fonts:

```
ptsans
ptsans-bold
ptsans-italic
ptsans-bold-italic
```

### Font Samples

Font	Sample
aleo	abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.,; ' " (!?) +-*/=
aleo-bold	<b>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</b> <b>1234567890.,; ' " (!?) +-*/=</b>
aleo-bold-italic	<i><b>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</b></i> <i><b>1234567890.,; ' " (!?) +-*/=</b></i>
aleo-italic	<i>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</i> <i>1234567890.,; ' " (!?) +-*/=</i>
banksia	abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.,; ' " (!?) +-*/=



## Font Samples (continued)

Font	Sample
banksia-bold	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>
droidsans	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>
droidsans-bold	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>
libre-caslon	abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=
libre-caslon-italic	<i>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</i>
libre-caslon-bold	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>
opensans	abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=
opensans-italic	<i>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</i>
opensans-bold	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>
opensans-bold-italic	<b><i>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</i></b>
quattrocento	abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=
quattrocento-old	<b>abcdefghijklmnopqrstuvxyz ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.;; ' " (!?) +-*/=</b>

## Font Samples (continued)

Font	Sample
terminal-dosis	abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.,; ' " (!?) +-*/=
terminal-dosis- bold	<b>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</b> <b>1234567890.,; ' " (!?) +-*/=</b>
ptsans	abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ 1234567890.,; ' " (!?) +-*/=
ptsans-italic	<i>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</i> <i>1234567890.,; ' " (!?) +-*/=</i>
ptsans-bold	<b>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</b> <b>1234567890.,; ' " (!?) +-*/=</b>
ptsans-bold-ita lic	<b><i>abcdefghijklmnopqrstuvwxyZ ABCDEFGHIJKLMNOPQRSTUVWXYZ</i></b> <b><i>1234567890.,; ' " (!?) +-*/=</i></b>

The `DefaultFont` parameter takes only the font family. The phone automatically applies the style suffix. You can use the following built-in font families:

- ptsans
- aleo
- banksia
- droidsans
- libre-caslon
- opensans
- quattrocento
- terminal-dosis



- When using the Tomáš skin, the font is fixed. A configured `DefaultFont` value doesn't take effect.
- When using a language other than English, make sure to use a font family that includes all the characters in the chosen language.

## Language Customization with Dictionary Files

Language customization enables you to change the language of the local interface by installing a dictionary file for each selectable language. Dictionary files are XML files stored in the internal folder `/dict/` and must be UTF-8 encoded. You must install custom dictionary files using the data package upload method.

By default, the phone comes with three built-in dictionary files:

- English-US
- English-UK
- Spanish

You can set select the language under **Preferences** on the UI.

Below is the full Spanish dictionary file that is shipped with the phone. The name of the language is specified in the `lang` attribute of the root element `<dictionary>` and is also the name that displays in the **Preferences** language selection.

```
<obiphone>
<!-- Each dictionary item has a key/value pair, a mode attribute and a namespace.
      When applying the dictionary, we match the key against the source phrase.
      If mode="i", the comparison is case-insensitive; otherwise the comparison
      is case-sensitive. The namespace ns must be matched also. -->

<dictionary lang="Español">
  <!-- App names -->
  <item key="Contacts" value="Contactos"/>
  <item key="Current Calls" value="Llamadas"/>
  <item key="Call History" value="Historial"/>
  <item key="Preferences" value="Preferencias"/>
  <item key="Settings" value="Ajustes"/>
  <item key="Product Info" value="Producto"/>
  <item key="Buddy List" value="Amistades"/>
  <item key="Net Services" value="Red Servicios"/>

  <!-- Line key function name -->
  <item key="AANS" value="Respuesta Automática" mode="i"/>
  <item key="acd" value="Distr. automático de llamadas" mode="i"/>
  <item key="BXFER" value="Transferencia Ciega" mode="i"/>
  <item key="BAC" value="Blq. Llamadas Anónimas" mode="i"/>
  <item key="BCI" value="Blq. ID de Llamadas" mode="i"/>
  <item key="BLF" value="Monitor de teléfono" mode="i"/>
  <item key="call" value="Llamada" mode="i"/>
  <item key="CFA" value="Desvío de Llamadas" mode="i"/>
  <item key="CWA" value="Llamada en Espera" mode="i"/>
  <item key="CONF" value="Unir a Conferencia" mode="i"/>
  <item key="disp-code" value="Código Disposición" mode="i"/>
  <item key="DND" value="No Molestar" mode="i"/>
  <item key="DNR" value="No Suena" mode="i"/>
  <item key="HOLD" value="En Espera" mode="i"/>
  <item key="hoteling" value="Hoteling" mode="i"/>
  <item key="acd" value="ACD Agent State" mode="i"/>
```

```
<item key="exe" value="Exec Call Filter" mode="i"/>
<item key="xass" value="Exec Assistant" mode="i"/>
<item key="sec" value="Security Class" mode="i"/>
<item key="LINE" value="Monitor Línea" mode="i"/>
<item key="lkgpg" value="Siguiente Tabla" mode="i"/>
<item key="MWI" value="Mensajes" mode="i"/>
<item key="pg1" value="Grupo de Páginas 1"/>
<item key="pg2" value="Grupo de Páginas 2"/>
<item key="SPD" value="Marcado Rápido" mode="i"/>
<item key="XFER" value="Transferencia" mode="i"/>
<item key="pres" value="Presencia" mode="i"/>
<item key="cpm" value="Monitor de Aparcamiento de Llamada" mode="i"/>

<!-- Product Info entries -->
<item key="Model" value="Modelo" mode="i"/>
<item key="OBi Number" value="Número OBi" mode="i"/>
<item key="MAC Address" value="Dirección MAC" mode="i"/>
<item key="WiFi MAC Address" value="Dirección MAC de WiFi" mode="i"/>
<item key="Serial Number" value="Número Serial" mode="i"/>
<item key="Software Version" value="Versión del Software" mode="i"/>
<item key="Hardware Version" value="Versión del Hardware" mode="i"/>
<item key="Customization Status" value="Estado de Personalización" mode="i"/>
<item key="Up Time" value="Tiempo de Actividad" mode="i"/>

<!-- User Preferences entries -->
<item key="Language" value="Lenguaje" mode="i"/>
<item key="Skin" value="Contraste" mode="i"/>
<item key="Background Picture" value="Imagen de Fondo" mode="i"/>
<item key="Default Ringtone" value="Timbre Predeterminado" mode="i"/>
<item key="Default Font" value="Fuente Predeterminada" mode="i"/>
<item key="Screen Saver" value="Protector de Pantalla" mode="i"/>
<item key="Screen Saver Delay" value="Protector de Pantalla de retardo" mode="i"/>
<item key="Screen Saver Type" value="Tipo de Protector de Pantalla" mode="i"/>
<item key="Screen Brightness" value="Brillo de la Pantalla" mode="i"/>
<item key="Preferred Audio Device" value="Dispositivo de Audio Preferido" mode="i"/>
<item key="Preferred Headset Device" value="Auriculares Preferido" mode="i"/>
<item key="Do Not Disturb" value="No Molestar" mode="i"/>
<item key="Do Not Ring" value="No Suena" mode="i"/>
<item key="Call Forward" value="Desvío de Llamadas" mode="i"/>
<item key="Call Waiting" value="Llamada en Espera" mode="i"/>
<item key="Block Anonymous Call" value="Bloq. Llamadas Anónimas" mode="i"/>
<item key="Anonymous Call" value="Llamadas Anónimas" mode="i"/>
<item key="Auto Answer Page" value="Página de Respuesta Automática" mode="i"/>
<item key="Join Page Group" value="Únete Grupo de Páginas" mode="i"/>
<item key="Ringer Volume" value="Volumen del Timbre" mode="i"/>
<item key="Speakerphone Volume" value="Volumen de Altavoz" mode="i"/>
<item key="Speakerphone Mic Gain" value="Ganancia Mic Altavoz" mode="i"/>
<item key="Handset Volume" value="Volumen de Auriculares" mode="i"/>
<item key="Handset Mic Gain" value="Ganancia Mic Auriculares" mode="i"/>
<item key="RJ9 Headset Volume" value="RJ9 Volumen de Auricular" mode="i"/>
```

```
<item key="RJ9 Headset Mic Gain" value="RJ9 Ganancia Mic Auricular" mode="i"/>
<item key="Equalizer" value="Equalizador" mode="i"/>
<item key="AEC" value="Cancelación de Eco Acústico" mode="i"/>
<item key="Home App Columns" value="Columnas de Aplicación Home" mode="i"/>
<!-- Screen Saver Type -->
<item key="Slide Show" value="Pantalla de Visualizacion" mode="i"/>
<item key="Turn Off Display" value="Desactivar la Visualización" mode="i"/>
<!-- Preferred Audio Device -->
<item key="Speaker" value="Altavoz" mode="i"/>
<item key="Handset" value="Auriculares" mode="i"/>
<item key="Headset" value="Auricular" mode="i"/>
<!-- Preferred Headset Device -->
<item key="RJ9 Headset" value="RJ9 Auricular" mode="i"/>

<!-- Call Forward -->
<item key="Destination Number" value="Número de Destino" mode="i"/>
<item key="Voice Service" value="Servicio de Voz" mode="i"/>

<!-- Phone Book -->
<item key="Phone Book" value="Agenda" mode="i"/>
<item key="Fields" value="Campos" mode="i"/>
<item key="Name" value="Nombre" mode="i"/>
<item key="First Name" value="Primer Nombre" mode="i"/>
<item key="Last Name" value="Apellidos" mode="i"/>
<item key="Number" value="Número" mode="i"/>
<item key="Mobile Number" value="Número Movil" mode="i"/>
<item key="Office Number" value="Número de Oficina" mode="i"/>
<item key="Home Number" value="Numero de Casa" mode="i"/>
<item key="Service" value="Servicio" mode="i"/>
<item key="Picture" value="Foto" mode="i"/>
<item key="Ringtone" value="Timbre" mode="i"/>
<item key="Email" value="Email" mode="i"/>
<item key="Company" value="Empresa" mode="i"/>
<item key="Group" value="Grupo" mode="i"/>
<item key="Groups" value="Groupos" mode="i"/>
<item key="Co-Workers" value="Compañeros de Trabajo" mode="i"/>
<item key="Customers" value="Clientes" mode="i"/>
<item key="Family" value="Familia" mode="i"/>
<item key="Friends" value="Amigos" mode="i"/>
<item key="No Groups Defined" value="Grupo No Definido" mode="i"/>
<item key="Speed Dial" value="Marcación Rápida" mode="i"/>
<item key="Phone Book Filter" value="Filtro de Agenda" mode="i"/>

<!-- Call History -->
<item key="Missed Calls" value="Llamadas Perdidas" mode="i"/>
<item key="Received Calls" value="Llamadas Recibidas" mode="i"/>
<item key="Outgoing Calls" value="Llamadas Realizadas" mode="i"/>
<item key="All Calls" value="Todas las Llamadas" mode="i"/>
<item key="List empty" value="Lista Vacía" mode="i"/>
```

```

<!-- Buddy List -->
<item key="My Status" value="Mi Estatus" mode="i"/>
<item key="My Presence" value="Mi Presencia" mode="i"/>
<!-- Presence status -->
<item key="unknown" value="Desconocido" mode="i"/>
<item key="offline" value="Fuera de Linea" mode="i"/>
<item key="online" value="En Linea" mode="i"/>
<item key="away" value="Fuera de Oficina" mode="i"/>
<item key="xa" value="Fuera de Oficina Extendida" mode="i"/>
<item key="dnd" value="No Molestar" mode="i"/>
<item key="Online (Available)" value="En Linea (Disponible)" mode="i"/>
<item key="Offline (Invisible)" value="Fuera de Linea (Invisible)" mode="i"/>
<item key="Extended Away" value="Fuera de Oficina Extendida" mode="i"/>
<item key="DND (Do Not Disturb)" value="DND (No Molestar)" mode="i"/>

<!-- Network Directory -->
<item key="Enterprise" value="Empresa" mode="i"/>
<item key="Group Common" value="Grupo en Comun" mode="i"/>
<item key="Enterprise Common" value="Empresa en Comun" mode="i"/>
<item key="Personal" value="Personal" mode="i"/>

<!-- Settings -->
<item key="Network" value="Reseteo de Fabrica" mode="i"/>
<item key="Product Information" value="Informacion Producto" mode="i"/>
<item key="WiFi" value="WiFi" mode="i"/>
<item key="Bluetooth" value="Bluetooth" mode="i"/>
<item key="Storage" value="Almacenamiento" mode="i"/>
<item key="User Preferences" value="Preferencias Usuario" mode="i"/>
<item key="Programmable Keys" value="Teclas Programables" mode="i"/>
<item key="Line Keys" value="Teclas de Linea" mode="i"/>
<item key="Side Car 1" value="Consola de Operadora 1" mode="i"/>
<item key="Side Car 2" value="Consola de Operadora 2" mode="i"/>
<item key="Voice Services" value="Servicios de Voz" mode="i"/>
<item key="Device Admin" value="Administracion Dispositivo" mode="i"/>
<item key="Lock Admin Settings" value="Bloqueo Config. del Admin." mode="i"/>
<item key="Unlock Admin Settings" value="Desbloquear Ajustes del Admin." mode="i"/>
<!-- Network -->
<item key="Network Settings" value="Ajustes de la Red" mode="i"/>
<item key="Addressing Type" value="Addressing Type" mode="i"/>
<item key="IP Address" value="Direccion IP" mode="i"/>
<item key="Subnet Mask" value="Mascara Subred" mode="i"/>
<item key="Default Gateway" value="Direccion Gateway" mode="i"/>
<item key="DNS Server" value="Servidor DNS" mode="i"/>
<item key="DNS Query Order" value="Orden de Consulta DNS" mode="i"/>
<item key="DNS Query Delay" value="Retrazo de Consulta DNS" mode="i"/>
<item key="AC Name" value="Nombre AC" mode="i"/>
<item key="Service Name" value="Nombre de Servicio" mode="i"/>
<item key="Password" value="Contraseña" mode="i"/>
<item key="VLAN" value="VLAN" mode="i"/>
<item key="LLDP-MED" value="LLDP-MED" mode="i"/>

```

```

<item key="ID" value="ID" mode="i"/>
<item key="Priority" value="Prioridad" mode="i"/>
<item key="NTP Server" value="Servidor NTP" mode="i"/>
<item key="Local Time Zone" value="Zona Horaria Local" mode="i"/>
<item key="Daylight Saving Time" value="Horario de Verano" mode="i"/>
<!-- WiFi -->
<item key="WiFi Setup" value="Configuracion WiFi" mode="i"/>
<item key="OBiWiFi Setup Mode" value="Modo de configuración OBiWiFi" mode="i"/>
<item key="Security" value="Seguridad" mode="i"/>
<item key="Signal Strength" value="Intensidad de Señal" mode="i"/>
<item key="MAC Address" value="Direccion MAC" mode="i"/>
<item key="0" value="" ns="wfs"/>
<item key="1" value="Débil" ns="wfs"/>
<item key="2" value="Pobre" ns="wfs"/>
<item key="3" value="Pasable" ns="wfs"/>
<item key="4" value="Bueno" ns="wfs"/>
<item key="5" value="Excelente" ns="wfs"/>
<!-- Storage -->
<item key="Device" value="Dispositivo" mode="i"/>
<item key="File System Type" value="Tipo de Sistema de Archivo" mode="i"/>
<item key="Capacity" value="Capacidad" mode="i"/>
<item key="Internal" value="Interno" mode="i"/>
<item key="USB storage 1" value="USB Flash Drive 1" mode="i"/>
<item key="USB storage 2" value="USB Flash Drive 2" mode="i"/>
<!-- Voice Services -->
<!--
    Highly technical entries under Voice Services are not translated.
-->
<!-- Admin Settings -->
<item key="Web Server Port" value="Puerto del Servidor Web" mode="i"/>
<item key="Web Admin Password" value="Contraseña del Administrador" mode="i"/>
<item key="Web User Password" value="Contraseña del Usuario" mode="i"/>
<item key="Syslog Server" value="Servidor Syslog" mode="i"/>
<item key="ITSP Provisioning Method" value="Método Aprovisionamiento ITSP" mode="i"/>
<item key="ITSP Provisioning Interval" value="Aprovisionamiento Interno ITSP"
mode="i"/>
<item key="ITSP Provisioning Config URL" value="Aprovisionamiento ITSP Config URL"
mode="i"/>
<item key="Auto Firmware Update Method" value="Metodos de Actualizacion Auto Firmware"
mode="i"/>
<item key="Auto Firmware Update Interval" value="Actualizacion Interno Auto Firmware"
mode="i"/>
<item key="Auto Firmware Update URL" value="Actualizacion Auto Firmware URL"
mode="i"/>
<!-- Provisioning and FW Upgrade Method -->
<item key="System Start" value="Inicio del Sistema" mode="i"/>
<item key="Periodically" value="Periodicamente" mode="i"/>
<!-- Dial App -->
<item key="Enter Number" value="Número de Entrada" mode="i"/>
<item key="Transfer Target" value="Destino de la Transferencia" mode="i"/>

```

```
<item key="Conference Target" value="Objetivo Conferencia" mode="i"/>

<!-- Soft Keys -->
<item key="Edit" value="Editar" mode="i"/>
<item key="Reboot" value="Reiniciar" mode="i"/>
<item key="End" value="Final" mode="i"/>
<item key="Hold" value="Espera" mode="i"/>
<item key="Resume" value="Resumen" mode="i"/>
<item key="Add to Conf" value="Unir a Conferencia" mode="i"/>
<item key="Conference" value="Conferencia" mode="i"/>
<item key="Park" value="Aparcar" mode="i"/>
<item key="Blind Transfer" value="Transferencia Ciega" mode="i"/>
<item key="Dispose Code" value="Deseche Código" mode="i"/>
<item key="Escalate" value="Escalar" mode="i"/>
<item key="Trace" value="Trazar" mode="i"/>
<item key="Rec.Start" value="Iniciar Grabacion" mode="i"/>
<item key="Rec.Stop" value="Parar Grabacion" mode="i"/>
<item key="Rec.Pause" value="Pausar Grabacion" mode="i"/>
<item key="Rec.Resume" value="Resumir Grabacion" mode="i"/>
<item key="Private Hold" value="Espera Privada" mode="i"/>
<item key="Answer" value="Contestar" mode="i"/>
<item key="Reject" value="Rechazar" mode="i"/>
<item key="Redial" value="Rellamar" mode="i"/>
<item key="Missed" value="Perdida" mode="i"/>
<item key="Dial" value="Marcar" mode="i"/>
<item key="Lines" value="Lineas" mode="i"/>
<item key="Switch Mode" value="Cambio Modo" mode="i"/>
<item key="Ph.Book" value="Agenda" mode="i"/>
<item key="Switch Line" value="Cambio Linea" mode="i"/>
<item key="Refresh All" value="Reiniciar Todo" mode="i"/>
<item key="+Buddy" value="Agregar Amigo" mode="i"/>
<item key="Search" value="Busqueda" mode="i"/>
<item key="Refresh" value="Reiniciar" mode="i"/>
<item key="+" value="Agregar" mode="i"/>
<item key="Add" value="Agregar" mode="i"/>
<item key="Remove" value="Remover" mode="i"/>
<item key="Clear List" value="Limpiar Lista" mode="i"/>
<item key="Edit Dial" value="Aditar Marcador" mode="i"/>
<item key="Save" value="Guardar" mode="i"/>
<item key="MyPresence" value="Mi Presencia" mode="i"/>
<item key="MyStatus" value="Mi Estatus" mode="i"/>
<item key="Filter" value="Filtrar" mode="i"/>
<item key="Select" value="Seleccionar" mode="i"/>
<item key="New" value="Nuevo" mode="i"/>
<item key="Clear" value="Limpiar" mode="i"/>
<item key="Sort" value="Ordenar" mode="i"/>
<item key="New Entry" value="Nueva Entrada" mode="i"/>
<item key="Group Filter" value="Filtrar Grupo" mode="i"/>
<item key="Import" value="Importar" mode="i"/>
<item key="Export" value="Exportar" mode="i"/>
```



```
<item key="Sync" value="Sincronizar" mode="i"/>
<item key="New Group" value="Nuevo Grupo" mode="i"/>
<item key="Uncheck All" value="No chequear Nada" mode="i"/>
<item key="Check All" value="Chequear Todo" mode="i"/>
<item key="Factory Reset" value="Reseteo de Fabrica" mode="i"/>
<item key="FW Update" value=" Actualizar FW" mode="i"/>
<item key="Pick Up" value="Contestar" mode="i"/>
<item key="Barge In" value="Interrumpir" mode="i"/>
<item key="Monitor" value="Monitorear" mode="i"/>
<item key="Sign Off" value="Cerrar Sesión" mode="i"/>
<item key="Wrap Up" value="Envolver" mode="i"/>
<item key="Scan" value="Escanear" mode="i"/>
<item key="Transfer Now" value="Transferir Ahora" mode="i"/>
<item key="Conference Now" value="Conferencia Ahora" mode="i"/>
<item key="Play" value="Seguir" mode="i"/>
<item key="Stop" value="Parar" mode="i"/>
<item key="Connect" value="Conectar" mode="i"/>
<item key="Rescan" value="Re-escanear" mode="i"/>
<item key="Disconnect" value="Desconectar" mode="i"/>
<item key="Forget" value="Olvidar" mode="i"/>
<item key="Reconnect" value="Reconectar" mode="i"/>
<item key="Explore" value="Explorar" mode="i"/>
<item key="Eject" value="Ejecutar" mode="i"/>
<item key="Mount" value="Montura" mode="i"/>
<item key="Back" value="Regresar" mode="i"/>
<item key="Copy" value="Copiar" mode="i"/>
<item key="Cancel" value="Cancelar" mode="i"/>
<item key="Paste" value="Pasar" mode="i"/>
<item key="New Folder" value="Nuevo Folder" mode="i"/>
<item key="Close" value="Cerrar" mode="i"/>
<item key="OK" value="OK" mode="i"/>
<item key="Replace" value="Reemplazar" mode="i"/>
<item key="Add As New" value="Aragrar Como Nuevo" mode="i"/>

<item key="&gt;" value="&gt;"/>
<item key="&lt;" value="&lt;"/>
<item key="Backspace" value="Retroceso" mode="i"/>
<item key="Devices" value="Dispositivos" mode="i"/>

<!-- Misc -->
<item key="Not Found" value="Extraviado" mode="i"/>
<item key="No Entries" value="No Entradas" mode="i"/>
<item key="Unassigned" value="No Asignado" mode="i"/>
<item key="Enable" value="Activar" mode="i"/>
<item key="enabled" value="Activado" mode="i"/>
<item key="disabled" value="Desactivado" mode="i"/>
<item key="Not Configured" value="No Configurado" mode="i"/>
<item key="default" value="Por Defecto" mode="i"/>
<item key="On" value="Encendido" mode="i"/>
<item key="Off" value="Apagado" mode="i"/>
```

```

<item key="Non-linear" value="No Linear" mode="i"/>
<item key="Status" value="Estatus" mode="i"/>
<item key="outgoing" value="A" ns="ct"/>
<item key="received" value="De" ns="ct"/>
<item key="up" value="Arriba" mode="i"/>
<item key="down" value="Abajo" mode="i"/>
<item key="nocfg" value="No Configurado" mode="i"/>
<item key="holding" value="Esperando" mode="i" ns="cs"/>
<item key="peerring" value="Timbrando" mode="i" ns="cs"/>
<item key="proceeding" value="Timbrando" mode="i" ns="cs"/>
<item key="peerring2" value="Timbrando (Espera)" mode="i" ns="cs"/>
<item key="peerring3" value="Timbrando (Conf)" mode="i" ns="cs"/>
<item key="dialtone" value="Tono de Marcación" mode="i" ns="cs"/>
<item key="dialing" value="Marcando" mode="i" ns="cs"/>
<item key="trying" value="Tratando" mode="i" ns="cs"/>
<item key="connected" value="Conectado" mode="i" ns="cs"/>
<item key="connected-HD" value="HD Conectado" mode="i" ns="cs"/>
<item key="ended" value="Llamada Finalizada" mode="i" ns="cs"/>
<item key="ring" value="Timbrando" mode="i" ns="cs"/>
<item key="OBiTALK" value="OBiTALK" mode="i"/>
<item key="Auto" value="" mode="i"/>
<item key="Auto: " value="" mode="i"/>
<item key="OBiTALK: " value="OBiTALK: " mode="i"/>
<item key="Not Available" value="No Disponible" mode="i"/>
<item key="AA1: " value="OBi Asistente: " mode="i"/>
<item key="AA2: " value="IVR: " mode="i"/>
<item key="seized" value="Incautado" ns="cs"/>
<item key="held_private" value="Held Privado" ns="cs"/>
<item key="held" value="Held" ns="cs"/>
<item key="parked" value="Llamar Estacionado" ns="cs"/>

<!-- Misc line key data -->
<item key="error" value="error" mode="i"/>
<item key="Service Not Ready" value="Servicio no está listo" mode="i"/>
<item key="Service Subs. Error" value="Error suscripción de servicio" mode="i"/>
<item key="BLF Subs. Error" value="BLF error suscripción" mode="i"/>
<item key="Hoteling Subs. Error" value="Hoteling error suscripción" mode="i"/>
<item key="Guest" value="Invitado" mode="i"/>
<item key="No Guest" value="Los huéspedes no" mode="i"/>

<!-- ACD State -->
<item key="available" value="Disponible" mode="i"/>
<item key="unavailable" value="No Disponible" mode="i"/>
<item key="wrappingup" value="Terminando" mode="i"/>
<item key="signedoff" value="Cerrar Sesión" mode="i"/>

<!-- Input App Titles -->
<item key="New Status" value="Nuevo Estatus" mode="i"/>
<item key="Edit Status" value="Editar Estatus" mode="i"/>
<item key="New Group Name" value="Nuevo Nombre de Grupo" mode="i"/>

```

```

<item key="Enter" value="Entrar" mode="i"/>
<item key="Blind Transfer Target" value="Objetivo Transferencia ciega" mode="i"/>
<item key="Supervisor Extension" value="Supervisor de Extensión" mode="i"/>
<item key="Disposition Code" value="Disposicion de Codigo" mode="i"/>
<item key="Forward All Number" value="Remitir Todo Número" mode="i"/>
<item key="Unavailable Reason Code" value="No disponible Código de Motivo" mode="i"/>
<item key="Enter Query" value="Entrar Consulta" mode="i"/>
<item key="Hoteling Passcode" value="Codigo de Contraseña Hoteling" mode="i"/>
<item key="Hoteling Guest Extension" value="Extension Huesped Hoteling" mode="i"/>
<item key="Enter Admin Password" value="Ingrese Contraseña del Administrador"
mode="i"/>
<item key="Enter Folder Name" value="Ingrese Nombre del Archivo" mode="i"/>
<item key="Exec Assist Divert Number" value="Exec Asist Numero de Desvio" mode="i"/>
<!-- Call related alerts -->
<item key="No Active Calls" value="No hay Llamadas Activas" mode="i"/>
<item key="Call Error" value="Error de Llamada" mode="i"/>
<item key="No Call Key available to make another call" value="No Tecla de llamada
Disponible para hacer otra llamada" mode="i"/>
<item key="The requested service is not available at the moment" value="En este
momento el Servicio requerido no esta disponible" mode="i"/>
<item key="Conference Error" value="Error de Conferencia" mode="i"/>
<item key="No conference port available to add participant" value="No hay puerto de
conferencia para agregar participante" mode="i"/>
<!-- OBiWiFi -->
<item key="OBiWiFi" value="OBiWiFi" mode="i"/>
<item key="Use USB Port 1 for OBiWiFi." value="Use el puerto USB 1 para OBiWiFi."
mode="i"/>
<item key="Connected with: " value="Conectado con: " mode="i"/>
<item key="Connecting with: " value="Conectar con: " mode="i"/>
<item key="Failed to authenticate with " value="Fallado para autenticar con "
mode="i"/>
<item key="Authentication Failed" value="Falla la Autenticación" mode="i"/>
<item key="Failed to associate with " value="Error al asociar con " mode="i"/>
<item key="Association Failed" value="Asociación no" mode="i"/>
<!-- Reboot -->
<item key="Reboot" value="Reiniciar" mode="i"/>
<item key="Press OK to reboot the phone." value="Presione OK para Reiniciar el
Telefono." mode="i"/>
<!-- Factory reset -->
<item key="Press OK to reset all settings to factory default. Phone will reboot
automatically."
value="Presione OK para Resetear Todos los Ajustes de Fabrica por defecto. El
Telefono se Reiniciara Automaticamente." mode="i"/>
<!-- Admin login -->
<item key="Unlock Settings Failed" value="Desbloquear Configuración Error" mode="i"/>
<item key="Invalid Password" value="Contraseña Invalida" mode="i"/>
<!-- OBiTalk -->
<item key="Add Device to OBiTalk" value="Agregue un dispositivo a OBiTalk" mode="i"/>
<item key="Code sent to OBiTalk..." value="Codigo enviado a OBiTalk..." mode="i"/>

```

```

    <item key="Device not allowed to join!" value="Dispositivo no se le permitió unirse!"
mode="i"/>
    <!-- Conference bridge -->
    <item key="Conf. Bridge Error" value="Error del Puente de Conferencia" mode="i"/>
    <item key="Call to Conference Bridge failed" value="Llamada al Puente Conferencia
fallado" mode="i"/>
    <item key="Conference Bridge ended unexpectedly" value="Puente Conferencia terminó
inesperadamente" mode="i"/>
    <item key="Last Call Total Charges" value="Gasto Total de la ultima llamada"
mode="i"/>
    <!-- Network directory -->
    <item key="Network Directory" value="Directorio Red" mode="i"/>
    <item key="refreshed" value="Actualizado" mode="i"/>
    <item key="Nothing refreshed" value="Nada Actualizado" mode="i"/>
    <item key="Network Directory Search" value="Busqueda de Directorio Red" mode="i"/>
    <item key="No items found for your query" value="No se han encontrado en su búsqueda
artículos" mode="i"/>
    <!-- Buddy list -->
    <item key="Buddy List Error" value="Error de Lista de Amigos" mode="i"/>
    <item key="Failed to load buddy list" value="Fracaso en Bajar Lista de Amigos"
mode="i"/>
    <!-- Call history -->
    <item key="Clear List" value="Limpiar Lista" mode="i"/>
    <item key="Press OK to remove all entries in the selected call history"
value="Presione OK para remover todas las entradas en el historial de llamadas
seleccionadas" mode="i"/>
    <item key="Duplicate Entry" value="Entrada Duplicada" mode="i"/>
    <item key="Select 'Replace' or 'Add As New' to proceed." value="Seleccione
'Reemplazar' o 'Agregar como Nuevo' para proceder." mode="i"/>
    <!-- Phone book -->
    <item key="Remove Entry" value="Remover Entradas" mode="i"/>
    <item key="Press OK to permanently remove this entry from the Phone Book."
value="Presione OK para remover permanentemente esta entrada de la Agenda."
mode="i"/>
    <item key="Discard Changes?" value="Descartar Cambios?" mode="i"/>
    <item key="Press OK to proceed. Changes to the Phone Book will be discarded."
value="Presione OK Proceder. Cambios a la Agenda sera descartado." mode="i"/>
<!-- Misc -->
    <item key="Progress" value="Progreso" mode="i"/>
    <item key="More" value="Mas" mode="i"/>
    <item key="Disconnected" value="Desconectado" mode="i"/>
    <item key="Secured with" value="Asegurado con" mode="i"/>
    <item key="Not in range" value="Fuera de rango" mode="i"/>
    <item key="Peer to peer" value="De igual a igual" mode="i"/>
    <item key="Ad-hoc" value="Ad-hoc" mode="i"/>
    <item key="Remembered" value="Recordado" mode="i"/>
    <item key="Not Discoverable" value="No detectable" mode="i"/>
    <item key="Available" value="Disponible" mode="i"/>
    <item key="Seconds remaining" value="Segundos Restantes" mode="i"/>
</dictionary>

```

```
</obiphone>
```

On boot up, your phone scans the dictionary directory for all installed dictionary files and prepares a list of language names by extracting the lang attribute in each file. The list of languages is presented to you under **Preferences > Phone > Language** in the phone UI. On the phone configuration web page the value for the `Language` parameter (under **User Settings – User Preferences**) is just a string and must be set to one of the available language names.

## Phone Book Pictures

You can associate pictures with each phone book entry. The phone stores these pictures in the internal `/people/` folder. You can use JPEG, PNG, GIF, and BMP file formats. The recommended pixel dimension is no larger than 100 × 100 pixels.



The pictures display on the phone screen as part of caller-ID when the caller's number matches an entry in the phone book.

## Ring Tones

Ring tones are included in the phone customization data package. The phone stores ring tones in the internal folder `/ringtones/`. Ring tone filenames must end with the `.wav` extension.

## Phone Customization Data Package

You must consolidate customizable data into one single data package to upload it onto the phone. The final uploadable data package is a `.tar` file that you can optionally zip. The size of the tar file must be no larger than 30 MB. Inside the tar file, organize the data with the same folder structure as the phone's internal storage:

```
/backgnd Wall paper (background pictures) and logo.raw (splash screen)
/dict Dictionary files
/fonts Text font files
/ringtones Ring tone files
/people Phone book contact pictures
```

For example, create a directory `/home/test/obiphone-data` on a Linux machine that contains an appropriate folder structure. Copy all the data files you want to upload to the phone in the corresponding folders in this directory and issue the following command lines to create the target tar file:

```
% cd /home/test/obiphone-data
% tar -zcvf ../obiphone-data.tar.gz *
```



You must issue the tar command inside of the working directory `obiphone-data`, as the phone can't strip the first level in the `.tar` file.

Additionally, you can use the following command line to generate the MD5 checksum of the target `.tar` file after you create it:

```
% md5sum obiphone-data.tar.gz
```

The MD5 checksum is used as the version of the target customization data package. You must configure it on the phones to trigger the phones to download a different version of the package if the currently installed package has a different MD5 checksum from the configured MD5 checksum.

## Uploading Customization Data Package to the Phone

After the tar file of the customization data package is created, it must be placed on a server to be uploaded to the phones when the file is requested from the server by the phones. On the phone side, the downloading of the customization data package from a server is controlled by a number of configuration parameters as described in the table below, not unlike the way phone firmware is updated. As you can tell quickly that these parameters have the same meaning as those similarly named parameters for downloading of firmware.

Note that your phone attempts to download a customization data package from the server after firmware update and configuration parameter provisioning are completed. If the MD5 checksum of the downloaded data packages does not match the configured value, the package is dropped and no data update is performed. Otherwise, the phone install the downloaded data files into the corresponding folders and then restarts (warm reboot).

### Customization Data Package Parameters

Parameter Group	Parameter	Description
Provisioning	Method	<p>This parameter controls if and when the phone should execute given <code>DownloadURL</code> to download the customization data package (if the installed version is not current). Choice of:</p> <ul style="list-style-type: none"> <li>Disabled = Do not execute <code>DownloadURL</code></li> <li>System Start = Execute <code>DownloadURL</code> just once on system start</li> <li>Periodically = Execute <code>DownloadURL</code> on system start, and then periodically at the interval specified in the <code>Interval</code> parameter</li> </ul> <p>Note: First download on system start occurs after firmware update and configuration provisioning are complete.</p>
	Interval	<p>When <code>Method</code> is set to <code>Periodically</code>, this is the interval in number of seconds between execution of <code>DownloadURL</code>. If value is 0, the phone executes <code>DownloadURL</code> just once on system start (that is, equivalent to setting <code>Method</code> to <code>System Start</code>).</p>
	DownloadURL	<p>In its simplest form, this is the URL to download the Phone Customization Data Package, such as</p> <pre>http://abcd.com/phone/cfg/datapkg.tar</pre> <p>The full syntax is a script similar to what can be specified in the <code>FirmwareURL</code> parameter for firmware update. Refer to the <i>Polycom OBi Edition Deployment Guide</i> for a description of all the syntaxes of this script.</p>

**Customization Data Package Parameters (continued)**

Parameter Group	Parameter	Description
	<b>MD5Checksum</b>	Standard MD5 checksum of the Customization Data Package (a 32-character string of hexadecimal digits). This value must be provided to the phone and it serves as the version of the target data package. The phone executes <code>DownloadURL</code> only if the currently installed data package has a different MD5 checksum from the value of this parameter.
	<b>Incremental</b>	This Boolean option, if enabled, causes the phone to install the customization data files without erasing the old data files first. Otherwise, the old data files are erased before saving the downloaded data files.
	<b>DnsLookupType</b>	Choice of: <ul style="list-style-type: none"> <li>• A Record Only</li> <li>• SRV Record Only</li> <li>• Try Both</li> </ul>
	<b>DnsSrvPrefix</b>	Choice of: <ul style="list-style-type: none"> <li>• No Prefix</li> <li>• With Prefix</li> <li>• Try Both</li> </ul>
	<b>Username</b>	(Optional) Username for authentication if <code>DownloadURL</code> uses the scheme <code>http://</code> or <code>https://</code>
	<b>Password</b>	(Optional) Password for authentication if <code>DownloadURL</code> uses the scheme <code>http://</code> or <code>https://</code>

## Internal Data Storage Paths for User Preferences Settings

There are a number of configuration parameters to directly or indirectly select one of several internally stored data files for the phone to perform certain tasks, such as background picture or ring tone. This section describes how the built-in and customized data files that support these settings are organized internally so you can provision these parameter values accordingly.

The data files are organized into three levels: OBi Built-in, ITSP (or Administrator) Customized, and User Customized. Each level has its own dedicated internal storage areas. The following table summarizes the user settings that use these data files and where the phone stores the data files for each level:

## Internal Data Storage Paths for User Preferences

Configuration Parameter	Description	Internal Data Storage Folders		
		Phone Built-in	ITSP Customized	User Customized
<b>User Preferences:: Language</b>	<p>The parameter value must match the lang attribute of the root element of one of the dictionary xml files found under the folders on the right. When there is a conflict, the user version has the highest priority, then the ITSP version, then your phone's version.</p> <p>On each bootup the folders are scanned to create a list of available languages from the lang attributes that users can select from the Preferences menu as their Language.</p>	/data/dict/ /	/scratch/it sp/dict/	/scratch/ph one/dict/
<b>User Preferences:: BackgroundPicture</b>	<p>The parameter value must be the full internal path name of a picture file stored in one of the folders on the right or a URL.</p> <p>If a URL is specified in the value, it must start with <code>http://</code> or <code>https://</code> and the phone downloads and caches the data internally until it is power cycled.</p> <p>All available pictures under these folders are also listed under the Preferences menu for users to select as their wall paper picture. The filenames must have one of the following extensions: PNG, JPG, JPEG, GIF, or BMP.</p>	/data/themes/ backgnd/ /	/scratch/it sp/backgnd/ /	/scratch/ph one/backgnd/ /
<b>User Preferences:: DefaultRingtone</b>	<p>The parameter value must be the full internal path name of a wave file stored in one of the folders on the right or a URL.</p> <p>If a URL is specified in the value, it must start with <code>http://</code> or <code>https://</code> and the phone downloads and caches the data internally until it is power cycled.</p> <p>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.</p>	/data/ring tones/	/scratch/it sp/ringtone s/	/scratch/ph one/rington es/



## Internal Data Storage Paths for User Preferences (continued)

Configuration Parameter	Description	Internal Data Storage Folders		
		Phone Built-in	ITSP Customized	User Customized
There is no configuration parameter that refers to data files in these folders.	All available pictures under these folders on the right are listed under the Picture menu of the built-in Phone Book application for users to select to assign to a contact. The data filenames must on of the following extensions: PNG, JPG, JPEG, GIF, or BMP.	/obi/people/	/scratch/it sp/people/	/scratch/ph one/people/
<b>User Preferences::DefaultFont</b>	The parameter value must be one of the available font family names found in the font.map files in the folders on the right. When there is a conflict, the user version has the highest priority, then the ITSP customized version, then your phone's version.	/data/fonts/	/scratch/it sp/fonts/	/scratch/ph one/fonts/

If a parameter value is to specify the filename of an internally stored data file, the full internal path must be specified. For example:

**User Settings – User Preferences::BackgroundPicture** = /scratch/phone/backgnd/CherryBlossom.png

**User Settings – User Preferences::DefaultRingtone** = /data/ringtones/Office A.wav

# Updating the Software

---

You can use different methods to upgrade the firmware and software on your VVX business IP phone, OBi Edition.

## Updating the Firmware

You may need to update the firmware on your phone to receive new features, enhancements, and bug fixes.

### *Update Firmware Using IVR*

The easiest way to update the phone's firmware is by dialing \*\*\* then pressing option 6. If a firmware update is available, you're prompted to press 1 to continue and then hang up.

After hanging up, the download and update process completes in a couple of minutes over a fast internet connection. The update can take much longer over legacy connectivity such as DSL or T1.

The headset button starts to flash once the firmware download completes and the phone prepares to reboot.

### *Update Firmware from the Local Interface*

Launch the **Settings** App from the Main menu and press the **F/W Update** softkey. A message pops up on the screen to describe whether new firmware is available. If available, press the **Update Now** softkey. Do not power off the phone until the update completes and the phone automatically reboots immediately afterward.

### *Update Firmware from the System Web Interface*

Download the firmware file from OBITALK.COM and store it on your computer. Then visit the phone's native web page from a web browser on the same computer and click the **Firmware Update** link on the left side panel of the landing page. On the firmware update page, enter the path to the firmware file (or click **Browse** to locate the firmware file on the computer).

### *Automated Firmware Update*

You can configure rules to automatically check and download new phone firmware from a server.

## Background Firmware Update

You can update the firmware in the background on VVX 250, VVX 350, and VVX 450 business IP phones, OBi Edition. Once enabled, the phone handles the firmware update without going through the Polyupdater, and the update process occurs in the background during normal operation. Users can still use the phone normally.

After the new firmware copies to the flash memory (and the phone isn't in a call), the phone auto-reboots to use the new firmware version.

You also can install new firmware directly from the system web interface by selecting the firmware file stored on a local disk drive.



If you purchase a phone with software version 6.3.1 or later, background firmware update is enabled by default.

### Two Firmware File Formats: *sip.ld* and *.fw*

The current VVX firmware file format is the `sip.ld` format, which is used by both UCS and OBi Edition. To allow firmware update in the background, the new firmware to update to must use a different firmware format which is referred to as the `.fw` format. The `.fw` format is supported only when the background firmware update feature has been enabled. When this update feature has been enabled, the `sip.ld` format is no longer supported.

Like the `sip.ld` format firmware, Poly generates the `.fw` format for each build, signed with a Poly Release Key. For example, the following firmware file may be generated:

- `Skyline-6-3-1-25001-release-2018-12-13.fw` (signed with Release Key)

But unlike the `sip.ld` format firmware, which requires a separate build for each phone model, the same `.fw` format firmware file can be applied to all three phone models: VVX250, VVX350, and VVX450. The filename of a `.fw` image can be anything and must be explicitly specified in provisioning rules.

### Enabling Background Firmware Update

Enabling background firmware update requires a one-time format transition process. This process takes about 4 to 5 minutes to complete and the phone is not usable during this period. It is highly recommended to not to disconnect the power during the process. You can refer to the `BackgroundUpdate` parameter status on the Status/System Status web page (Product Information section) to check whether background firmware update feature has been enabled on the phone.

You can start this transition process by appending the `,bgu-conv=enable` keyword to the `FirmwareURL` parameter, such as:

```
http://192.168.1.1/backgroundupdate/,bgu-conv=enable
```

As soon as the device verifies that the firmware in the `http://192.168.1.1/backgroundupdate/` path is current, it starts the process automatically with a notification message displayed on the phone screen until the process is done.

Once the transition process is completed, the phone auto reboots and factory resets. All existing configuration, call logs, and phone book are lost. The `BackgroundUpdate` parameter is also changed to `Enabled`. In case the procedure is interrupted or failed, you will have to use the Polyupdater to recover the device. It is sometimes necessary to reinstall a `sip.ld` formatted image into the device.

## Using the Background Firmware Update Feature

After the feature is enabled, the device only takes a `.fw` formatted image. Also, when using auto firmware update during provisioning, a complete URL including the firmware file name must be given 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 filename results in a 404 or File not found error. The following is an example of a correct URL:

```
http://192.168.1.1/Skyline-6-3-1-25001-devel-2018-12-13.fw
```

When installing a `.fw` firmware from the device web page, the phone will NOT take a Config XML file as firmware file any more, as is the case of a `sip.ld` file. You must provide the path to the firmware file directly. If the given file is not a valid `.fw` firmware file, the firmware update procedure fails.

## Reverting Back to sip.ld

Reverting back to use `sip.ld` firmware is equivalent to disabling background firmware update (to a `.fw` firmware). This requires a one-time format (inverse) transition process on the device that has background firmware update enabled. To do this, specify a complete URL including the firmware filename of a corresponding `sip.ld` formatted image corresponding to the phone model in use in the `FirmwareURL` parameter with the special keyword `bgu-conv=disable` appended. For example:

```
http://192.168.1.1/ucs/3111-48840-001.sip.ld,bgu-conv=disable
```

When the phone finds the special keyword in `FirmwareURL`, it first downloads the firmware file header according to the given URL and makes sure it is a valid `sip.ld` firmware image. Then it saves the firmware update URL into the Polyupdater and marks a flag to let the Polyupdater restore the flash partition on the next boot, and then it auto-reboots. On the next boot up, the Polyupdater proceeds with reverting to the original (UCS style) flash partitioning and subsequently upgrades to the given `sip.ld` firmware. Due to flash repartitioning, the phone configuration is factory reset and all existing configuration, call logs, phone book, and so forth are lost as a result.

# Customization Data Auto Update

You can automatically update the phone customization data at the ITSP level.

For more information, see [Phone Customization Data](#).

## Auto Update Operation

Similar to the Auto Firmware Update feature, the Customization Data Auto Update feature is configured by a set of parameters. When it is enabled, your phone attempts to download the Customization Data package according to the scripts specified in the `DownloadURL` parameter. Then the data in the package is validated by the given MD5 checksum before it is installed to internal storage. As soon as data installation successfully completes, your phone restarts itself (warm boot) and won't try to download the package again until the MD5 checksum is changed, even if the feature is still enabled.

The Auto Update operation is always performed after firmware update and all configuration provisioning are completed.

## Auto Update Configuration Parameters

This feature is configured by the following parameters.

### Auto Update Configuration Parameters

Parameter	Description	Default Setting
<b>Method</b>	<p>Current operational method of Provisioning. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>Disabled</code> = Do not download from <b>DownloadURL</b></li> <li>• <code>System Start</code> = Download from <b>DownloadURL</b> just once on system start</li> <li>• <code>Periodically</code> = Download from <b>DownloadURL</b> on system start, and then periodically at the interval specified in the <b>Interval</b> parameter</li> </ul> <p>Note: First download on system start is performed after firmware update and configuration provisioning are complete.</p>	Disabled
<b>Interval</b>	<p>When <b>Method</b> is set to <code>Periodically</code>, this is the number of seconds between download from <b>DownloadURL</b>. If value is 0, device downloads once only on system start (equivalent to setting <b>Method</b> to <code>System Start</code>).</p>	0
<b>DownloadURL</b>	URL of Customization Data package.	
<b>MD5Checksum</b>	Standard MD5 checksum (hexadecimal string) of the Customization Data package.	
<b>Incremental</b>	When enabled, the Customization Data package is installed incrementally without erasing the old data.	false
<b>DnsLookupType</b>	<p>Choice of:</p> <ul style="list-style-type: none"> <li>• <code>A Record Only</code></li> <li>• <code>SRV Record Only</code></li> <li>• <code>Try Both</code></li> </ul>	A Record Only
<b>DnsSrvPrefix</b>	<p>Choice of:</p> <ul style="list-style-type: none"> <li>• <code>No Prefix</code></li> <li>• <code>With Prefix</code></li> <li>• <code>Try Both</code></li> </ul>	No Prefix
<b>Username</b>	Optional <b>Username</b> for authentication if URL scheme is <code>http://</code> or <code>https://</code> .	
<b>Password</b>	Optional <b>Password</b> for authentication if URL scheme is <code>http://</code> or <code>https://</code> .	

# Hardware and Accessories

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This section provides information on configuring power management options and available VVX business IP phone, OBi Edition, accessories for your users.

## Powering VVX business IP phones, OBi Edition, with an Ethernet Switch Connection

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

You can power your phone through an AC power adapter or through a Power over Ethernet (PoE) cable connected to the phone's **SW** Ethernet switch port.



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

## Available Accessories

VVX business IP phones, OBi Edition, support the following accessories:

- 5 VDC power adapter: Powers the phone when Power over Ethernet (PoE) isn't available.
- Polycom Wi-Fi wireless network adapter: Connects the phone to a wireless network (VVX 250, VVX 350, and VVX 450 only).
- USB flash drive: Use to copy ring tones, background images, and other media to the phone (VVX 250, VVX 350, and VVX 450 only).
- USB headsets (VVX 250, VVX 350, and VVX 450 only).

# Poly VVX EM50 Expansion Modules

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

The following features are available on the VVX EM50 expansion module:

- Assign a function to each line key
- Thirty line keys and three display pages
- Supports 90 lines on each expansion module and 48 phone lines
- Connect up to two expansion modules to a VVX 450 business IP phone to support a total of 228 lines
- The expansion module has a USB port to connect low-powered devices



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

## Set Up a VVX EM50 Expansion Module

VVX EM50 expansion modules are powered and signaled by VVX business IP phones, OBi Edition, and require minimal setup. Expansion modules automatically configure and update when you connect them to the phone.

### To set up a VVX EM50 expansion module

- 1 Connect a VVX EM50 expansion module to the USB port on the side of the VVX 450 business IP phone, OBi Edition.

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

## VVX EM50 Expansion Module Line Keys

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

The line keys on the expansion module are divided into 15 left and 15 right keys on each page. Each expansion module supports three pages, extending its line keys to 90.

See [Feature Keys](#) for more information.

## VVX EM50 Expansion Module Background Picture

The default background image on the VVX EM50 expansion module (side car) is the same as the phone background. When you set the background image on the phone, the expansion module background automatically updates. If you attach more than one EM50 expansion module, both show the same background picture.

You can load a different background image for the expansion module by specifying the image in the **User Preferences::SideCarBackgroundPicture** parameter. The optimum image resolution is 480pixels wide by 800 pixels high.



# Audio Features

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You can create customized tone patterns and ring tones for your VVX business IP phones, OBi Edition.

## Tone Patterns



Tone and Ring Profile A default settings are set for North American telephone standards. Tone and Ring Profile B default settings are set for Australian telephone standards. You can download tone profiles for other countries from the OBITALK forum.

## *Tone Profile Features of Your Phone*

Tone profiles use the following format:

```
[field-1];[field-2];[field-3];...;[field - 6]
```

Use a semicolon to separate the configuration fields. You can't use spaces in the tone profile patterns.

### Field-1 Composition

Field-1 describes frequency components used for tone synthesis. It supports as many as three different frequencies.

The frequency expression is a string of numeric values with the notation + or -. The numeric values are the frequency's decimal values in Hz and amplitude in dBm (maximum 3 dBm).

Different frequencies are separated by a comma.

**Example:** 350-18,440-18,550+2

- The first frequency is at 350 Hz with strength at -18 dBm
- The second frequency is at 440 Hz with strength at -18 dBm.
- The third frequency is at 550 Hz with strength at +2 dBm

### Field-2 Composition

Field-2 describes the overall tone playback duration in seconds.

The expression is a numeric value, and supports as many as three decimated digits. The numeric value can be negative, zero, positive, or skipped:

- Negative value: Tone plays indefinitely
- Zero value: Tone playback is skipped
- Positive value: Normal playback duration

- No value: Tone plays indefinitely

**Example:** 30.234

- Tone playback terminates after 30.234 seconds.

## Field–3 to Field–6 Composition

Fields 3 through 6 share the same definition. Each field describes one single cadence segment. Together, these four fields form a macro-segment, which repeats until tone playback expires.

The expression is a string of numeric values with the special notation '/', '(', ')', and '!'.  
 $t(f_0/on_0+off_0, f_1/on_1+off_1, f_2/on_2+off_2, f_3/on_3+off_3)$

$t(f_0/on_0+off_0, f_1/on_1+off_1, f_2/on_2+off_2, f_3/on_3+off_3)$

- $t$ : The cadence segment duration in seconds.
  - Negative value: Tone plays indefinitely
  - No value: Tone plays indefinitely
  - Zero value: The duration of this particular segment is zero
  - Positive value: Normal playback duration
- $f_0/1/2/3$ : The frequency component(s) used for the synthesis. Use one of following options (0 through 7):
  - 0: No frequency specified (silence)
  - 1: The first frequency
  - 2: The second frequency
  - 3: The first and second frequencies
  - 4: The third frequency
  - 5: The first and third frequencies
  - 6: The second and third frequencies
  - 7: The first and second frequencies if two or more than two frequency components, or the first frequency if only one frequency component is available

If you don't provide a value for  $f_0/1/2/3$ , the phone automatically uses the combination of the first one or two available frequency components.

- $on_0/1/2/3$ : The tone active time in seconds.
  - Negative value: Not allowed
  - No value: Infinite tone active time
  - Others: Normal tone active time (as many as three decimated digits)
- $off_0/1/2/3$ : The tone inactive time in seconds
  - Negative value: Not allowed
  - No value: Infinite tone inactive time
  - Others: Normal tone inactive time (as many as three decimated digits)

**Example:** 4(1/.3+2.34, 3/2+1.5)

- Use the first frequency to generate a tone for 0.3 seconds.
- Follow this tone with 2.34 seconds of silence.

- Use a combination of the first and second frequencies to generate a tone for 2 seconds.
- Follow this tone with 1.5 seconds silence.
- The cadence repeats for 4 seconds.

## ***Tone Examples***

These examples show the interpretation of a few common tone patterns:

### ***Dial Tone***

DIAL, "350-18,440-18"

Dial tone is generated as a mixture of two frequency components:

350 Hz at -18 dBm and 440 Hz at -18 dBm

The expiration time is infinite, and tone active time is infinite.

### ***Busy Tone***

BUSY, "480-18,620-18;10;(.5+.5)"

Busy tone is generated as a mixture of two frequency components:

480 Hz at -18 dBm and 620 Hz at -18 dBm

The expiration time is exactly 10 seconds. It has only one cadence segment, which has tone active 0.5 second and tone inactive 0.5 second.

### ***Prompt Tone***

PROMPT, "480-16;10"

Prompt tone is generated from a single frequency component:

480 Hz at -16 dBm. The expiration time is exactly 10 seconds. It has only one cadence segment, which has tone infinite active time.

### ***SIT Tone***

SIT\_1, "985-16,1428-16,1777-16;20;(1/.380+0,2/.380+0,4/.380+0,0/0+4)"

Special information tone (SIT) is generated from a set of frequency components:

- First frequency: 985 Hz at -16 dBm
- Second frequency: 1428 Hz at -16 dBm
- Third frequency: 1777 Hz at -16 dBm

The expiration time is exactly 20 seconds. It has only one cadence segment, which includes four on-off sections. The segment has infinite repeating time:

- The first on-off section: generated by the first frequency component, and it has 0.38 tone second active time and 0 inactive time.
- The second on-off section: generated by the second frequency component, and it has 0.38 tone second active time and 0 inactive time.

- The third on-off section: generated by the third frequency component, and it has 0.38 tone second active time and 0 inactive time.
- The fourth on-off section: only generate silence since no frequency component is specified. It has tone 0 second active time and 4 seconds inactive time.

## Stutter Tone

STUTTER, "350-18,440-18;20;.2(.1+.1);()"

Stutter dial tone is generated from a mixture of two frequency components:

350 Hz at -18 dBm and 440 Hz at -18 dBm. The expiration time for the entire tone is exactly 20 seconds. It has two cadence segments.

- The first segment includes only one on-off section, on 0.1 second and off 0.1 second, and on-off repeats for 2 seconds.
- The second segment includes one on-off section, and has infinite repeating time and infinite tone active time, and plays until the entire tone duration has elapsed.

For more information on Tone Profile A & B parameters, see the [Tone Profile X Web Page \(X = A, B\)](#) table in the [Device Web Page and Configuration Parameter Reference](#) section.

For more information on call waiting parameters, see the [Tone Profile X Web Page \(X = A, B\)](#) table in the [Device Web Page and Configuration Parameter Reference](#) section.

## Ring Tones and Ring Patterns

The general format of an OBi ring profile is [field-1];[field-2];...;[field - 5]

Use a semicolon to separate as many as five configuration fields. You can't use spaces in the ring profile patterns.

### Field-1 Composition

Field-1 describes the overall ringing duration in seconds.

The expression is a numeric value and supports as many as three decimated digits.

The numeric value can negative, zero, and positive:

- Negative value: Ringing lasts indefinitely
- No value: Ringing lasts infinitely
- Zero value: Ringing is skipped
- Positive value: Normal ringing duration

**Example:** 30.5

- This example illustrates a ringing tone that ends after 30.5 seconds.

### Field-2 to Field-5 Composition

Fields 2 through 5 share the same definition, and each field describes one single cadence segment. Together, the four fields form a macro-segment, which repeats until ringing expires.

The expression is a string of numeric values with the special notation '(', ')', and ','.

`t(on_0+off_0,on_1+off_1,on_2+off_2,on_3+off_3)`

- `t`: The cadence segment duration in seconds.
  - Negative value: Ringing indefinitely
  - No value: Ringing indefinitely
  - Zero value: Ringing is skipped
  - Positive value: Normal ringing duration
- `on_0/1/2/3`: The ring active time in seconds.
  - Negative value: Not allowed
  - No value: Infinite ring active time
  - Others: Normal ring active time (as many as three decimated digits)
- `off_0/1/2/3`: The ring inactive time in seconds.
  - Negative value: Not allowed
  - No value: Infinite ring inactive time
  - Others: Normal ring inactive time (as many as three decimated digits)

**Example:** `4(.3+2.34,2+1.5)`

- This example illustrates a ringing tone comprised of two segments.
- Ringing is active for 0.3 seconds, followed by 2.34 seconds of silence, then ringing for 2 seconds, and followed by 1.5 seconds of silence. This cadence operates repeatedly for 4 seconds.

# Voice Services

---

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
- OBiTALK

An  $SP_n$  service can be a generic SIP voice service. SIP/SP services include an extension from a PBX or a subscriber account with a service provider. Every SP service user account requires a user name and usually a password for authentication. The service provider assigns an extension number or DID number to the user account. The assigned number may be the same as the user name of the account.

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

## ITSP Profiles

The configuration of an SP service is divided into a service provider part and a service subscriber part.

### *Service Provider Part*

The service provider part 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.

### *Service Subscriber Part*

The service subscriber part comprises of parameters that may vary for each specific subscriber account from the service provider.

The service subscriber, known as an SP service, includes the  **$SPn::X\_ServProvProfile$**  parameter that binds the SP service with an ITSP Profile. By default, the  **$SPn::X\_ServProvProfile$**  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.



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

## Audio Codec Profiles

Two codec profiles are available on your phone, and they're selectable per trunk (OBiTALK, SP $n$ ,  $n = 1$  through 6). To select a codec as the preferred codec in this profile, set the priority of that codec to be highest among all the enabled codecs in the profile. You can assign each of the SP and OBiTALK services to a codec profile in its corresponding configuration. The codec list to use when setting up a call on the underlying service is formed from the list of enabled codecs in the chosen profile and ordered according to the assigned priorities in the profile. For codecs with the same priority setting, the codec that appears first on the codec profile web page has a higher priority.

## Specifying Target Phone Numbers

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

Two formats specify a target phone number:

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

The case-insensitive service names to use for each service in the full number specifications the following:

- sp $n$  for SP $n$  Service for  $n = 1 - 6$
- pp for OBiTALK Service

## Overview of Common Trunk Configuration

Trunks of every kind share some common characteristics that you can configure.

### *Trunk Capacity*

Trunk Capacity is the maximum number of simultaneous calls allowed on the trunk. You can set the value for each service using the `MaxSessions` parameter. The default value is 2 for all services. For OBiTALK service, the maximum value is 4. For other SIP/SP services, do not set the value higher than the maximum number of simultaneous calls allowed by the service provider.

## Service Enable

Before you put a trunk into service, enable the trunk in the phone configuration. There's an `Enable` parameter for each service, and this parameter is checked by default.

## Service Account Credentials

All SP services require credentials, but OBiTALK services don't require credentials. At the minimum, you must configure a user name for the SP service account using the following parameters:

- **`SPn Service- Service Credentials::AuthUserName`**
- **`SPn Service- Service Credentials::AuthPassword`** (Optional; use if you require a password for server authentication)

In a less common situation where the user name used for SIP authentication is different from the account user name, set the account user name in **`SPn Service- Service Credentials::URI`** and the different user name for authentication only in `AuthUserName` (required). If `URI` isn't specified, it's used for both as the account user name and for SIP authentication. Note also that if `AuthUserName` isn't specified, your phone considers the service as disabled.

## Servers

OBi Edition phones support the following servers and voice services:

- SIP servers
- OBiTALK servers

No matter what technology the service provider side 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). Note that since OBiTALK servers are known by your phone, there's no need to configure the server domains for this service. For other SIP/SP services, configure the proxy server in the **`ITSP Profile X – SIP::ProxyServer`** parameter. If the listening port is nonstandard, configure the correct value in **`ITSP Profile X – SIP::ProxyServerPort`**. The `OutboundProxy` parameter in the same parameter group is often needed when the device-facing server is a session border controller (SBC). Similarly if the outbound proxy isn't listening at the standard port, configure the correct port value in `OutboundProxyPort`. The `RegistrarServer` and `RegistrarServerPort` parameters are rarely needed; your phone assumes that the SIP Registrar is the same as the SIP Proxy Server if they aren't specified separately.

SIP Transport refers to the transport protocol to use to exchange SIP messages with the server: UDP, TCP, and TLS, which are all supported by your phone. Use the **`ITSP Profile X – SIP::ProxyServerTransport`** parameter to configure the transport protocol. 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 `OutboundProxy` is specified, your phone starts the TCP/TLS connection with the `OutboundProxy` instead. With the `OutboundProxyTransport` parameter, it's possible to choose a different transport to use with the `OutboundProxy` and with the `ProxyServer`.

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.



## SIP/SP Services

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

Each ITSP configuration is grouped as an ITSP Profile. There are six ITSP profiles available, referred to as ITSP Profile A, B, C, D, E, and F, respectively. SP service account specifics are grouped under the heading *SP<sub>n</sub> Service*, where  $n = 1 - 6$ . An ITSP Profile includes the `ProxyServer`, `OutboundProxy`, and `DigitMap` parameters, but, it doesn't include account-specific parameters. An SP Service includes account-specific parameters such as `AuthUserName` (usually, but not necessarily, the same as the phone number of the account), `AuthPassword`, `CallerIDName`, and `X_ServProfile` (which ITSP Profile to apply the ITSP-specific parameters from). If both SP Services use the same ITSP, it's usually possible to configure just one ITSP Profile with both SP Services referring to the same profile. However, if abstraction of an ITSP Profile isn't sufficient to cover a particular 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.

The *SP<sub>n</sub> Service* using ITSP Profile X is enabled by your phone only when at least the following parameters are set:

- **ITSP Profile X – SIP::ProxyServer** = Not Blank
- **SP<sub>n</sub> Service::Enabled** = true (or checked on the native phone web page)
- **SP<sub>n</sub> Service::AuthUsername** = Not Blank

where  $X = A - F$  and  $n = 1 - 6$ . Otherwise, your phone considers the service disabled.

## OBiTALK Service

OBiTALK is a proprietary protocol for communications among OBi Edition phones and to OBiTALK device management servers. The protocol is intended for two main purposes:

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

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

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

The **OBiTALK Service::Enable** parameter enables the OBiTALK service by default, unless disabled through ZT customization.

You may want to limit OBiTALK calls just to the OBiHai echo server. To do this, the administrator can change the value of **OBiTALK Service::DigitMap** to: (`<ob>222222222|ob222222222`). You can change or add more OBi numbers to this digit map by following the same pattern. A simple way to disable OBiTALK voice calls completely is by setting **OBiTALK Service – Calling Features::MaxSessions** to 0. You can't do echo test in that case.

# Directories and Contacts

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Your phone supports a directory search function with an external server using the Lightweight Directory Access Protocol (LDAP). To use this function, you must configure an LDAP service on the phone. Users then invoke the LDAP directory search application by selecting the option from the phone main menu or by pressing a softkey.

## LDAP Service Setup

You can point the **Network Directory** option on the main menu of the phone to an LDAP service using the following parameters.

### LDAP Parameters

Parameter Group	Parameter	Description
<i>IP Phone – LDAP – Server</i>	<b>Host</b>	The hostname can be an IP address or domain name, with optional <code>ldap://</code> or <code>ldaps://</code> scheme prefix. For example: <code>192.168.15.186</code> , <code>ldap.forums.com</code> , <code>ldap://ldap.testathon.net</code> are all acceptable hostname formats. If scheme is not specified, <code>ldap://</code> is implied.
<i>IP Phone – LDAP – Server</i>	<b>Port</b>	LDAP Server listening (TCP) port. The standard port is 389 for <code>ldap://</code> and 636 for <code>ldaps://</code> . If the port value is 0 or blank, the phone uses the corresponding standard port.
<i>IP Phone – LDAP – Server</i>	<b>Password</b>	The Bind password for Simple or SASL authentication Note that this parameter is case-sensitive.
<i>IP Phone – LDAP – Server</i>	<b>TLSSecurityProfile</b>	The security profile for 802.1X authentication.

LDAP Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LDAP – Search Parameters</i>	<b>BindDN</b>	<p>A Distinguished Name (DN) that is authorized to use the LDAP service. If none is specified, the query is regarded as an anonymous one which may or may not be acceptable to the server.</p> <p>The <code>BindDN</code> value is usually derived from a user name that typically looks like an email address, such as <code>admin@ldap.example.com</code>. In which case, the corresponding <code>BindDN</code> would be:  <code>CN=admin,OU=users,DC=example,DC=com</code></p> <p>Note that in the last example, the DN includes only the last two parts of the domain name for illustration purpose only. The DN may as well include the <code>DC=ldap</code> field, for example. But the key point is that this value must agree exactly with how it is specified at the server side.</p>
<i>IP Phone – LDAP – Search Parameters</i>	<b>SearchBase</b>	<p>This parameter specifies the starting point of the LDAP search. It is a case-insensitive comma-separated list of <code>{object}={value}</code> pairs, where <code>{object}</code> can be any of the following:</p> <ul style="list-style-type: none"> <li>• <code>CN</code> (Common Name)</li> <li>• <code>OU</code> (Organization Unit)</li> <li>• <code>O</code> (Organization)</li> <li>• <code>C</code> (Country)</li> <li>• <code>DC</code> (Domain)</li> </ul> <p>If the value is not specified, the phone by default derives the search starting point from the value of <b>LDAP – Server::Host</b>. For example, if <code>Host</code> is <code>ldap.example.com</code>, the default <code>SearchBase</code> value is <code>DC=example,DC=com</code></p> <p>Note that it is a common convention to use just the last two parts of the service domain as a search base, but it is not necessarily so. The phone assumes this convention when <code>SearchBase</code> is not specified.</p>
<i>IP Phone – LDAP – Search Parameters</i>	<b>ProtocolVersion</b>	<p>Protocol version. Either 3 or 2. 3 is the default.</p>
<i>IP Phone – LDAP – Search Parameters</i>	<b>TLS_ReqCert</b>	<p>Controls whether to verify the server's certificate on a TLS connection. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>never</code></li> <li>• <code>demand</code></li> </ul> <p>Default value is <code>never</code>, which means not to verify the server's certificate.</p>

LDAP Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LDAP – Search Parameters</i>	<b>ResultsPerPage</b>	<p>Specifies how many results to display on screen per page. Valid values are:</p> <ul style="list-style-type: none"> <li>• 20</li> <li>• 40</li> <li>• 60</li> <li>• 80</li> <li>• 100</li> </ul> <p>Default value is 20.</p>
<i>IP Phone – LDAP – Search Parameters</i>	<b>DefaultSearchFilter</b>	<p>Default search filter to append to each search. This must be specified as a complete and valid LDAP search filter. For example:</p> <pre>( (objectclass=contact)(objectclass=person))</pre> <p>Default value is: (objectclass=*)</p>
<i>IP Phone – LDAP – Search Parameters</i>	<b>QueryFields</b>	<p>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</p> <ul style="list-style-type: none"> <li>• {ldap-attr} is the standard ldap-attribute-name, such as sn, givenName, telephoneNumber, cn, . . . . This is the only required attribute in each field.</li> <li>• {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.</li> <li>• {type} is either A or N, for alphanumeric or number type respectively and is case insensitive. If not specified, it is assumed to be A.</li> </ul> <p>For example: givenName/First Name, TelephoneNumber, Mobile</p> <p>Default value is: givenName/First Name, sn/Last Name, telephoneNumber/Tel, mobile/Mobile, homePhone/Home</p>

LDAP Parameters (continued)

Parameter Group	Parameter	Description
<b>IP Phone – LDAP – Search Parameters</b>	<b>ResultFields</b>	<p>Comma-separated list of LDAP fields to display for each entry of the search result. Each field has three attributes separated by two slashes (/):  <code>{ldap-attr}/{caption}/{type}</code> where</p> <ul style="list-style-type: none"> <li><code>{ldap-attr}</code> is the standard ldap-attribute-name, such as <code>sn</code>, <code>givenName</code>, <code>telephoneNumber</code>, <code>cn</code>, . . . This is the only required attribute in each field.</li> <li><code>{caption}</code> is optional. This is the caption to display on the screen for the displayed value. If not specified, the <code>{ldap-attr}</code> value is used in its place.</li> <li><code>{type}</code> is either <code>S</code>, <code>C</code>, or <code>N</code>, for String, Callable Number, or Picture, respectively. It is case insensitive. If not specified, it is assumed to be <code>S</code>.</li> </ul> <p>Default value is:  <code>cn/Common Name,sn/Last Name,givenName/First Name,telephoneNumber/Tel/y,mobile/Mobile/y,homePhone/Home/y</code></p>
<b>IP Phone – LDAP – Search Parameters</b>	<b>NameFieldPreference</b>	<p>Comma-separated list of LDAP attributes to be used as the Caller ID Name to display on screen, ordered by preference. The first non-empty value in the list is used.                      Default value is <code>cn,givenName sn</code>.</p>
<b>IP Phone – LDAP – Search Parameters</b>	<b>NumberFieldPreference</b>	<p>Comma-separated list of LDAP attributes to be used as the Number to display on screen and to call by default, ordered by preference. The first non-empty value in the list is used.                      Default value is <code>telephoneNumber,mobile,homePhone</code>.</p>
<b>IP Phone – LDAP – Search Parameters</b>	<b>PhotoFieldPreference</b>	<p>Comma-separated list of LDAP attributes to be used as the photo to display on screen, ordered by preference. The first non-empty value is used.                      Default value is <code>thumbnailPhoto</code>.</p>
<b>IP Phone – LDAP – Search Parameters</b>	<b>SortByAttribute</b>	<p>The LDAP attribute to use for sorting the search results by the server. Default value is <code>cn</code></p>
<b>IP Phone – LDAP – Search Parameters</b>	<b>Attribute</b>	<p>Comma-separated list of attributes to request from server.</p>

SASL Parameters (not applicable for LDAP version 2)

## LDAP Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – LDAP – SASL Authentication Parameters</i>	<b>SASL_AuthMethod</b>	Method to use for SASL authentication. If the value is Disabled, the phone uses Simple authentication only. Choice of: <ul style="list-style-type: none"> <li>Disabled (the default)</li> <li>Plain</li> <li>MD5</li> </ul>
<i>IP Phone – LDAP – Search Parameters</i>	<b>SASL_AuthCID</b>	The authentication ID for SASL authentication. The format of this ID depends on the actual SASL mechanism used.

## Client Authentication

LDAP v2 supports `ldap://` and `ldaps://` with Simple Authentication only. LDAP v3 adds support for TLS and SASL Authentication. Simple authentication involves sending the LDAP server the FQDN of the client and the corresponding password in clear-text, which has security issues unless you use `ldaps://` or TLS.

Simple Authentication and Security Layer (SASL) [RFC2222] is a framework for authentication. To use SASL, you must set the **LDAP – SASLAuthMethod** parameter to either `Plain` or `MD5`.

For more informations on each of these SASL mechanisms, go to <http://www.openldap.org>.

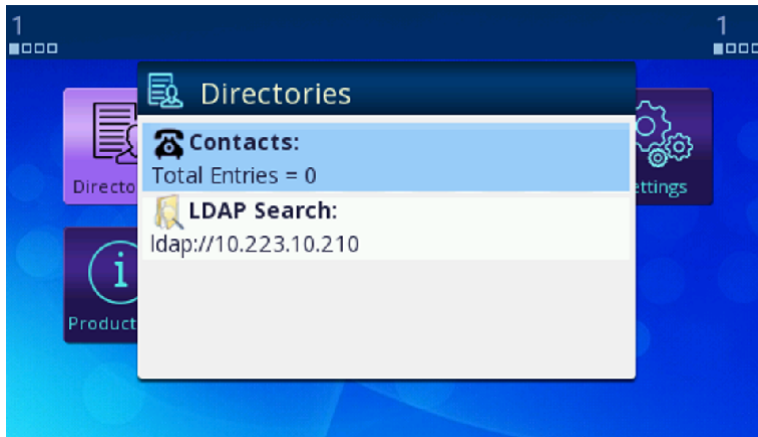
## LDAP Directory Search Application

Users can start the LDAP application on the phone from the **Main** menu or with a softkey (with ID = ldap).

### Invoke LDAP by Main Menu – Directories Option

Include the Main menu Item ID directories in the **Phone Settings – GUI Menus::MainMenu1** parameter. When the LDAP service is properly configured, the LDAP option appears under the **Directories** menu, as shown:

## LDAP display example



## Invoke LDAP by Main Menu – Network Directory Option

First you must include the Main menu item ID `netdir` in the **Phone Settings – GUI Menus::MainMenu1** parameter. To make the **Network Directory** option on the Main menu invoke the LDAP service, use the following settings:

**IP Phone Settings – Network Directory::Enable** = `true` (checked)

**IP Phone Settings – Network Directory::VoiceService** = `LDAP`

## Invoke LDAP by Softkey – LDAP

To use a softkey to invoke the LDAP application, add the `ldap` softkey to any of the configurable softkey sets. For example, configure the following value for the **Home** softkey set with the LDAP softkey in the fourth position:

```
redial,cfwd,dnd,ldap
```

## Search Fields

By default, your phone displays the following search fields:

- Last Name (sn)
- First Name (givenName)
- Tel Num (telephoneNumber)
- Mobile Num (mobile)
- Home Num (homePhone)

One or more of these search fields may be specified per query; all the search criteria are added together to form the search filter, and the value is case-insensitive. To specify partial match of an attribute value, use a `*` wildcard character. For example: set Last Name = `s*` to query all entries with the sn that starts with an s or S.

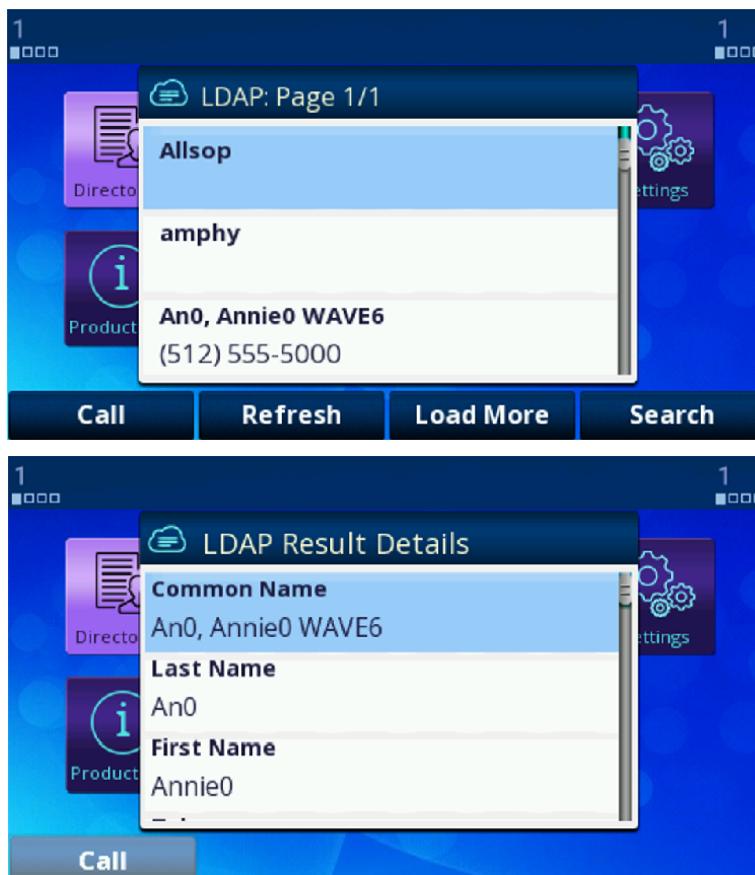
You can customize the search fields presented to you by specifying the fields in the `QueryFields` parameter. This is a comma-separated list of fields. Refer to the LDAP parameter table above for the syntax.

## Result Fields

The search results are presented in a table where a Name, a Number, and a Picture are shown, when the values are available. Which LDAP attributes in the result are to be used for the Name, Number, and Picture display can be controlled by setting the `NameFieldPreference`, `NumberFieldPreference`, and `PhotoFieldPreference` parameters. Each of these parameters is a comma-separated list of LDAP attributes arranged in order of preference. The first attribute in the list that has a non-empty value is used for the corresponding display field.

You can see more attribute values of an entry by highlighting the entry on the top-level search result table and pressing the **OK** key on the phone. This brings up the LDAP Result Details screen of the selected entry. You can customize the attributes to display on this screen with the `ResultsFields` parameter, which is a comma-separated list of fields to present to you. Refer to the LDAP parameter table for the syntax of this parameter.

### Search result examples





## Sorting of Results

The phone relies on the server to sort the results. Sorting is based on a single LDAP attribute which can be customized with the `SortByAttribute` parameter. The default value is `cn`.

## Replace the Built-In Phone Book with LDAP

You can replace the built-in phone book with LDAP, such that when you press the **phbk** softkey or select **Contacts** from the Main menu, it launches the LDAP feature instead of the built-in (local) phone book. Below is the configuration for this:

**User Preferences – Phone Book Settings::ActionURL** = `phone://netdir`

**Phone Settings – Network Directory::VoiceService** = `LDAP`

## Import/Export Phone Book Contents

To enable Phone Book import from a PolyPhoneBook XML, specify a URL in the `ImportURL` parameter (under User Preferences page Phone Book Settings. When a valid `ImportURL` is configured into the phone, the Import softkey is shown on the top level screen in the Contacts app, by which you can invoke the import operation. For example:

```
ImportURL = http://192.168.15.1 0/myserver.com/apolyphonebook?mac=$MAC,no-verify,tlsprofile=1
```

Note that the comma-separated parameters after the URL are optional and are used to control the HTTP connection only.

### Phone Book Import/Export

Web Page	Section	Parameter	Description
User Preferences	Phone Book Settings	ImportURL	A URL for importing into the phone book from a PolyPhoneBook XML
User Preferences	Phone Book Settings	ExportURL	A URL for uploading the internal phone book as a PolyPhoneBook XML. You can append an optional comma-separated parameter <b>servertype=type</b> to the URL; type can be 0 or 1. See the examples for the meaning of each type.

To enable Phone Book export to a PolyPhoneBook XML, specify a URL in the `ExportURL` parameter (under User Preferences page Phone Book Settings. If a valid `ExportURL` parameter is configured into the phone, the Export softkey is shown on the top level screen of the Contacts app, by which you can trigger the export operation. Note that the phone tries to save the file on the server with the filename `PolyPhoneBook-yearmonth-day-hh-mm-ss.xml`, where year-month-day-hh-mm-ss is the current date-time in GMT. For example:

```
ExportURL = http://192.168.15.100/hfs,server-type=1
```

Note that `server-type` is an optional comma-separated parameter to control the format of the HTTP Post message body. If `server-type` is not specified, the phone sends a message body like this:

```
POST /hfs/ HTTP/1.1
Host: 192.168.15.100
User-Agent: VVX450
Cache-Control: max-age=0
Accept-Language: en-us, en
Accept-Charset: ISO-8859-1, utf-8
Content-Type: application/xml
Content-Length: 362
Connection: Close
```

```
<?xml version="1.0" encoding="utf-8"?>
<PolyPhoneBook version="1.0">
<item>
<name>Cae Samson</name>
<number>999563321</number>
<ringtone>/data/ringtones/Office B.wav</ringtone>
<mobilenumber>9996632356</mobilenumber>
<officenum>9995563321</officenum>
<email>asam@abce.com</email>
<defaultnumber>generic</defaultnumber>
</item>
</PolyPhoneBook>
```

**When server-type=1 (for example, HFS), the message body would be like this:**

```
POST /hfs/ HTTP/1.1
Host: 192.168.15.100
User-Agent: VVX450
Cache-Control: max-age=0
Accept-Language: en-us, en
Accept-Charset: ISO-8859-1, utf-8
Content-Type: multipart/form-data; boundary=573e34e87ee05a3d
Content-Length: 540
Connection: Close
```

```
--573e34e87ee05a3d
Content-Disposition: form-data; name="file";
filename="PolyPhoneBook-2019-10-1-20-47-37.xml"
Content-Type: application/octet-stream
```

```
<?xml version="1.0" encoding="utf-8"?>
<PolyPhoneBook version="1.0">
<item>
<name>Cae Samson</name>
<number>999563321</number>
<ringtone>/data/ringtones/Office B.wav</ringtone0>
<mobilenumber>9996632356</mobilenumber>
<officenumber>9995563321</officenumber>
<email>asam@abce.com</email>
<defaultnumber>generic</defaultnumber>
</item>
</PolyPhoneBook>
--573e34e87ee05a3d--
```

When server-type=1 (for example, Google Drive), the message body looks like this:

```
POST /hfs/ HTTP/1.1
Host: 192.168.15.100
User-Agent: VVX450
Cache-Control: max-age=0
Accept-Language: en-us, en
Accept-Charset: ISO-8859-1, utf-8
Content-Type: multipart/related; boundary=476f0c72febd8223
Content-Length: 572
Connection: Close
```

```
--476f0c72febd8223
Content-Type: application/json; charset=UTF-8
```

```
{
"name": "PolyPhoneBook-2019-10-1-20-48-40.xml"
}
```

```
--476f0c72febd8223
Content-Type: application/octet-stream
<?xml version="1.0" encoding="utf-8"?>
<PolyPhoneBook version="1.0">
<item>
<name>Cae Samson</name>
<number>999563321</number>
<ringtone>/data/ringtones/Office B.wav</ringtone>
<mobilenumber>9996632356</mobilenumber>
<officenumber>9995563321</officenumber>
<email>asam@abce.com</email>
<defaultnumber>generic</defaultnumber>
</item>
</PolyPhoneBook>
--476f0c72febd8223--
```

# Call Controls









You can configure call features as a phone-level feature or a line-level feature. A phone feature applies to all calls on the phone regardless which line a call is on. Call waiting, for example, is a phone feature. A line feature applies only to calls on a specific line. For example, busy lamp field (BLF) is a line feature.

Some features may have both a phone feature version and a line feature version. For example, you can provision Do Not Disturb as a phone feature for all calls and as a line feature for each line for the SP services (SP1–SP6) and OBiTALK services.









## Call States

As a call progresses from beginning to end, it goes through a number of defined stages commonly known as call states or states. The following table includes the call states for calls on your phone.

### List of Call States

Icon	Call State	Description	Available Operations	LED Pattern
	Dial Tone	Dial tone plays to prompt user to enter the target number to call.	End	Steady green
	Dialing	User is entering a target number to call.	End	Steady green
	Trying	Trying to call the dialed number but the called party hasn't started ringing yet.	End	Steady green
	Peer Ringing	The called party is ringing.	End	Fast blinking green
	Ringing	An incoming call is ringing the phone.	Reject, Answer	Fast blinking red
	Connected	Connected on a call and both sides are talking.	End, Hold, Transfer, Blind Transfer	Steady green
	Holding	User has placed the call on hold.	End, Resume, Add to Conf., Transfer, Blind Transfer	Slow blinking red
	Ended	Call failed due to various reasons, such as invalid number, service not available, called party busy, and so forth.	End (Remove the Call)	Slow blinking green

**List of Call States (continued)**

Icon	Call State	Description	Available Operations	LED Pattern
	Idle	No Call.	—	Off
	Registration Error	Error registering with the server.	—	Off
	Remote Hold	Remote side placed the call on hold.	End, Hold, Transfer, Blind Transfer	Steady green
	Messages Waiting	Idle state with unread messages in the mailbox.	—	Off
	Connected-HD	Call connected with HD Audio.	End, Hold, Transfer, Blind Transfer	Steady green
	Connected-HDS	Call connected with Secure HD Audio.	End, Hold, Transfer, Blind Transfer	Steady green
	Connected-S	Call connected with Secure Audio.	End, Hold, Transfer, Blind Transfer	Steady green
	Service Down	Service down.	—	Off

## Core Call Features

Your OBi Edition phone provides two types of call features: core and premium features. Core call features include the following:

- Managing line capacity
- Managing complex operations between devices
- Making outgoing calls
- Handling incoming calls
- Ending, holding, resuming, and foregrounding calls
- Managing call waiting and call transfer
- Managing conference calls
- Setting up and using auto-attendant and auto-answer services
- Managing convenience features such as speed dial, caller ID, and anonymous call handling
- Forwarding calls

### *Line Capacity*

Line capacity refers to the maximum number of simultaneous calls that can be active per line. Some of the calls may be in the Holding state. The configuration of each voice service has a `MaxSessions` parameter

that controls how many simultaneous calls to allow on that service. The default value is 2 for all lines. Set the number equal to or less than what the underlying service provider can support.

## Making Outgoing Calls

### Digit Map

A digit map is a succinct way of describing a set of number patterns. The **Phone Settings::DigitMap** parameter determines the set of number patterns that users can dial. You can refer to named digit maps with the `(Mname)` syntax. For example, the default **Phone Settings::DigitMap** parameter refers to digit maps defined for SP1, SP2, SP3, SP4, and OBiTALK services with the reserved names `(Msp1)`, `(Msp2)`, `(Msp3)`, `(Msp4)`, and `(Mpp)` respectively:

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

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



The `(Mpli)` refers to the digit map of the Primary Line.

### Outbound Call Routes

After a user dials a complete number, the phone determines which service to use for the call by applying the call routing rules defined in the **Phone Settings::OutboundCallRoute** parameter. This parameter may refer to named digit maps defined elsewhere in the phone configuration. The default value as shown below refers to `(Msp1)`, `(Msp2)`, `(Msp3)`, `(Msp4)`, and `(Mpp)`, which are digit maps defined for SP1, SP2, SP3, SP4, and OBiTALK services, respectively:

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



The `pli` refers to the Primary Line.

### Primary Line

The Primary Line is the preferred line to use for outgoing calls when the number entered by users doesn't include a line-section-prefix (such as `**1` or `**2`). Define the Primary Line in the **Phone Settings::PrimaryLine** parameter using one of the following case-sensitive values:

- SP1 Service
- SP2 Service
- SP3 Service
- SP4 Service
- SP5 Service

- SP6 Service
- OBiTALK Service
- Trunk Group 1
- Trunk Group 2

The reserved name `pli` found in the phone `DigitMap` and `OutboundCallRoute` parameters is substituted by the corresponding name of the primary line: `sp1`, `sp2`, `sp3`, `sp4`, `sp5`, `sp6`, `pp`, `tg1`, and `tg2` for SP1, SP2, SP3, SP4, SP5, SP6, OBiTALK, Trunk Group 1, and Trunk Group 2, respectively.

## Dialing “Speed Dials 99” Numbers

You can define as many as 99 speed dial numbers in the configuration. Speed Dial numbers are placed as 1 – 99, and you can define the corresponding one/two-digit number (that is, 1 – 99) in **User Settings - Speed Dials**. These speed dial numbers are unrelated to the Speed Dial feature keys.

You can associate the speed dial numbers with phones or devices reachable via an Internet or landline service or the OBiTALK network. Be careful with the Speed Dial setup as this may conflict with any Speed Dials set up on the OBiTALK portal. The Speed Dials you set up on the OBiTALK portal always overwrite anything setup via the phone connected to the OBi. It’s also possible for a user to set up a Speed Dials 99 number using a star code (by default, the star codes for this are \*74, \*75, and \*76).

## Handling Incoming Calls

### Inbound Call Routes

Incoming calls come to the phone via any configured Line. How the phone should route 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`.

### Caller ID Display

During a call, the caller’s name and number displays on the screen when available. When only the caller number is available, the phone attempts to match the number against the phone book or buddy list to find a corresponding entry. If the phone finds a matching entry, it displays the name and picture from that entry on the screen.

If the call comes from a SIP service, the service provider provides the caller-id picture in the INVITE request by including a Call-Info header with a (http or https) URL to the picture file and the attribute “purpose=icon”. For example:

```
Call-Info: <http://abc.com/user/somepic.png>;purpose=icon;org="ABC Publishing"
```

These are the acceptable picture formats: `png`, `bmp`, `gif`, and `jpg`. Notice that in the last example, the header includes an `org` attribute. This value propagates the **Organization** field in the Caller-ID display.

You can also 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 Set):

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

## Transfer Signaling

As transferor, if the call with the transferee is on  $SP_m$  and the call with the transfer target on  $SP_n$  where  $SP_m$  and  $SP_n$  point to the same ITSP Profile X (including the case  $m = n$ ) and **ITSP Profile X –**

**General::SignalingProtocol** is SIP and **ITSP Profile X – SIP::X\_UseRefer** is true, the phone attempts to transfer the call by sending a REFER request to the transferee with the transfer target's SIP URL in a Refer-To header. For other cases, the phone tries to bridge the transferee and transfer target call legs internally.

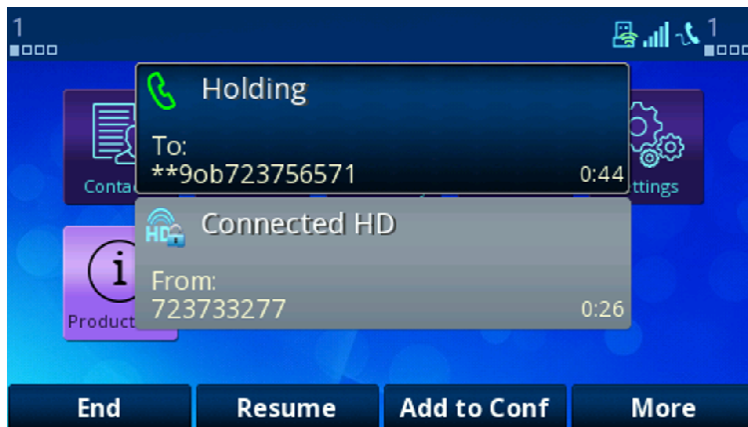
## Limitations of Transfer by Internal Bridging

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 would be acceptable by the other peer, the administrator should understand this limitation and configure the codec profiles accordingly. For example, if SIP service 1 is used, then a call transfer involving a SIP service 1 call leg must make sure the other call leg supports the G711U codec.

## Conference Calls

A conference call is a conversation involving 2 or more remote parties. In order to start a conference, there must be at least two calls and with at least one of them in the Connected State and one of them in the Holding State. The following picture shows such scenario with two calls. To start a conference, highlight one of the calls in the Holding State and select the **Add to Conf** softkey option.

### Starting a conference call



Your phone supports two methods to conference multiple parties:

- Local mixing/bridging
- External conference bridge

The UI is slightly different in each case.

## Local Mixing/Bridging

After starting a 3-way conference as described earlier, you can see the two remote parties both in the Connected State.



Note: The Opus codec supports 3-way calling with both legs using Opus on VVX 250, VVX 350, and VVX 450, but NOT on VVX 150.

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

## External Conference Bridge

To enable external conference bridge operation, configure the `X_ConferenceBridge` parameter on the `SPn Service` web page with the userid or (SIP) URL of the external conference bridge and also enable the option **Phone Settings – Calling Features::UseExternalConferenceBridge**. It should be noted that the phone assumes that only conferees that are on the same SP service or using the same ITSP profile as the conference bridge can be referred to the bridge. For conferees that are referable, the phone keeps them in the conference using local mixing and are subject to the local mixing limit. For example, if you have a conferee who is connected through the OBiTALK service, the phone keeps the call with that conferee in the Connected State as well as the call to the conference bridge in the Connected State and applies local mixing to the two calls.

## Auto Attendant

Your phone has an Auto Attendant (AA) feature that can be invoked by including `aa` as the destination of the inbound call routing rules of a trunk (such as SP1 or OBiTALK) and have incoming calls matching those rules on that trunk routed to the AA. You can specify one of two methods to connect with the AA in the routing rules: a) Ring the AA directly and have it answer the call normally after a configurable delay, or b) Have the AA call back the current caller or another number you designated. To use the AA feature on the phone, the **Auto Attendant – Auto Attendant 1::Enable** parameter must be enabled. The phone supports only two simultaneous AA calls. Additional calls routed to the AA are rejected as busy.



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

## AA Callback Service

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

The first method is by statically configuring it in a trunk's `InboundCallRoute`. A rule can be added to the `InboundCallRoute` parameter to have the AA call back the caller's or any other number, if the caller hangs up before the AA answers. The rule should indicate that `aa` (`{callback-number}`) is the target destination of the call, where `{callback-number}` is the number that the AA should call back if the caller hangs up before the AA answers the call. For example, the following rule:

```
{(<*1>(14089913313|12121559801)):aa($1)}
```

says that: if 14089913313 or 12121559801 calls, the call is routed to AA. If the caller hangs up before the AA answers, the AA calls the number represented by `$1`, which is a macro that is expanded into the caller number after processing by the digit map on the left side of the colon. In this particular example, the callback number is the caller's number prepended by `**1`. The outbound service to be used for the AA to call back is determined according to the **Auto Attendant – Auto Attendant 1::OutboundCallRoute** parameter.

The **Auto Attendant – Auto Attendant 1::CallbackAnswerDelay** parameter controls the number of milliseconds before AA answers when a callback number is specified as shown in the example. The default value is 10000 ms. Without the `{{callback-number}}` argument, the AA behaves in the normal way and the answer delay is governed by the `AnswerDelay` parameter (in milliseconds).

The second method is by selecting AA option 3 to “Enter a callback number” after the AA answers the call, the caller explicitly enters the number to be called back by the AA. If a valid number is entered, AA says “Thank You” and “Goodbye”, and then starts calling back 2 seconds after the current call has ended. If the number entered is invalid, the AA plays the SIT tone followed by an error message. Note that the variable `$1` (representing the caller’s number) is carried over to the subsequent AA callback call. The AA DigitMap can include `$1` to be used in a callback context. For example, the following rule in the AA DigitMap:

```
(<00:**1$1>|... )
```

says that if the AA dials 00, the device transforms it into the caller’s number prepended by `**1`. In other words, if the caller wants the AA to call back the current number (typically the case), they can enter `00#` after selecting option 3 on the AA menu. Note that `$1` can only be used as part of a substitution element in the digit map. It must not be used for matching elements since its value is unknown.

#### Automated Attendant

AA Option	Default AA Announcement	What Happens Next:
1	Press 1 to continue this call.	Ring the phone
2	Press 2 to make a new call.	If “UsePIN” authentication is enabled and you enter a matching PIN, the AA immediately prompts you to enter the PIN followed by the pound (#) key. If the entered PIN is not a match, the AA gives you two additional attempts to enter the PIN. If the third attempt does not match, the AA announces a thank you message and disconnects the call.
3	Press 3 to enter a callback number.	If a valid number is entered, AA says “Thank you” and “Goodbye”, hangs up, and then calls the number back in 2 seconds. If the given number is invalid, AA plays SIT tone followed by an error message. Tip: You can dial <code>00#</code> to have the AA call back your current number.

## Auto Answer and Intercom

Intercom, sometimes (called 2-way) paging, refers to a call with the following characteristics:

- The called phone answers automatically, usually immediately without ringing. If the handset is on the cradle, the speakerphone or the headset is turned on automatically as the call is answered. Typically the answering phone also plays a short beep to alert the called user right after answering the call. Use the option **User Preferences::IncomingPageAlertTone** to control whether the phone plays a beep when auto-answering. You can also tell the phone auto-mute when auto-answering with the option **User Preferences::AutoAnswerAutoMute**.
- The calling phone may support PTT (Push-To-Talk) in addition such that the call ends as soon as the caller releases the respective PTT key on the calling phone. PTT does not apply to the called phone. The called user talks and ends the call normally just like any other call.

In this document, an Intercom call isn't referred to as paging so it won't be confused with another paging application called Page Groups, which is based on Multicast/RTP/RTCP. See [Multicast Paging Groups](#) for more information.

The phone supports two methods to signal to the called device to auto-answer the 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

The method to use is controlled by the **ITSP Profile X – SIP::X\_AutoAnswerMethod** parameter.

Regardless of the method selected, for incoming calls, your phone processes the “answer-after” parameter in a Call-Info header or the “info” and “delay” parameters in an Alert-Info header, whichever is present in the inbound INVITE, by automatically answering the call after ringing for the number of seconds specified in those parameters (usually 0). When the call is automatically answered, all other current calls on the phone are interrupted the standard way as this is a new call added to the foreground. The auto-answer behavior can be turned off by disabling the **Phone Settings – Calling Features::AutoAnswerEnable** option. You can change the setting from the phone under the **Preferences** menu. The administrator may further define a feature key with the `Auto Answer On/Off` function to give you a shortcut to disable auto-answering intercom calls whenever you do not want to be interrupted. The color of the LED reflects the current auto-answer on or off state as a visual reminder to you. If `AutoAnswerEnable` is turned off, the intercom call is presented to you as a normal incoming call and rings the phone normally.

Depending on the method selected, for outgoing calls, your phone may request the called party to auto-answer by inserting either “answer-after=0” parameter in a Call-Info header or “info” and “delay” parameters in an Alert-Info header in the outbound INVITE request. This behavior can be invoked by dialing a star code that invokes the barge-in (or bar for short) action (\*96 by default) before the call and it only applies to the next immediate outgoing call. However, the softswitch may or may not recognize this parameter or pass it through to the called phone.

A softswitch may have its own special way of letting a phone user invoke auto-answer feature on the called party. For example, FreeSwitch uses the feature code `8+{extension}` to signal auto-answer. For example, you can dial 81002 to request extension 1002 to auto-answer the call.

## ***Auto-Answer Incoming Call Based on Inbound Call Routing Rules***

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

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

The `autoans` syntax supports two optional semi-colon separated attributes:

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

## ***Auto-Answer when Caller Requests to Barge-In (with OBiTALK Service)***

In this scenario, the caller requests the called phone to auto-answer in order to join (or barge-in) an active call currently on the called phone. Many softswitches offer you this feature by performing the audio mixing at the server end without the phone knowing anything about the operation.

With its proprietary signaling, your phone offers the Barge-In feature at the device end with the OBiTALK service (that is available to OBi-to-OBi calls only). The calling phone requests to barge-in when you dial a star code to enable the star code variable `$barge-in1` before dialing the target OBi number. The following star code setup example uses `*98` for this purpose:

```
*98, Barge In, set($barge-in1,1)
```

The called phone automatically answers and conferences in the calling phone, subject to resources available at that moment.



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

## **Barge-In as a Coach**

The calling phone to barge in may in addition request to coach the called phone instead of fully participate in the active call. The calling peer that barges in is known as the Coach, while the called peer is known as the Coachee. The original peer at the other end of the active call that the coachee has been talking to is referred to as the Patron. Precisely, coaching means: The coach hears a mix of the coachee and patron speeches, the coachee hears a mix of the coach and patron speeches, but the patron hears only the coachee. To request to coach the called peer, the coach dials a star code to enable the star code variable `$coach1` before dialing the coachee OBi number. The following star code setup example uses `*99` for this purpose:

```
*99, Barge In, set($coach1,1)
```



This feature is also known as ‘Whispering’ in some literature.

## ***Push To Talk***

The phone supports push-to-talk mode with the feature key function Speed Dial, Busy Lamp Field, and Page Groups 1 through 10. See the corresponding feature key section on how to enable the PTT mode with each function.

## Speed Dial

### Speed Dial Feature Key

The phone administrator may allocate one or more feature keys on the phone to be used as Speed Dials. To find out which feature keys are set up as Speed Dials, you can:

- For a VLK, check the function icon on the respective VLKW on the display
- For any feature key, press and hold down the key until the respective feature key item is shown on the display. Then check the function icon of the feature key item.

Note that only the administrator can designate the function for each feature key. The user is not able to change the function of a feature key as assigned by the administrator. However, you can configure the `Number` and `Service` parameter of a speed dial feature key from the phone. You can press and hold down the key until the corresponding feature key item is shown on the display, then view or change the target number and the service (or line) to use when calling from that speed dial key. To call from a speed dial key, press and release the feature key normally.

- To enable the PTT mode on a speed dial, the administrator must include the `ptt` flag in the `Number` parameter of the speed dial.

The general syntax is **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:

- 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 more digits manually on the phone with the given {prompt} shown on the screen. You then press the **OK** softkey to continue
- A – Wait for the called party to answer before continuing



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`

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

## Speed Dial Numbers

Each phone supports 99 speed dial numbers. The 99 speed dial slots are numbered from 1 to 99 and are invoked by dialing a 1 or 2-digit number corresponding to the slot number. Speed dials may be dialed from the phone or via the Auto Attendant.

The 2-digit numbers “01”, “02”, ..., “09” are not valid. You must dial the 1-digit number “1”, “2”, ..., “9” for slot number 1 through 9.

Speed dial values can be set using the configuration web page, remote provisioning, or star code (see the [Controlling Calls Using Star Codes](#) section in this document for more details). The value may be a number just like the one you normally dial, with or without any service access code prefix, such as: \*\*9200112233, \*\*214089991123, 4280913, and so forth. It may also include explicit trunk information with the general format TK(number), where TK= SPn (n = 1 - 6), or PP. For example, PP(ob200112233), SP2(14089991123), and so forth.

If trunk information is not specified in the speed dial entry, your phone applies **DigitMap** and **OutboundCallRoute** when making the call. Otherwise, neither **DigitMap** nor **OutboundCallRoute** is applied.

## Block Caller ID

Block Caller ID (or Anonymous Call) allows you, when making an outgoing call, to prevent your name and number information from appearing on the called phone. This is a feature that can be enabled for all calls on the phone or only for calls on a specific service. For calls on an SP service, the feature may be offered locally by the phone, or by the softswitch serving that SP service on the phone. For SP1 – SP6, the parameters that are related to this service are (where the X in ITSP Profile X is the value of **ServProvProfile** of the line):

### Blocking Caller ID

Parameter Group	Parameter	Description
<i>ITSP Profile X – SIP</i>	<b>X_InsertRemotePartyID</b>	If enabled, the phone includes Remote-Party-ID header in the INVITE. This header contains similar caller information as an unmasked FROM header. In case the FROM header is masked, the user information in this header may be used for authentication or accounting purpose.
<i>ITSP Profile X – SIP</i>	<b>X_InsertPrivacyHdr</b>	If enabled, the phone includes a Privacy: id header when anonymous call is enabled.
<i>ITSP Profile X – SIP</i>	<b>X_UserAnonymousFrom</b>	If enabled, the FROM header is totally masked.
<i>SPn Service – Calling Features</i>	<b>AnonymousCallEnable</b>	Enable Anonymous Call feature on this line.
<i>SPn Service – Network Provided Services</i>	<b>AnonymousCall</b>	Anonymous Call service is provided by the softswitch. The UI sets and gets the setting at the service provider instead.
<i>Phone Settings</i>	<b>AnonymousCall</b>	Enabling this option is equivalent to enabling the feature for all services.



This service requires ITSP support.

If the feature is offered by the phone for the SIP/SP service, it does the following to the outgoing INVITE:

```
From: "Anonymous" <sip:12345@abc.com>;tag=spx13040-5345425
From: "Anonymous" <sip:anonymous@localhost>;tag=spx13040-5345425
Privacy: id
```

If the feature is offered by the softswitch, the phone does nothing but to make the option available to view and set by you from the UI.

The feature can be made available in several ways by a combination of phone and softswitch. When the feature is available, the administrator lets you enable or disable the feature from the phone using one of the following methods:

- Using a star code: To use Caller ID Block for one call only, dial \*67 and then the destination number. To use Caller ID Block on a persistent basis, dial \*81 from the phone. After that, all calls use the Caller ID Block feature until you cancel the Caller ID Block. To cancel Caller ID Block, dial \*82 from the phone.
- Using a feature key: Administrator defines a feature key with the function **Block Caller ID**.

## ***Block Anonymous Call***

**Block Anonymous Call** allows you to block incoming calls that has have no identifying caller ID number. Incoming calls are presented with a busy signal or busy call treatment (such as **Call Forward On Busy**). The administrator can make this feature available to end-user configurations with one of the following methods:

- Star code: To use **Block Anonymous Call**, dial \*77 from the phone. To cancel **Block Anonymous Call**, dial \*87.
- Feature key with the **Block Anonymous Call** function.

This feature has both phone versions and line versions.

## ***Calling Line ID Display***

When making an outgoing call, your phone accepts SIP UPDATE method from the softswitch that may contain the called party's name and number. Your phone then updates the phone screen with the called party information from the SIP UPDATE request. There are no configuration parameters for this feature.

Your phone can also identify the called party ID from the 18x and 2xx responses to the outbound INVITE message.

## ***Call Forwarding***

**Call Forwarding** allows you to send incoming calls to another number of your choosing. Calls can be forwarded to a number reachable from the landline service, VoIP service, or OBITALK network. Three types of call forwarding are supported:

- Call Forward Unconditional (same as Call Forward All)

- Call Forward On Busy
- Call Forward On No Answer

There is one set of Call Forward settings per line and 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.

## Call Forward Numbers

Calls may be forwarded to numbers on the same service or on another service. Therefore each call forward number stored in your phone's configuration **MUST** include call routing information to let the device know which voice service should be used to forward the call to. The general format of a call forward number is:

TK(number)

where TK is the abbreviated name of a voice service. Valid values of TK are SP $n$  for the SP $n$  Voice Service where  $n = 1 - 6$ , or PP for the OBiTALK Service.

The number to forward to must be in the final form that is acceptable by the service provider. Your phone won't apply any Digit Map or Call Routing Rules on it.

Examples: SP1(14089991234) , PP1(ob200333456)

## Call Forward on No Answer

When you use **Call Forward on No Answer**, all calls are forwarded to the number you indicate only when you do not answer the call. To enable **Call Forward on No Answer**, dial \*62. Your phone prompts you to enter the number to which the calls are forwarded. Dial the number plus the # key and a confirmation tone is played. To disable **Call Forward on No Answer**, dial \*63. A confirmation tone is played.

### Call Forward on No Answer

Parameter Group	Parameter	Description
<i>ITSP Profile X – SIP</i>	X_Use302ForCallFoward	
<i>SPn Service – Calling Features</i>	CallForwardUnconditionalEnable	Enable Call Forward Unconditional feature on this line.
<i>SPn Service – Calling Features</i>	CallForwardUnconditionalNumber	Call Forward Unconditional Number. Must be in full number format.
<i>SPn Service – Calling Features</i>	CallForwardOnBusyEnable	Enable Call Forward On Busy feature on this line.
<i>SPn Service – Calling Features</i>	CallForwardOnBusyNumber	Call Forward On Busy Number. Must be in full number format.
<i>SPn Service – Calling Features</i>	CallForwardOnNoAnswerEnable	Enable Call Forward On No Answer feature on this line.
<i>SPn Service – Calling Features</i>	CallForwardOnNoAnswerNumber	Call Forward On No Answer Number. Must be in full number format.
<i>SPn Service – Network Provided Services</i>	CallForwardOnNoAnswerRingCount	



## Call Forward on No Answer (continued)

Parameter Group	Parameter	Description
<i>SPn Service – Network Provided Services</i>	CallForwardUnconditional	
<i>SPn Service – Network Provided Services</i>	CallForwardOnBusy	
<i>SPn Service – Network Provided Services</i>	CallForwardOnNoAnswer	

## Call Forward Signaling

If the called service is on  $SP_m$  and the call forward target is on  $SP_n$  where  $SP_m$  and  $SP_n$  point to the same ITSP Profile X (including the case  $m = n$ ) and *ITSP Profile X – General::SignalingProtocol* is SIP and *ITSP Profile X – SIP::X\_Use302ForCallForward* is true, 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. For other cases, the phone tries to bridge the caller and forward to the target internally.

## Limitations of Call Forward 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 would be acceptable by the other peer, the administrator should understand this limitation and configure the codec profiles accordingly. For example, if SIP service 1 is used, then a call transfer involving a SIP service 1 call leg must make sure the other call leg supports the G711U codec.

## Do Not Disturb

**Do Not Disturb (DND)** enables you to set your phone to immediately forward calls made to it to the number set up as your voicemail number or account. If no voicemail account is set up, your phone returns a busy signal to the caller until you disable **DND**.

To turn on **DND** from a phone attached to your phone, dial \*78. To turn off **DND** from your phone, dial \*79.

## Do Not Disturb (DND)

Parameter Group	Parameter	Description
<i>SPn Service – Calling Features n=1–6</i>	DoNotDisturbEnable	Enables <b>Do Not Disturb</b> feature on this line.
<i>SPn Service – Network Provided Features n=1–6</i>	DoNotDisturb	
<i>Phone Settings</i>	DoNotDisturb	
<i>Phone Settings</i>	DoNotDisturbFeatureProvider	

## Do Not Ring

Enabling **Do Not Ring** disables the ringer. This is equivalent to silent mode. The phone screen still indicates when a call is incoming. A feature key can be assigned with the corresponding LED indicating the on or off status of **Do Not Ring**.

## Message Waiting Indication – Visual and Tone Based

Message Waiting Indication notifies you when a new voice message is available. Your phone supports both Visual and Tone based Message Waiting Indication. With Tone-based Message Waiting Indication, you are alerted to a waiting message by a “stutter” dial playing when the phone is taken off-hook. Typically, this stutter tone is removed once you listen to the message(s). Visual-based Message Waiting Indication turns on a light when there is a message waiting.



The MWI LED on your phone supports only the color red. When specifying an LED pattern for MWI, use only the R (red) or X (off) color values.

### Message Waiting Indication

Parameter Group	Parameter	Description
<i>SPn Service – Calling Features n=1–6</i>	<b>MWIEnable</b>	Controls the stutter tone.
<i>SPn Service – Calling Features n=1–6</i>	<b>VMWIEnable</b>	Controls the Message Waiting Light at the top edge of your phone.
<i>SPn Service – Calling Features n=1–6</i>	<b>MessageWaiting</b>	This is the message waiting state. It is configurable so that the administrator can clear the status if necessary. However, the administrator normally should not provision this parameter.
<i>ITSP Profile X – SIP</i>	<b>MWISubscribe</b>	
<i>ITSP Profile X – SIP</i>	<b>MWISubscribeExpires</b>	

MWI accepts an unsolicited NOTIFY for event message-summary. You may subscribe to the message-summary event package and process NOTIFY within the subscription dialog.

## Multicast Paging Groups

Each phone supports two multicast groups that are configurable by the administrator. The administrator may also assign a feature key for users to join or leave each group. The default settings on your phone work in most LAN environments, so usually all that is required is for the feature to be enabled and a feature key mapped to each phone for the service. Push to talk can also be enabled for Page Groups. This can be configured under **Phone Settings – Page Group 1/2/3/4/5/6/7/8/9/10**. Consult your phone administrator if to see if the group paging feature is enabled at your location.

## Multicast Paging Groups

Parameter Group	Parameter	Description
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>GroupName</b>	A user friendly name to label the group on the phone UI.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>MulticastAddress</b>	Must be a valid IPv4 Multicast Address. Default is 224.1.1.100 for both paging groups.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>MulticastPort</b>	Default is 65322 for page group 1 and 65324 for page group 2. Note that each group must use a different port number.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>PushToTalk</b>	A Boolean option to control whether the page group key (softkey or feature key) should act as a PTT key. That is, you can talk only while pressing+holding the key down. Default is <i>false</i> .
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>TTL</b>	TTL value of outgoing (multicast) RTP packets. Default is 2.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>AudioCodec</b>	Audio codec to use for outgoing page. Default is G711U.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>TxPacketSize</b>	Outgoing RTP packetization in milliseconds. Default is 20.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>RTCPTxInterval</b>	Interval in milliseconds between sending outgoing RTCP packets when paging. No RTCP packets sent if value is 0, which is the default. Note that a RTCP Bye packet is always sent when ending an outgoing page regardless of this setting.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>SilenceSuppression</b>	A Boolean option to control if Silence Suppression should be used for outgoing page. Default is <i>false</i> .
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>ParticipantName</b>	A string value to identify the current user who is paging to the group in outgoing RTCP packets.
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>PlayToneOnIncomingPage</b>	A Boolean option to control whether to play a short Paging Tone before playing a new incoming page. Default is <i>true</i> .
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>PreferredAudioDevice</b>	Select the preferred audio device to play incoming page. Enter either <i>System</i> , <i>Headset</i> , or <i>Speaker</i> . Default is <i>System</i> to let the phone picks according to other user preferences
<i>Page Group N</i> <i>N = 1 through 10</i>	<b>PageTimeout</b>	Limit of outbound page duration in number of seconds to avoid accidentally jamming the page group channel. The duration would be unlimited if the value is 0, which is also the default.

## Multicast Paging Groups (continued)

Parameter Group	Parameter	Description
<b>Page Group N</b> <i>N = 1 through 10</i>	<b>EnableJoinLeaveGroupUserControl</b>	A Boolean option to control whether to allow you to Join or Leave a page group by pressing (and releasing it quickly) the pg1 through pg10 softkey. Default is <b>true</b> . When it is set to <b>false</b> , press the key once to talk and press it again to go back to listen. Although in that case you can't choose the leave the page group, you can still use the <b>AutoAnswerIntercom</b> option from the User Preferences menu not to play any of the incoming pages.
<b>Speaker Volume</b>		Initial speaker volume each time the phone starts playing the audio from that page group. If the value is 0, then the current speaker (ringer) volume is used. Default is 0.
<b>Stats</b>		A comma-separated list of 3 running statistics of the number of packets received on this group: <code>rx-pkts-last-hour,rx-pkts-lastminute,rx-pkt-last-second</code> where the values are update once every 10 seconds.

## Music On Hold (MOH)

The phones have native support of Music On Hold (MOH). You may configure a MOH server in the **Phone Settings – Calling Features**:`MOHServiceNumber` parameter. The MOH Server number can be an external SIP or OBITALK addressable device, such as `pp(ob600559558)` or `sp3(moh-server)`. The expected behavior of the MOH Server device is such that when called, it automatically answers the call and starts streaming audio to the calling device.

Your phone has a built-in MOH Server that can be invoked by specifying: `an({prompt})` (where `an` stands for the internal announcement server), and `prompt` is a specification of the source to play. The devices includes a Jazz track that can be used for music on hold. To enable the built-in track, specify the `MOHServiceNumber` as `an(jazz)`

## Premium Call Features

Your OBi Edition phone provides these and other premium call features:

- Busy Lamp Field (BLF)
- Managing call park, hold, and pickup
- Managing Shared Call Appearances (SCA)
- Creating and managing Buddy Lists
- Recording calls







- Providing Advice of Charges (AOC)
- Making emergency calls

## Busy Lamp Field (BLF)

(SIP/SP only.) BLF is a common collaborative feature for a user to monitor the extensions of other users from their phone. BLF is a feature of the monitoring phone, The monitored extension usually does not do anything special to be monitored. The detection and notification of events occurring at the monitored extensions are carried out by the softswitch.

When BLF is supported by a specific SP service on your phone, the BLF function can be assigned to a feature key bound to that service to monitor an extension in the context of that SP service (That is another extension on the same hosted voice service or PABX system). A feature key that is assigned the BLF function is referred to as a BLF Key. One BLF Key monitors one extension only. The administrator can assign as many BLF Keys as there are feature keys available, to monitor as many extensions. The following table lists the typical events that are monitored with a BLF Key. The table also shows the operation that can be invoked when the respective event occurs, and the LED pattern indicating that particular event.

### List of All BLF States

Icon	BLF Event	Description	Available Operation	LED Pattern
	Ringing	Monitored extension is ringing.	Answer the call. This operation is called a Directed Call Pickup.	Fast blinking red
	Busy	Monitored extension is on a call or making a call.	Barge in to do one of the following: <ul style="list-style-type: none"> <li>• Fully participate in the call or conference.</li> <li>• Listen to the conversation without talking.</li> <li>• Listen to the conversation but talk only to the monitored extension (also known as coaching or whispering).</li> </ul> There can be other variations along these lines.	Steady red
	Holding	Monitored extension is holding one or more calls.	Resume and take over the call.	Slow blinking red
	Call Parked	A call is parked against the monitored extension.	Pick up the call from the parking slot.	Periodic short blink in red
	Idle	Monitored extension is idle.	Make or transfer a call to the extension.	Steady green
	Offline	Monitored extension is offline.	—	Steady amber

## BLF Operation: Speed Dial

When used as a speed dial, your phone determines the number to call the monitored extension according to the following logic, based on attributes specified in the `Number` parameter of the BLF key:

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

## BLF Operation: Directed Call Pickup

Directed Call Pickup can be done in one of two ways: a) Feature Code or b) INVITE+Replaces. The method to use is configured under the ITSP Profile of the SP Service that is bound to the BLF key.

For the Feature Code method, the Feature Code to use for Directed Call Pickup is also configured under the same ITSP Profile; 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 else the `{userid}` attribute). (BroadSoft, FreePBX)

For the INVITE+Replaces method, your phone sends an INVITE with a Replaces header that identifies to the softswitch the ringing call to pick up from the monitored extension. (MetaSwitch)

## BLF Operation: Barge In

When the monitored extension is on a connected call, the monitoring phone may barge in to join the call, if the softswitch supports the operation and if the feature is enabled in the phone configuration. This operation requires the specification of a barge-in feature code that can be configured under the ITSP Profile of the SP service that is bound to the BLF key. To barge in, your phone sends a normal INVITE to the number formed by concatenating the barge-in feature code with the number of the monitored extension (the `ext` attribute of the `Number` parameter, if exists or else the `{userid}` attribute). (BroadSoft, FreePBX)

## BLF Operation: Call Pickup

When a call is parked against the monitored extension, the monitoring phone can pick up the parked call by sending a normal INVITE to the number formed by concatenating a call pickup feature code with the number of the monitored extension (the `ext` attribute of the `Number` parameter, if exists or else the `{userid}` attribute). The call pickup feature code can be configured under the ITSP Profile of the SP service that is bound to the BLF key. (BroadSoft)

## BLF Operation: Resume

The Resume operation is intended to resume (and take over) a holding call on the monitored extension. At the time of writing, there is no known softswitch that supports this operation.

## BLF Configuration

Select a feature key **Key N** to configure the following parameters:

## BLF Configuration

Parameter Group	Parameter	Description
<b>Key N</b> <b>N = 1,2,3...</b>	<b>Function</b>	Must be <i>Busy Lamp Field</i>
<b>Key N</b> <b>N = 1,2,3...</b>	<b>Service</b>	Select the SP service that provides this function and signaling method must be SIP. For example <i>SP1</i>
<b>Key N</b> <b>N = 1,2,3...</b>	<b>Number</b>	<p>General Syntax:</p> <pre> [[group]/][{userid}][?][;ext={extension}] [;ptt][;spd={speed-dial}] </pre> <p>where</p> <ul style="list-style-type: none"> <li>• <i>{userid}</i> is the userid of the monitored extension. It may be a generic user name, a DID number, or an internal extension number assigned to that extension.</li> <li>• <i>{extension}</i> is the (callable) extension number assigned to the monitored extension if it is different from the userid of that extension. For example, an extension may have <i>{userid} = ObihaiUser2</i> and <i>{extension}=1002</i>. If the <i>ext</i> field is specified, then its value is used for speed dial, directed call pickup, barge in, and call pickup. Otherwise <i>{userid}</i> is used. In the case of BroadSoft, it is required to use the <i>{extension}</i> instead of the <i>{userid}</i> for such operations in this context.</li> <li>• if the <i>ptt</i> flag is specified, your phone handles the call as a push-to-talk call, when the key press is interpreted as a speed dial to the monitored extension.</li> <li>• <i>{speed-dial}</i> can be used to specified an alternative number to call when pressing the key as a speed dial. This can be useful for adding some prefix digits to force certain behavior of the call. For instance, adding a prefix 8 to the extension tells the called party to auto-answer in FreeSwitch. When combined with the <i>ptt</i> flag, this provides the familiar Intercom user experience.</li> </ul>
<b>Key N</b> <b>N = 1,2,3...</b>	<b>Name</b>	A nickname or user-friendly name for the monitored extension. It can be used by the phone in some UI elements.
<b>Key N</b> <b>N = 1,2,3...</b>	<b>Group</b>	Not used.
<b>ITSP Profile X – SIP – Feature Configuration</b> <b>X = A-F</b>	<b>X_BLFSubscribeExpires</b>	BLF implementation relies on the (SIP) subscription to the dialog event package of the monitored resource. This value specifies in seconds the expires value of the subscription. The phone renews the subscription well before the nominal expiration time.

## BLF Configuration (continued)

Parameter Group	Parameter	Description
<b>ITSP Profile X – SIP – Feature Configuration</b> <i>X = A-F</i>	<b>X_DirectedCallPickupMethod</b>	Specifies the method to use for directed call pickup (answer the call that is ringing on the monitored extension). The Choice of: <ul style="list-style-type: none"> <li><b>Feature Code:</b> Phone dials the number that is formed by prefixing the number of the monitored extension with Directed Call Pickup Feature Code. This method is used by BroadSoft, FreePBX, and FreeSwitch.</li> <li><b>INVITE+Replaces:</b> Phone sends an INVITE with a Replaces header that specifies to the server the call that it wishes to take over. This method is used by MetaSwitch.</li> </ul>
<b>ITSP Profile X – SIP – Feature Codes</b> <i>x = A-F</i>	<b>DirectedCallPickup</b>	Feature Code for directed call pickup.
<b>ITSP Profile X – SIP – Feature Codes</b> <i>x = A-F</i>	<b>CallPickup</b>	Feature Code for call pickup.
<b>ITSP Profile X – SIP – Feature Codes</b> <i>X = A-F</i>	<b>BargeIn</b>	Feature Code for barge in.
<b>SPn Service – BLF Features</b> <i>n = 1-6</i>	<b>X_BlfBargeIn</b>	Enable Barge In option for BLF keys associated with this SPn service.
<b>SPn Service – BLF Features</b> <i>n = 1-6</i>	<b>X_BlfCallPickup</b>	Enable Call Pickup option for BLF keys associated with this SPn service.
<b>SPn Service – BLF Features</b> <i>n=1-6</i>	<b>X_BlfDirectedCallPickup</b>	Enable Directed Call Pickup option for BLF keys associated with this SPn service.

## Floating BLF Key Assignment

It is possible to reserve a block of BLF keys to members of a group of extensions without hard-coding the extension for each reserved key. The idea is to SUBSCRIBE to a single resource that includes a list of extensions to monitor, instead of subscribing to each individual extension. Upon receiving a NOTIFY from the server with the list of extensions, your phone assigns each extension to a BLF key reserved to the subscribed group using an internal algorithm. To reserve a BLF key to a member of the group, specify the Number parameter of a BLF key as: `Number = {group-name}/`

For example:

```
Number = sales-team/
```



The slash (/) at the end indicates the key is reserved to a member of the group named `sales-team`. You may also specify a preferred extension to use that key with the syntax: `Number = {group-name}/{preferred-extension}?`

For example:

```
Number = sales-team/1003?
```

The question-mark (?) at the end indicates that 1003 is merely a suggestion and monitors the extension only if it is present in the list of extensions returned by the server in a NOTIFY message body. If 1003 is not in the extension list from the server, your phone is free to assign this reserved BLF to another key when it is running out of free keys to assign. So, if all the BLF keys are floating without specifying they are a suggestion, the order of extension assignment follows precisely the order of the extensions in the list received from the server.

## SIP for BLF

For each BLF extension that does not belong to any group, your phone subscribes to the dialog event package for each extension in the context of the SP service specified in the `Service` parameter of the corresponding feature key. For extensions belonging to the same group, your phone only maintains one subscription to the group-name and none for the individual extensions in the group. In the subscribe request, your phone indicates (in an Accept header) support for the following content-types in NOTIFY message bodies to be returned by the server:

- `application/multipart/related`
- `application/rfmi+xml`
- `application/dialog-info+xml`

For floating key assignment, your phone expects the NOTIFY message body to include a resource list (Content-Type: `application/rfmi+xml`) that is compatible with RFC4662. Here is an example with two extensions specified in the resource list:

```
<?xml version="1.0" encoding="UTF-8"?>
<list xmlns="urn:ietf:params:xml:ns:rfmi" uri="sip:sales-team@as.broadworks.net"
  version="0" fullState="true">
  <resource uri="sip:ObihaiUser2@as.broadworks.net">
    <name>Obihai User2 </name>
    <instance id="DNAbROacM9" state="active" cid="7jTC13@broadworks"/>
  </resource>
  <resource uri="sip:ObihaiUser9053@as.broadworks.net">
    <name>Test User</name>
    <instance id="MT8FRckGPc" state="active" cid="cJ489p@broadworks"/>
  </resource>
</list>
```

To notify the call states of a monitored extension, your phone expects the NOTIFY message body to include a dialog-info XML (Content-Type: `dialog-info+xml`) that is compatible with RFC4235. Here is an example:

```
<?xml version="1.0" encoding="UTF-8"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="1"
  state="full" entity="sip:ObihaiUser2@as.iopl.broadworks.net">
  <dialog id="Y2FsbGhhbGYtNjI4MjM4NzU6MA==" direction="recipient">
    <state>proceeding</state>
  </dialog>
</local>
```

```

    <identity display="ObihaiUser2 ObihaiUser2">
      sip:ObihaiUser2@as.iopl.broadworks.net
    </identity>
    <identity display="ObihaiUser2 ObihaiUser2">
      tel:+12404982562;ext=2562
    </identity>
  </local>
  <remote>
    <identity display="ObihaiUser1 ObihaiUser1">
      sip:2561@as.iopl.broadworks.net;user=phone
    </identity>
  </remote>
</dialog></dialog-info>

```

When the softswitch is capable of notifying call park status for the monitored extension, it is expected the status is reported inside a dialog-info XML as well. Here is an example:

```

<?xml version="1.0" encoding="UTF-8"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
  xmlns:sa="urn:ietf:params:xml:ns:sa-dialog-info"
  xmlns:bw="http://schema.broadsoft.com/callpark"
  version="3" state="partial"
  entity="sip:north00@txasdev87.net">
  <dialog id="Y2FsbGhhbGYtMzM6MA==">
    <state>confirmed</state>
    <bw:callpark>
      <bw:parked>
        <identity display="Alice south">
          sip:9726987601@as.bw.com;user=phone
        </identity>
      </bw:parked>
    </bw:callpark>
  </dialog>
</dialog-info>

```

## BLF With OBiTALK Service

While most BLF implementation requires the softswitch to act as the state aggregator, OBiTALK service offers a peer-to-peer version of BLF that can be quite handy. Every OBi Edition phone can notify its phone status to a configured OBi number configured in the **OBiTALK Service – Calling**

**Features::**CallStatusAggregator parameter. The notified phone monitors each notifying phone with a BLF function key that is configured with the Voice Service field set to OBiTALK and number set to the monitored 9-digit OBi number.

The following phone status can be monitored:

- Offline
- Idle (no calls)
- Ringing (at least 1 call ringing)
- Holding (at least 1 call holding)
- Connected (at 1 call connected)

On the monitoring phone, the following operations can be applied to the monitored OBi number:

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



The call on the monitored phone to be picked up, barged in, or coached does not need to be on OBiTALK service. If the call is not on OBiTALK service, the monitored phone bridges the call with the OBiTALK call leg with the monitoring phone that initiates the operation.

## Call Park and Call Pickup

Call Park and Call Pickup (or Call Retrieval) are complementary operations not unlike call hold and call resume, except the pickup (vs. resume) operation can be carried out on extensions other than the one that parks (vs. holds) the call. Generally speaking, a “parking lot” for calls with many slots can be setup at the softswitch such that a call may be parked at an unoccupied slot from one extension and picked up by the same or another extension from the same slot. In some implementations the parking slot is referred to as a park orbit.

### Call Park Methods

Your phone supports both the types of call park described above. The method to use can be set independently for each SP service, under the ITSP Profile bound to that SP service, with the **ITSP Profile–SIP::X\_CallParkMethod** parameter. The `Feature Code` method must be used for the Park-Against-An-Extension method, while the `REFER` must be used for the Park-In-An-Orbit method.

With the `Feature Code` method, the call park feature code must be specified under the same ITSP Profile using the **ITSP Profile–SIP-Feature Codes::Park** parameter. When user presses the “Park” softkey for a highlighted call on screen, your phone parks the call (in Holding or Connected state) against the local extension of the underlying SP service by sending a normal INVITE to the number formed by concatenating the call park feature code with the extension number of the local extension. This INVITE (to park) is sent on the same SP service that the call to be parked is on. Note that in the context of parking a call, the local extension of the SP service is taken from the **SPn Service::X\_MyExtension** parameter, if it is specified. Otherwise, the normal userid of the SP service is used. The phone UI does not have a softkey to park a call against an arbitrary extension. To do that, you must make another call by dialing the call park feature code followed by the extension to park against. Your burden can be alleviated by defining a number of speed dials with the feature code+extension preconfigured so you can press the speed dial key to park the call in one of the predefined extensions.

With the `REFER` method, your phone parks the call in an orbit by SIP-referring the peer of the call to be parked to the number that is the orbit to be parked at. This operation is equivalent to that of a blind transfer to a special extension. When you press the **Park** softkey of a highlighted call on screen, your phone pops up a dialog box to ask you to enter the orbit number to park the call at. Alternatively, you can press the Call Park Monitor feature key of an unoccupied park orbit to park the call with a single key press. The Call Park Monitor feature key is described in the next section.



The **Park** softkey won't be seen unless the **SPn Service–Network Provided Services::CallPark** parameter is enabled.

## Call Park Monitor and Call Pickup Methods

Generally speaking, when a call is parked-against-an-extension with a feature code, you can pick it up by dialing the call pickup feature code followed by the extension number the call is parked against. To the phone this is just an ordinary call and it does not need to know, or interpret, the call pickup feature code. The user has the full responsibility of dialing the feature code and the parked-against extension explicitly and correctly. Similarly when a call is parked-in-an-orbit and if the softswitch has a way to allow calls to be picked up from an orbit that is equivalent to making a call to a special number, users can make those calls without any special support from the phone. There are scenarios when your phone can monitor if certain extensions have calls parked against them, or if certain park orbits are holding calls.




For calls parked-against-an-extension, your phone may monitor call park status by directly (SIP) subscribe to an extension's call park status (by enabling the **SPn Service–Network Provided Services::CallParkStatus** option), or indirectly through BLF monitoring of that extension (see the section Busy Lamp Field). Note that your phone supports direct subscription to call park status (in the context of an SP service) of the underlying SP's own extension only, but not an arbitrary extension. Note also that if the underlying SP is a shared line, then the direct subscription to call park status of that extension is not necessary if the softswitch already includes call park status in the notifications of the shared line status to the phone. Your phone supports call park status subscription by subscribing to the x-broadworks-callpark event package, and expects notification that includes an x-broadworks-callpark-info XML document in the message boxy (Content-Type:application/x-broadworks-callpark-info+xml). Here is an example:

```
<?xml version="1.0" encoding="UTF-8"?>
<x-broadworks-callpark-info xmlns="http://schema.broadsoft.com/callpark">
  <callpark>
    <parked>
      <identity display="Alice south">
        sip:876601@as.bw.com;user=phone
      </identity>
    </parked>
  </callpark>
</x-broadworks-callpark-info>
```

With the park-against-an-extension method, you can define a group of extensions to serve as a more generic parking lot, and use BLF keys to monitor the call park status of each key. With the `spd` attribute of the `Number` parameter, these BLF keys include the call park feature code as a prefix to act as a short-cut to park a call in an empty slot.

For calls parked-in-an-orbit, the administrator can define a feature key with the function Call Park Monitor to monitor each orbit (one feature key per orbit; each feature monitors a different orbit number). Your phone treats the Call Park Monitor function always exactly like the BLF function; it subscribes to the dialog event package for each key (or for a group of keys using the same group syntax in the `Number` parameter of the keys). It is however expected that the softswitch either notifies no calls or one call in Ringing state per orbit, as summarized in the following table.

## Call Parking

Call Park Monitor Status	Description	Available Operation	Icon	LED Pattern
One Call Ringing	A call is parked in this orbit	Pickup the call. This operation is done via Directed Call Pickup		Fast blinking red
No Calls	No call parked in this orbit	Park the call on this orbit		Off
Offline	Status of the park orbit is unknown	—		Steady amber

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

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

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

## Call Park Ring

When there is one or more Call Park Monitor keys defined, the Call Park Ring can be enabled to play a short ring at regular interval as an indication that at least one of the monitored call park orbits has a call parked.

This feature can be enabled using the boolean parameter **Phone Settings – Calling**

**Features::CallParkRingEnable**, which is `false` by default. There are two internal short ring tones that can be chosen for this ring: `ding` or `blorp`. Which short ring to use, as well as the ring interval and total ring duration can be set in the **User Preferences::CallParkRingtone** parameter, which takes a string value of the following general format:

```
{ring-tone-url},duration={d-value},interval={i-value}
```

where:

- {ring-tone-url} can be `ding` or `blorp` or an external URL that starts with `http://` or `https://`. For best results, the ring tone for this should be about 1 to 2 seconds long.
- {d-value} is the total duration in seconds to play the ring for, as long as there is a parked call. It must be greater than 0 or equal to 0 to mean play forever. The duration parameter is optional; default value is 60.
- {i-value} is the interval between the ring bursts in seconds. It must be greater than 0. The interval parameter is optional; default value is 10.

To let phone user change the `CallParkRingEnable` option from the User Preferences UI on the phone, add the `cpr` option in any one of the **GUI Menus::PreferencesMenu<sub>n</sub>** ( $n = 1 - 6$ ). The default on-screen label for this option is Call Park Ring and the user can set the value to either `Enable` or `Disable`.



- The call park ring won't start in any of the following situations:
  - ▲ Phone is ringing
  - ▲ Phone is off-hook, speakerphone is on, or headset is on
- Regardless of the duration setting, the call park ring stops when any of the following happens:
  - ▲ No more calls are parked in any of the monitored parking orbits
  - ▲ Phone starts ringing
  - ▲ Phone goes off-hook, or the speakerphone or headset is turned on
- If a new call is parked while the call park ring is playing, the duration timer restarts so it can potentially ring for a full duration for the newly added call.

Tips: When call park ring is playing, an easy way to stop it (until a new call is parked) is by taking the phone off-hook or turning on the speakerphone or headset momentarily.

## Call Hold Audio Notification Tone

This feature enables setting a periodic audio notification tone to tell you when you have another call on hold. Set the following parameter value to enable this notification:

- **IP Phone > Phone Settings > Calling Features::CallHoldRingEnable = True**

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

## Shared Line and Shared Call Appearances (SCA)

(SIP/SP only.) A Shared Line is a service account or extension that is installed on a group of phones, such that if a sharing phone is using that extension, other sharing phones are notified. There can be multiple simultaneous calls on a shared line. The maximum allowed simultaneous calls on a shared line should be a fix number. Each call on a shared line is called a Shared Call Appearance (SCA). If your phone has a shared line configured as one of the services, it should have as many call keys defined on the phones that are bound to that service. SCAs of a shared line are ordered with an index 1– $n$  as SCA1, SCA2, SCA3, SCA $n$ , where  $n$  is the maximum number of calls permitted for that line. On your phone, the SCAs are assigned sequentially in ascending order according to the VLK index of the corresponding call key bound to the shared line. For example, suppose a shared line with four SCAs is configured on a phone with VLK1, VLK3, VLK7, and VLK8 assigned as the four call keys bound to that line, then the calls hosted on VLK1, VLK3, VLK7, and VLK8 are SCA1, SCA2, SCA3, and SCA4, respectively.

There are two common implementations of SCA, namely, the Call-Info Method and the SLA/BLA (dialog) method. Use the **ITSP Profile–SIP–Feature Configuration::X\_ShareLineMethod** parameter to select which method to use. It can be one of the following choices:










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

- `dialog;ma`: This is similar to the `dialog;sla` method, and is based on a more current version of the same draft (draft-anil-sipping-bla-04). The name of the event package name is changed to `dialog;ma`.

To designate an SP service as a shared line, enable the option **SPn Service–Share line**

**Features::X\_ShareLine**. The **SPn Service–Calling Features::MaxSessions** parameter should be set the value of the maximum number of call appearances allowed on the shared line. You must also have the same number Call Appearance feature keys defined to bind to this SP service. A Call Appearance key bound to a shared line behaves similarly to one that is bound to a private line, except that when the (private) call state is idle (that is, the call appearance is not being used by this phone), the SCA state is shown. The SCA state (or shared call state) indicates the call state on the sharing phone that is using that call appearance, and is communicated to this phone by softswitch via (SIP) notification. The following SCA states are supported by the phone:

#### List of SCA States

Icon	SCA State	Description	Available Operation	LED Pattern
	Seized	A sharing phone has seized the SCA to make an outgoing call.	—	Blinking red (30 ms on, 30 ms off)
	Trying	A sharing phone is trying an outgoing call.	—	Steady red
	Proceeding	A sharing phone is making an outgoing call and the called party is ringing.	—	Steady red
	Connected	A sharing phone is on a connected call with that SCA.	Barge In to: <ul style="list-style-type: none"> <li>• Monitor the call only</li> <li>• Fully participate in the call</li> </ul>	Steady red
	Held	A sharing phone is on a call with that SCA and has placed the call on hold.	Resume and take over the call.	Slow blinking red
	Private Held	A sharing phone is on a call with that SCA and has placed the call on private hold.	—	Slow blinking red (100 ms on, 100 ms off)
	Call Parked	A call is parked against the extension.	Pick up the parked call.	Blinking red (50 ms on, 50 ms off)
	Idle	None of the sharing phones is using the SCA.	Seize the SCA to make an outgoing call.	Off
	Error	Protocol error.	—	Steady amber
	Remote Hold	Remote side placed the call on hold.	Barge In to: <ul style="list-style-type: none"> <li>• Monitor the call only</li> <li>• Fully participate in the call</li> </ul>	Steady red

SCA implementations are based on subscribe/notify framework. With the call-info method, the subscription expires value to the call-info and line-seize events are set in `X_CallInfoSubscribesExpires` and `X_LineSeizeSubscribeExpires` (under **ITSP Profile X – SIP**), respectively. With the `dialog;ma` and `dialog;sla` methods, the subscription expires value to the respective event is controlled by the `X_DialogSubscribesExpires` parameter.

## Line Seize

With any Shared line design, each sharing phone should perform a proper line-seize before attempting to make a call on a SCA. This lets the system work correctly when multiple phones are trying to make calls at the same time.

With the call-info method, a phone seizes a SCA by subscribing to the line-seize event package for that SCA, with a short expires value. If successful, the phone receives a 200 class response for the subscribe method, and also a NOTIFY to indicate that the subscription is active and what the actual expires value of the subscription allowed by the server.

With the `dialog;sla` or the `dialog;ma` method, a phone seizes a SCA by sending a NOTIFY to the state-agent with the state of the requested SCA set to `trying`. If successful, the phone receives a 200 class response for the NOTIFY request. Typically, the state-agent shortens the subscription interval with the phone that owns the SCA to detect quickly if the phone has crashed or encountered network issues.

## Buddy List

A Buddy List is a contact list with presence information incorporated. Your phone supports Buddy Lists based on the XMPP standard. This feature is enabled with the option **SPn Service – Network Provided Services::BuddyList**. Note that the service can be enabled for each SP service independently. To configure your buddy list service, the following additional parameters are required:

### Buddy List Parameters

Parameter Group	Parameter	Description
<b>SPn Service – SIP Credentials</b>	<b>X_XmppDomain</b>	The XMPP server domain name. The phone resolves this name as DNS A Record only. For example: <code>impliop1.broadsoft.com</code>
<b>SPn Service – SIP Credentials</b>	<b>X_XmppUserName</b>	The user name for authentication to the XMPP server. For example: <code>ObihaiTester@impliop1.broadsoft.com</code>
<b>SPn Service – SIP Credentials</b>	<b>X_XmppPassword</b>	The password for authentication to the XMPP server

For the Buddy List App to show, it must be enabled for a service under **SPn Service – Network Provided Services::BuddyList** to display contact images, ensure you also enable the `QueryVcard` option further down the page. The value of the user selectable presence is stored in the **SPn Service::MyPresence** parameter.

The user selected status value is stored in the **SPn Service::MyStatus** parameter. You can add new status values or remove old ones using the **New** and **Remove** softkey. All the status values are available to all SP



services and are stored as a comma-separated list of phrases in the **PhoneSettings::MyStatusHistory** parameter.

## Presence Monitor




You can configure a feature key with the function `Presence Monitor` to monitor the presence of a buddy in a buddy list. Each presence monitor key monitors exactly one buddy. You can configure as many Presence Monitor keys as you need, but do not allocate the same buddy to more than one key. The following parameters are required for configuring a presence monitor key:

- `Function` = `Presence Monitor`
- `Service` = The SP Service where the XMPP service is offered, such as `SP3`
- `Number` = The JID of the buddy to be monitored. This typically looks like an email address, such as: `kkytte@gmail.com`. Only the userid portion of the JID is needed. You may specify `. .` at the end such that a partial match of the JID is sufficient to find the buddy in the buddy list. For example, if the full JID of the buddy is: `abcd-12345@gmail.com`, you may specify `abcd. .` for the `Number` field to monitor this buddy, provided there is no other JID in the buddy list that starts with `abcd`.

## Call Recording Controls

(SIP/BroadSoft.) Your phone supports the call recording functions available with a BroadSoft application server, by providing the controls for call recording during a call. This feature can be enabled on a per SP service basis by enabling the option **SPn Service – Network Provided Services::CallRecording**. When the feature is enabled, the phone shows one of the following call recording states in the call items of the Calls App:

### Call Recording Controls

Call Recording State	Description	Available Softkey Options	Icon
<code>off</code>	Recording has been turned off	Rec.Start	
<code>on</code>	Recording has been turned on	Rec.Stop, Rec.Pause	
<code>paused</code>	Recording has been turned on but paused at the moment	Rec.Resume	
<code>na</code>	Feature is not enabled		

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

## Hold and Talk Event Package

(SIP Only.) Your phone supports unsolicited notification of the hold and talk event package in the context of an SP service. Note that these notifications are sent outside of any dialogs and are not dialog specific. When the phone receives a hold event notification, it holds all the connected calls on the underlying service. When the phone receives a talk event notification, it answers all the incoming calls and resumes all the holding calls on the underlying service.

There is no configuration for these features.

## Advice of Charges (AOC)

(SP/SIP Only.) Your phone accepts contents of the Content-Type: application/vnd.etsi.aoc+xml in the message body of an INFO or BYE request, or 2xx response to a BYE request. It parses the charges information and displays them on the screen in the corresponding call item of the Calls App. The final charges information received with a BYE request or 2xx response to a BYE request is displayed in a separate pop up window at the end of the call.

There is no configuration for this feature.

## Emergency Calls

The administrator can define one or more numbers as emergency numbers by adding the prefix EM# to those numbers using the **Phone Settings::DigitMap** and a corresponding rule in **Phone Settings::OutboundCallRoute** to route those calls to a specific voice service to handle the call. The following example defines an emergency number 911 and routes the call to go out from SP1 when the number is dialed:

```
DigitMap = (<EM#>911|other rules ...)
```

```
OutboundCallRoute = {(911):sp1},{(311):sp2}, other rules ...
```

The phone detects that you are calling an emergency number if the number to call has the prefix EM# after applying the phone digit map on the dialed number. It then applies the emergency call treatment to that call for the duration of the call:

- You cannot hold or end the call;. Only the remote party can end it.
- You can start the call with the headset, but you cannot switch to use a headset subsequently after the call is started. You can only switch between the handset and speakerphone.
- You cannot start or resume any other calls.
- You cannot press the **Home** key to get to the Home screen to start another App.
- Call waiting is disabled.
- All the feature keys are disabled.

## Emergency Call Web Hook

This feature adds the capability to trigger a webhook when making an emergency phone call. This feature uses the new `EmergencyCallActionURL` parameter on the **IP Phone > Phone Settings** web page to specify an Action URL that dials the chosen emergency provider.

## Call Diversion History

(SIP Only). The softswitch may keep a history of Diversion headers for each call forward transaction as it tries to ring an extension. A Diversion header is added each time the called extension redirects (or diverts) the call. When an INVITE message arrives at your phone after a series of redirections, it may include a history of all the diversion headers (with the latest one appearing first). Your phone shows the call diversion history in the call item of the Calls App, if available.

There is no configuration for this feature.

## Digit Map Configuration

A digit map can be used to match digits to ensure a complete number is dialed, transform dialed digits, and block numbers from being dialed. Its structure is a series of rules that read from left to right. Each digit map is composed of one or more rules surrounded by parentheses ( ). These parentheses MUST NOT be omitted.

Your phone applies the first rule that matches the format of the dialed number, so it's important you get your rule order correct.

A digit map is a group of rules written in the following fashion:

```
Digit Map = ( rule | rule | rule | rule | rule )
```

A vertical bar | separates each rule. Once again, the digit map must include the parentheses ( ), or your phone won't read the entered text as a digit map.

You can include "white space" within your digit map rule to make it more readable. Your phone ignores the spaces when reading the rules in the digit map. This helps make it easier to read your rules. The following example shows a rule to match phone numbers dialed to the London area code from within the United Kingdom:

```
(020[378]xxxxxxx) is equivalent to (020 [378]xxx xxxx)
```

The syntax used in this example is discussed later.

## Digit Map Elements

The digit map rule serves to match numbers based on the characteristics of the form and content the entered number (or alphanumeric string, for example, when entering a SIP URI on your phone).

A rule is made up of a series of elements. Each element of the rule is matched from left to right in sequence with the entered string of numbers (or characters).

The following tables detail all available elements with which to create a rule:

Any combination of: 0-9 * # + - A-Z a-z Excluding: m M s S x X	<b>Literals</b> It matches digit sequences with exactly the same literals. m, M, s, S, x, X have special meaning in the digit map syntax. Use literals to explicitly match a string.
---	---

**Example:**

To explicitly match the New York phone number 1-212-555-7722 as dialed from within North America (according to the NANP), create the following rule from literals:

12125557722 or 1 212 555 7722 — spaces can be added for clarity.

‘ ‘

**‘Quoted Literals’**

Everything inside a pair of single quotes is treated as a literal except for the single quote ‘ character itself.

**Example:**

To explicitly match the SIP URI `matt@sipservice.com`, include ‘ ‘ to surround the address, because the address includes reserved characters. Write your matching rule as:

```
\matt@sipservice.com'
```

x

**x Wildcard**

The “lowercase x” – x – wildcard matches any digit from 0-9. It is important to note that x is case-sensitive.

**Example:**

If we consider phone numbers within a specific area code, all of the same length, we can write a rule that explicitly matches digits at the start of the number followed by any sequence of digits of a defined, fixed length. When dialing a number in the Dalton, Georgia area code of 706 from within North America, we would dial 1 706 then the following 7 digits of the number, as such we would write our matching rule as follows:

```
1 706 xxx xxxx
```

.

**. Matching Function**

The “full stop” or “dot” - . - matching function matches 0 or more of the previous element, such as x, X or @.

It’s typically used to capture a string of entered numbers or characters of arbitrary length.

**Example:**

The expression `xx.` would catch any number that is entered with 1 or more digits. Although we have used two x’s in our rule, the dot implies 0 or more x, so in the event of `0*x` the rule `xx.` caters for a single digit number being entered. Let’s say we want to match any international numbers dialed to New Zealand (country code 64) but don’t want to define all the possibilities of the NZ number plan within our digit map. In this case we can write one of the following rules:

```
011 64 xx.    Using US international dialing notation
```

```
0011 64 xx.   Using Australian international dialing notation
```

```
00 64 xx.     Using Standardized international dialing notation
```

[ ]	<p><b>Set</b></p> <p>“Square brackets” – [ ] – enclose a set of numerals and/or characters that are used to match a single digit or character.</p> <p>It’s useful to use a set to match specific digits that form part of a number. Alphanumeric and wildcard characters are allowed inside a set, such as [x], [X#], [@#], [a-zA-Zx] .</p>
<p><b>Example:</b></p> <p>Let’s look at specifically matching the numbers 1,2,5,6,7 and 8. We can write this set as [125-8] where we have specified the digits 1 and 2, then written the numbers 5,6,7,8 as the sequence 5-8. To put this into context, let’s look at the London number range where each number can take the form of 020 3xxx xxxx, 020 7xxx xxxx, or 020 8xxx xxxx – while we could write each of these as individual rules, it is tidier to represent them as follows:</p> <p>020 [378]xxx xxxx</p>	

[^ ]	<p><b>Exclusion Set</b></p> <p>An exclusion set includes a leading caret, or “hat”, within a set of “square brackets” - [^ ] .</p> <p>An exclusion set matches any single alphanumeric character that is not within the set.</p>
<p><b>Example:</b></p> <p>To match any arbitrarily long sequence of digits that does not start with * , write your matching rule as follows:</p> <p>[^*]xx .</p> <p>In the case of Sydney, Australia, local numbers are 8 digits long, starting with any digit between 3 and 9. We could write this rule as [3-9]xxx xxxx or using an exclusion set we could write:</p> <p>[^0-2]xxx xxxx</p>	

!	<p><b>! Call Bar</b></p> <p>To bar users from calling numbers that match a rule, add an “exclamation mark” - ! - in front of that rule in the digit map. The rule is then referred to as a barring rule.</p>
<p><b>Example:</b></p> <p>To bar all calls to 1900 numbers (regardless of length), use the following rule:</p> <p>!1900xx .</p>	

<b>&lt; elements : literals &gt;</b>	<p><b>Element to Literal Transformation</b></p> <p>An element-to-literal transformation allows substituting the digit sequence matching elements with the given literals. The expression is contained within a set of “pointy brackets” - &lt; &gt; - and elements are separated from literals using a colon:</p>
<p><b>Usage Notes:</b></p> <p>Single quote ‘ ‘ syntax is NOT needed or allowed for the literals in this context. Special characters may be used here as they do not apply in this context either.</p> <p>Elements can be empty, in which case the colon - : - can be omitted. This case is useful for inserting some extra digits in certain part of the dialed digits.</p> <p>The literals part can be empty also, but the colon - : - MUST NOT be omitted. This case is useful for removing part of the dialed digits. Elements and literals MUST NOT both be empty.</p> <p><b>Example:</b></p> <p>Use this rule to remove digits from a dialed number, add digits to a dialed number, or transform a dialed number. To transform one number into another, for example to transform the entered digits of 112 to send out 000, write:</p> <p>&lt;112:000&gt; Take 112 and replace it with 000 .</p> <p>Next, strip off preceding digits in a number. In this case, if you dial 02 xxxx xxxx, this rule sends on just the xxxx xxxx without the 02:</p> <p>&lt;02:&gt;xxxx xxxx Take 02, replace it with nothing, then match the next 8 digits.</p> <p>To add to the start of a dialed number, for example to add 02 to the start of an 8-digit number, write:</p> <p>&lt;02&gt;xxxx xxxx or &lt;:02&gt;xxxx xxxx</p>	

<b>S, S0, S1 ... S9</b>	<p><b>Digit Timer</b></p> <p>The digit timer should only be used either as the first element of a rule (for a hot or warm line implementation) or as the last element of a rule as a means of overriding the default inter-digit timer. The digit timer – S – is case-sensitive.</p>
<p><b>Example:</b></p> <p>The notation S0, S1, S2, S9 gives digit timer values of 0, 1, 2 and 9 seconds respectively. S is equivalent to S1 (1 second). S0 is the same as “blank”. You can concatenate multiple S elements together if you need more than 9 seconds timeout, such as S9S5 for a 14-second timeout. To create a hotline to the number 1-408-890-6000 we would write:</p> <p>&lt;S0:14088906000&gt;</p>	

The next two elements, (map) and (Mlabel), imply that the phone digit maps are recursive. Recursive digit maps allow digit maps to be reused and make their specification more compact and readable. It is important that you do not specify digit maps that lead to infinite recursion. For example, a digit map must not include a named embedded digit map that references itself.

(map)	<p><b>Embedded Digit Map</b></p> <p>An embedded digit map contains elements within a pair of brackets – ( ) – that are used to match subsequent digits.</p>
<p><b>Example:</b></p> <p>To match any number that starts with *74, followed by 1 or 2 digits we would write our matching rule as follows:</p> <pre>*74 (x xx)</pre>	

(Mlabel)	<p><b>Named Embedded Digit Map</b></p> <p>A named embedded digit map is used for matching subsequent digits, where label refers to one of abbreviated terminal names or a user-defined digit map label. As an example, (Msp1) refers directly to the digit map contained within <b>Voice Services &gt; SP1 Service</b> of the phone's native web portal.</p>
<p><b>Example:</b></p> <p>Let's consider one of the phone's default digit map rules that directs a call preceded by **2 to SP2. We want to take all the digits after **2 to be matched with SP2's digit map. To do this we write our rule as:</p> <pre>**2 (Msp2)</pre>	

The following advanced elements allow you to further compact your rules for more elegant digit map syntax:

X	<p><b>X Wildcard</b></p> <p>The "uppercase x" – X – wildcard matches any digit from 0-9 as well as * making it useful for use digit maps that include star codes. It is important to note that X is case-sensitive.</p>
<p><b>Usage Notes:</b></p> <p>X is equivalent to [x*] or [0-9*x]</p>	
<p><b>Example:</b></p> <p>To catch all three digit numbers including star codes, write your matching rule as follows:</p> <pre>xxx</pre>	

@	<p><b>@ Wildcard</b></p> <p>The "at symbol" - @ - wildcard matches any alphanumeric character except #.</p>
<p><b>Example:</b></p> <p>To match any arbitrarily long alphanumeric sequence (except #) that does not start with * we would write our matching rule as follows:</p> <pre>[^*]@@.</pre>	

?	<p><b>? Matching Function</b></p> <p>The “question mark” - ? - matching function matches zero or one of the previous element, such as x, X or @.</p> <p>This function can be used to capture a string of entered numbers or characters of multiple known fixed lengths.</p> <p>Example:</p> <p>The expression <code>xxxx?</code> matches any entered number that is either 3 or 4 digits in length. Let's say we live in an area where local numbers vary in length. This example uses the Brompton area in the UK that has 4- and 5-digit local numbers. Rather than using two rules in our digit map such as <code>(... xxxx xxxxx ...)</code> to match the local numbers, you can write a more elegant matching rule as follows:</p> <p><code>xxxxx?</code></p>
---	--

## Digit Map Rule Examples

Here are some further examples of digit map rules:

### Digit Map Rules

Function	Digit Map Rule
Match any 11-digit number starting with 1408.	<code>1408 xxx xxxx</code>
Match any 7-digit number and prepend 1408 to the number when making the call.	<code>&lt;1408&gt; xxx xxxx</code> <code>or &lt;:1408&gt; xxx xxxx</code>
Match any number that starts with 011 followed by one or more digits.	<code>011xx.</code>
Prepend '+' to any 11-digit number that starts with 1.	<code>&lt;+&gt;1xxxxxxxxxxx</code>
Match any number that starts with **1 020 3, 7 or 8 followed by 7 digits and remove the **1 prefix when making the call.	<code>&lt;**1:&gt;020 [378]xxx xxxx</code>
Create a hotline to 1234.	<code>&lt;:1234&gt; or &lt;S0:1234&gt;</code>
Create a warm line to 1234 that is called if the user doesn't enter any digits within 4 seconds.	<code>&lt;S4:1234&gt;</code>
Match any number with at least 8 digits that ends with 8537683, such as 1 510 853 7683, 9 853 7683.	<code>xx.853 7683</code>
Match any number with at least 10 digits that ends in 408-890-6000, such as 1 408 890 6000, 001 408 890 6000, +1 408 890 6000.	<code>@. 408 890 6000</code>
Add a # to the end of any number with 1 or more digits.	<code>xx.&lt;#&gt;</code>
Bar calls to premium rate numbers beginning with 084, 087, and 09.	<code>!08[47]x. and !09x.</code>

Now we can create an example digit map using a few of the rules above. One important function of a digit map is to determine if the user has entered sufficient digits during dialing, given the array of number combinations available, a digit map normally contains more than one rule. To create a digit map that includes



a few of the example rules in the table, we write our digit map enclosed in parentheses ( ) and with each rule separated by a | bar as follows:

```
(<1408> xxx xxxx | @. 408 890 6000 | xx.<#> | !08[47]x. | !09x.)
```

Spaces can be used in the digit map. It is fine to include spaces to help make your digit map more readable.

## Matching Against Multiple Rules in a Digit Map

One important function of a digit map is to determine if you dialed sufficient digits during dialing. A digit map normally contains more than one rule. The Digit Map Processor (DMP) must return the best matched rule at some point, or declare that the input digit sequence is invalid. The DMP keeps refining its decision as each digit is entered until it reaches a final decision, or is forced to make a timely decision when the interdigit timer expires.

The DMP restarts the interdigit timer on every newly entered digit. The duration of this timer can be either long or short. The long and the short timer values are set by default to 10 seconds and 2 seconds, respectively, and are configurable per handset via the **DigitMapLongTimer** and **DigitMapShortTimer** parameters. Whether to use the long or short interdigit timer depends on the current rule matching states. The DMP maintains a matching state for each rule in the digit map as it processes each input digit. The following states are defined:

- **Partially Matched (PM)** – The rule partially matches the accumulated input sequence. Initially all rules are in this state before any digit is entered. Rules in this state have the potential of becoming EM or IM as more digits are entered. Example: 1234 partially matches the rules xxxxxxxx, 1xxxxx, 1234567, <123:>xxxx.
- **Exactly Matched (EM)** – The rule exactly matches the accumulated input sequence. However, any further input digit turns this rule into the MM state. Example: 1234 exactly matches the rules xxxxx, 1234, 1xxxx, <123:5678>x.
- **Indefinitely Matched (IM)** – The rule matches the accumulated input sequence indefinitely, with a variable length such that the rule can potentially stay as IM as more matching digits are entered. Example: 011853 indefinitely matches the rules xx., 011xx., <011:>xx.
- **Mismatch (MM)** – The rule doesn't match the accumulated input sequence. This state won't change as more digits are entered. Example: 1234 mismatches the rules 123, 1xx, 12345.

Rules in the EM or IM state are candidates to be selected by the DMP. After processing a new digit, the DMP returns a final decision if any of the following conditions holds:

- All rules are the MM state. The DMP returns an error.
- One or more rules are in the EM state with no rules in the IM state. DMP returns the best matched EM rule. If the best matched rule is a barring rule, DMP returns an error instead.

Otherwise, DMP starts the short interdigit timer if there is at least one rule in the EM state, or else the long one. When the interdigit timer expires, DMP makes a timely decision by returning the best matched rule at that moment if one is found, or else a timeout error. Again if the best matched rule in this case is a barring rule, DMP returns an error instead. Note that the timer to wait for the first input digit isn't governed by the interdigit timer, but the duration of dial tone being played and could be a lot lengthier than the long interdigit timer.

The best matched rule is the one that has the most specific literals matching the input digit sequence. For example, the input sequence 1234 matches the rule 123x better than 1xxxx. On the other hand, an EM rule is always selected over an IM rule.

Finally, the default interdigit timer can be overridden by appending the  $S_n$  element at the end of the rule ( $n = 0-9$ ).

Consider this simple digit map:

```
(<1408>xxx xxxx)
```

As soon as you enter 7 digits, the DMP returns a complete number by prepending the accumulated digits with 1408.

Consider another simple map:

```
(xx.)
```

After you dial one or more digits, the DMP returns the accumulated digits as a complete number when the long interdigit timer expires.

Combine the last two maps:

```
(xx. | <1408>xxx xxxx)
```

After you dial one or more digits (but fewer than seven digits), the DMP returns the accumulated digits as a complete number when the (long) interdigit timer expires. As soon as seven digits are entered, the DMP returns 1408 followed by the accumulated seven digits when the (short) interdigit expires. On the eighth digit and beyond, however, the DMP considers the first rule only and returns the accumulated digits as-is when the (long) interdigit timer expires.

Now add an S4 timer to the second rule:

```
(xx. | <1408>xxx xxxxs4)
```

In this case, the DMP behaves exactly the same as the last, except that the short interdigit timer the DMP uses upon receiving the seventh digit is overridden by a 4-second timer. Thus you've as long as 4 seconds instead of 2 to dial the eighth digit.

## ***Force an Interdigit Timeout with a Pound(#) Key***

When dialing, you can force an interdigit timeout with a # key instead of waiting for the DMP to timeout its own long or short timer. This is allowed as long as the # key doesn't match the current element of any PM rules. Otherwise the DMP consumes the # key instead of triggering a timeout.

Consider the digit map (33xx.)

If you enter 333#, the DMP immediately returns the number 333.

Now consider the digit map (33xx.|333#1234x.)

If you enter 333#, the DMP won't return, but continues to wait for further input or for its interdigit timer to expire. Note that the first rule "33xx." is now in the MM state since the digit # doesn't match "x". You can continue to enter 1234#, or 1234 and wait for a long interdigit timeout for the DMP to successfully return 333#1234.

## ***Invoke a Second Dial Tone in a Digit Map***

You can tell the device to start a tone after a certain pattern of digits have been dialed by specifying the element {t=<tone>} within a digit map, where <tone> is a 1- to 3-letter name of the tone to play. The tone stops when you enter the next digit. For example:

```
(**1{t=di2} (Msp) | **8{t=od} (Mli))
```

tells the device to play Second Dial Tone when you dial \*\*1, or play Outside Dial Tone when you dial \*\*8. Here is a full list of acceptable (case-insensitive) values of <tone>:

- bu = Busy Tone

- `cf` = Call Forwarded Dial Tone
- `cm` = Confirmation Tone
- `co` = Conference Tone
- `cw1` - `cw10` = Call Waiting Tone 1-10
- `di` = Dial Tone
- `di2` = Second Dial Tone
- `fb` = Fast Busy Tone
- `ho` = Holding Tone
- `od` = Outside Dial Tone
- `pr` = Prompt Tone
- `rb` = Ringback Tone
- `ro` = Reorder Tone (same as fast busy)
- `si1` - `si4` = SIT TONE 1 - 4
- `st` = Stutter Tone
- `0` - `9`, `*`, `#`, `a` - `d` = DTMF 0 - 9, \*, #, A - D

## Change an Interdigit Long Timer Dynamically After a Partial Match

The device starts off with the interdigit long timer set to the configured **DigitMapLongTimer** value when processing a new digit sequence by a digit map. You can change the long timer as some patterns are partially matched by embedding the syntax `{L=<time>}` within a rule in the digit map, where `<time>` is the desired number of seconds for the long timer. For example:

```
(011 853 xxxx xxxx{L=5}x. |xx.)
```

Here the long timer is shortened to 5 seconds after you enter `011 853 + 8` digits. Hence, the device declares that a complete number is collected in 5 seconds when it doesn't receive any more digits. Without the `{L=5}` syntax, you have to wait for 10 seconds (by default) for the same to happen.

## User-Defined Digit Maps

The **User Settings – User-Defined Digit Maps** section of the device configuration web page provides for 10 user-definable digit maps. These digit maps are referred to as User-Defined Digit Map 1 to 10. Each user-defined digit map is specified with two parameters:

- **Label:** An arbitrary string for referencing this digit map in other digit map specification. The value should be 2 to 16 characters long. For example, "friends". In this case, `(Mfriends)` can be referenced in other digit maps.
- **DigitMap:** A digit map to restrict the numbers that can be dialed or called with this service. Maximum length is 511 characters.

By default both parameters are empty, except for User-Defined Digit Map 1. See the following section.

### A User-Defined Digit Map For IPv4 Dialing

The default values of the parameters for User-Defined Digit Map 1 are set to the following values to support IPv4 dialing:



- `$CFB` = call forward busy enable (trunk-specific; valid values: 0 for disable, 1 for enable).
- `$CFN` = call forward no-answer enable (trunk-specific; valid values: 0 for disable, 1 for enable).
- `$CFAN` = call forward unconditional number (trunk-specific; valid value: a token representing a call forward number).
- `$CFBN` = call forward busy number (trunk-specific; valid value: a token representing a call forward number).
- `$CFNN` = call forward no-answer number (trunk-specific; valid value: a token representing a call forward number).
- `$MWS` = message waiting state (trunk-specific; valid value: 0 for no new messages, 1 for one or more new messages).
- `$DND` = do-not-disturb enable (trunk-specific; valid values: 0 for disable, 1 for enable).
- `$BAC` = block-anonymous caller enable (trunk-specific; valid values: 0 for disable, 1 for enable).
- `$BCI` = block outbound caller-ID enable (trunk-specific; valid values: 0 for disable, 1 for enable).
- `$CWA` = call-waiting enable (global; valid values: 0 for disable, 1 for enable).
- `$BCI1` = block caller-ID once (global; valid value: 1 for enable).
- `$UBCI1` = unblock caller-ID once (global; valid value: 1 for enable).
- `$LBM1` = Loopback media (audio samples) once in the next call.
- `$LBP1` = Loopback RTP packets once in the next call.
- `$CDM1` = Codecs to enable in the next call (temporarily overriding any codec preferences in device configuration). Each bit of its value represents one audio codec:
  - Bit0 (LSB) = G711u
  - Bit1 = G711a
  - Bit6 = G729
 Other bits are reserved.
- `$LDN` = last dialed number (for redial) (global; read-only).
- `$BAR1` = Enable Barge-In 1 on the next call (global; valid value: 1 for enable).
- `$Bxrn` = Blind Transfer Target Number (global; valid value: a token representing the target number).
- `$LCR` = last caller's number (for call return) (global; read-only).
- `$SPD[n]` = number for the speed dial  $n$  ( $n = 1 - 99$ ) (global; valid value: literal or token representing a phone number).
- `$CODE` = the digit(s) representing the variable part of a star code (see examples below; read-only).

Variable names are case-insensitive.

## Star Code Script Actions (ACT)

The general format of an action: `ACT(par, par, ...)`

The following actions are supported:

- `set (VAR, token)` = Set the given `VAR` to the value represented by `token`.
- `call (token)` = Call the number represented by `token`. **Phone Settings::OutboundCallRoute** is applied when making the call (but not the `DigitMap`).

- `rpdi(token)` = Repeat dial the number represented by token.
- `coll(VAR)` = Collect a number from the user and store it as the value of the parameter(s) represented by VAR. The number is collected with **Phone Settings::DigitMap** applied.
- `say(token)` = Display the value represented by token in an on-screen notification message. Values are announced as a list of alphabets or numbers, where token can be a literal (such as 1234) or another variable (such as \$CFAN or SP1(\$CFBN)).
- `wifiap` = Enable Wi-Fi access point mode for quick Wi-Fi configuration from a web browser.
- `blst` = Add the number of the last caller to the **X\_BlockCallers** list.

Action names are case insensitive.

## Star Code Script Format

General Format: `code, name, action1, action2, action3, ...`

- `code` = The star code, such as \*72. It may contain a variable part enclosed in parentheses, such as \*74(x|xx). The variable part as entered by the user are stored in the variable \$CODE.
- `name` = A descriptive name of the function of this star code, such as Call Forward Unconditional.
- `action1, action2, ...` = A valid action with parameters.

Actions are carried out one-by-one in the order as specified in the script.

Restrictions:

- At most 1 `coll` action per code.
- Either 1 `say` or 1 `call` action at most per code, and it must be the last action in the script.

## Star Code Script Examples

The following examples are taken from some of the default star code scripts in your phone.

- \*69, Call Return, `call($LCR)`  
Calls the number of the caller who last rang your phone.
- \*07, Redial, `call($LDN)`  
Redials the last dialed number.
- \*72, Call Forward Unconditional, `coll($cfan), set($cfa,1)`  
Your phone collects a number from you according to the digit map, then sets the `CallForwardUnconditionalNumber` on all trunks to the collected value, and finally sets the `CallForwardUnconditionalEnable` on all trunks to Yes.  
To modify the script to enable `CallForwardUnconditional` on SP1 only, change it to  
➤ \*72, Call Forward Unconditional SP1, `coll(SP1($cfan)), set(SP1($cfa),1)`
- \*67, Block Caller ID Once, `set($BCI1,1)`  
Enables masking caller ID information one time for the next call on any trunk.

# Customizing Devices

You can customize your VVX business IP phones, OBi Edition using applications and various keysets.

## Applications

Applications enable the functionalities of VVX business IP phones, OBi Edition.

### ***Built-In Phone Applications***

You can launch most of the built-in phone apps from the **Home** screen. The following apps are available:

#### **Built-In Phone Applications**

<b>Display Caption (Default)</b>	<b>Action URL</b>	<b>Summary</b>
Contacts	Phone://phone-book	A list of contacts where you can add, remove, or modify entries.
Current Calls	Phone://calls	A list of all the current calls on the phone, where you can manipulate the calls individually, or start more calls. The app also pops up automatically when a new call is added
Call History	Phone://call-history Secondary Apps: Phone://call-history/all Phone://call-history/received Phone://call-history/missed Phone://call-history/outgoing	A list of as many as 200 previous calls with time-stamp, duration, call statistics, and other information for each entry. The call history is further divided into four groups: All Calls, Missed Calls, Outgoing Calls, and Received Calls, which you can access via a corresponding shortcut as a secondary app.
Preferences	Phone://preferences	A list of user preferences settings, such as background picture, screen saver, language, screen brightness. The order of the items and contents is configurable.
Settings	Phone://settings	A list of phone settings (usually for administrators) such as Network, Firmware Update. The order of items and contents is configurable.
Product Info	Phone://prod-info	Information about the product such as hardware and software versions, device MAC, and so forth. This is a read-only list. The order of the items and contents is configurable.
Buddy List	Phone://buddy	A Buddy List service based on XMPP.

You can find more information on the operations of these apps in the *Polycom VVX Business IP Phones, OBi Edition User Guide*.

## OBiPhoneXML Applications

The VVX business IP phones, OBi Edition, provide a programmatic interface for developing third-party applications that you can download and execute on the phones. You can develop these applications (called OBiPhoneXML apps) using a proprietary mark-up language called OBiPhoneXML to create solutions involving deployed phones and servers.

Typical applications include:

- Access to corporate directories
- Voicemail navigation
- Call queue monitoring
- Integration with customer databases
- Generation of QR code

OBi edition phones now support automatic generation of a QR code that displays the phone model and MAC address when selecting any of options in the phone's **Product Info** menu. All phones show a QR code for Model, MAC: {mac} when clicking any of the items on the **Product Info** menu from the phone interface.

In addition, VVX 250/350/450 phone support implementing custom QR codes using an ObihaiIPPhoneImageFile XML. The following example shows pushing a custom QR code for an online survey.

```
{
  "password": "1234",
  "body": "<ObihaiIPPhoneImageFile timeout='30'><Title icon='1'
bgcolor='blue' bold='yes'>Please Complete
Survey:</Title><Image>http://10.9.80.175:8080/qrcode_1.png</Image><IconList>
<Icon
index='1'>https://www.obitalk.com/Downloads/assets/custserviceicons/030-error.
png</Icon></IconList></ObihaiIPPhoneImageFile>"
}
```

For more information about OBiPhoneXML apps, see the *Polycom OBiPhoneXML Applications API Guide*.

## Softkey Set Customization

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. Often, the set that displays is based on the highlighted item on a screen with multiple items listed. You can customize some of the softkey sets by entering a list of softkey specifications as a comma-separated list in the corresponding softkey parameters.



## Softkey Set Parameter Syntax

The value of a softkey set parameter is a comma-separated list of softkey specifications, with the following general format: `{softkey-spec}[|{softkey-spec}], {softkey-spec}[|{softkey-spec}]`, and so forth.

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 example, the `missed|lines` option specifies that the **Missed** softkey displays when it isn't hidden, otherwise the **Lines** softkey displays. If the alternative softkey is an empty key (unspecified or not found), an **<Empty>** softkey displays as the alternative. Otherwise, a hidden softkey is packed and doesn't occupy any softkey slot on the display.

For example: the `set barge, monitor, , newcall` shows **Barge In, Monitor, <Empty>**, and **New Call** when `barge` and `monitor` aren't hidden, and **<Empty>, New Call** when `barge` and `monitor` are hidden. On the other hand, the `set barge|,monitor|,,newcall` shows the same when `barge` and `monitor` aren't hidden, but **<Empty>, <Empty>, <Empty>, New Call** when they are.

## Softkey Specification

General format: `{id}[?][;attr[;attr[...]]]`

`{id}` is the standard softkey ID, as shown in the table below.

`?` is an optional syntax to designate if the key shouldn't be hidden if the hidden condition is matched.

You can include one or more `attr` attribute elements separated by a semicolon (;). Attributes are usually in the following format `{attribute-name} [= {attribute-value}]`. You can enclose `{attribute-value}` in double quotes (") to preserve all the enclosed characters. Otherwise, all the white spaces are removed. The enclosing double quotes aren't counted as part of the value.

For example: `number=**922222222, name="Echo Server", or url="http://abc.com/call.png, auth-token=abcdefg"`. That is, the phone interprets all contents within the double quotes as part of the attribute value.

All softkeys support the optional label and icon attribute, which is the text and icon that shows on the screen inside the softkey window. Without the label or icon attributes specified, the default label and icon of the softkey is used. Some softkeys that you can enable or disable offer a `label1` and `icon1` option that is used when the underlying function is in the "on" state. The **acd** softkey also supports `label2`, `label3`, `icon2`, and `icon3` options to cover the four states of an ACD Agent.

In addition, all softkeys support the optional style attribute that specifies how to style the icons and labels for the softkey. The value of the style attribute should be equal to the ID attribute of the `<style>` elements in the `<SoftKeyStyles>` XML specified for the **Screen Item Customization::SoftKeyStyles** parameter. If you don't specify the style attribute or it's not found, the `<style>` element with `id=default` is applied to the softkey. Otherwise, the softkey is styled with the internal default style.

You can specify the following aspects in a softkey style:

- Softkey background picture when the key is in normal state, pressed down or highlighted state, and alert state. You can set the picture as an internal picture file or an http or https URL.
- The placement of the label and icon within the softkey window (left/right/center/top/bottom), which you can adjust with x and y offset
- The color, size, and font for the label
- The height and width of the softkey icon and label

## Assignable Softkeys

You can assign these softkeys to the configurable softkey sets.

### Assignable Softkeys

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
aans	(User Preferences) Auto Answer Intercom Calls		Auto Answer	Auto Answer			Any
acd	ACD Agent Sign On or Off		Signed Out	Available	Unavailable	Wrapping Up	Any
achis	All Calls History		All Calls				Any
acturl	Action URL		XML App				Any
bac	(User Preferences) Block Anonymous Call Example: <code>bac;label="Hide CID"; label1="Show CID"; number&gt;**9222222222</code>		Block Anonymous Call	Block Anonymous Call			Any
bci	(User Preferences) Block Caller ID (also known as Anonymous Call)		Block Caller ID	Block Caller ID			Any
cfa	(User Preferences) Call Forward Unconditional (also known as Call Forward All) enable/disable.		Call Forward	Call Forwarded			Any
chis	Call History		Call History				Any
cwa	(User Preferences) Call Waiting enable/disable		Call Waiting				Any
dnd	(User Preferences) Do Not Disturb		Do Not Disturb	Do Not Disturb			Any
dnr	(User Preferences) Do Not Ring		Do Not Ring	Do Not Ring			Any

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
emlogin	A login utility to collect a user name and password from the user and trigger ITSP provisioning (by executing the currently configured ITSP ConfigURL) and use the given user name/password for authentication. The ITSP can use this step for binding a device to the login user name.		Prov Login				Any
lcr	(Last) Call Return		Call Return				Any
ldap	LDAP directory search		LDAP				Any
ldn	Last Number Redial		Redial				Any
lines	Shows a list of available of lines on screen to choose from to make a new call.		Lines				Any
mchis	Same as missed		Missed				Any
missed	Missed Calls History	No New Missed Calls	Missed				Any
mwi	Message Status (also known as Message Waiting Indication)		Messages				Any
ochis	Same as redial.		Redial				Any
phbk	Invokes the Phone Book app.		Phone Book				Any
rchis	Received Calls History		Received Calls				Any
redial	Outgoing Calls History		Redial List				Any
pg1	Page Group 1		Page Group 1	Page Group 1			Any
pg2	Page Group 2		Page Group 2	Page Group 2			Any

**Assignable Softkeys (continued)**

<b>Softkey ID</b>	<b>Description</b>	<b>Hidden Condition</b>	<b>Default Label</b>	<b>Default Label1</b>	<b>Default Label2</b>	<b>Default Label3</b>	<b>Where</b>
pg3	Page Group 3		Page Group 3	Page Group 3			Any
pg4	Page Group 4		Page Group 4	Page Group 4			Any
pg5	Page Group 5		Page Group 5	Page Group 5			Any
pg6	Page Group 6		Page Group 6	Page Group 6			Any
pg7	Page Group 7		Page Group 7	Page Group 7			Any
pg8	Page Group 8		Page Group 8	Page Group 8			Any
pg9	Page Group 9		Page Group 9	Page Group 9			Any
pg10	Page Group 10		Page Group 10	Page Group 10			Any

## Assignable Softkeys (continued)

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
sd	<p>Speed Dial. It takes four additional optional attributes:</p> <ul style="list-style-type: none"> <li>• name: the caller name corresponding to the speed dial number</li> <li>• number: the speed dial number, which can be full or simple</li> <li>• ptt: (no value) enable Push-To-Talk option</li> <li>• send: {digit-codes} to automatically dial some digits after the call, where {digit-codes} is a sequence of the following case-sensitive codes: <ul style="list-style-type: none"> <li>▲ 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.</li> <li>▲ S – Pause for 3 seconds.</li> <li>▲ s – Pause for 1 second.</li> <li>▲ U"{prompt}" – Prompt users to enter one or more digits manually with the given {prompt} shown on the screen. Users then press the <b>OK</b> softkey to continue.</li> <li>▲ A – Wait for the called party to answer before continuing.</li> </ul> </li> </ul> <p>Note that the phones start executing the first code in {digits} when the call receives early media or when the call is answered, whichever happens first.</p>		Speed Dial				Any

## Assignable Softkeys (continued)

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
sd (cont.)	<p>For example:  <code>send=Ass1234U"Enter  Passcode"5678</code></p> <p>It's acceptable to have the send attribute specified in a speed dial without a number (for example, <code>sd;send=1234#;label=Park</code>). In this case, the speed dial is used to send out the list of digits on a connected call, which you should only use in the <b>CallConnected</b> softkey set.</p> <p>Example:  <code>sd;label="Echo Test";  number&gt;**9222222222  sd;send=1234;label=Park</code></p>		Speed Dial				Any
ciurl	Invokes the Call-Info URL of the call as an action URL.	Call-Info URL not available	Call Info				Ringing, CallConnecting, CallConnected, CallHolding
dial	Calls the entered number.		Dial				Dialing, OnDialing
backspace	Removes the digit or character to the left of the cursor in the input box.		Backspace				Dialing, OnDialing
mm	Returns to the Main menu to select an item as the number to call.		Main Menu				Dialtone
mode	Switching Input Mode		Switch Mode				Dialtone, Dialing, OnDialing
switch.line	Switching Line		Switch Line				OnDialing
answer	Answers the selected incoming (and ringing) call.		Answer				Ringing

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
join	Transfers one conferee to the other in a 3-way (local-mixed) conference call.	Less than or more than two calls on the phone are currently in the Connected state.	Join				CallConnected
reject	Rejects the selected incoming (and ringing) call as busy.		Reject				Ringing
newcall	Starts a new call with dial tone.		New Call				Ringing, CallConnected, CallHolding, CallConnecting, CallParked, SCAInUse
end	Ends the selected call.		End				CallConnecting, CallConnected, XferTrying, XferRinging, XferConnected, ConfTrying, ConfRinging, ConfConnected
hold	Holds the selected call. This key is mutually exclusive with the Resume key.		Hold				CallConnected
privhold	Holds the selected call privately when the Call Appearance is an SCA.		Private Hold				CallConnected

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
resume	Resumes the selected call. This key is mutually exclusive with the Hold key.		Resume				CallHolding
conf			Conference				CallConnected, CallHolding
add2conf	Resumes the holding call to add to the current conference.	No other call on the phone that is currently in the Connected state	Add To Conf				CallHolding
add2conf	Answers the incoming call to directly add it to the current conference.	No other call on the phone that is currently in the Connected state	Add To Conf				Ringing
transfer	<p>Transfers the selected call. It takes these additional optional attributes:</p> <ul style="list-style-type: none"> <li><code>name</code>: the caller name corresponding to the speed dial number</li> <li><code>number</code>: the speed dial number, which can be full or simple</li> </ul> <p>Example:  <code>transfer;label="Supervisor"</code>  <code>;</code>  <code>number=sp2(1034)</code></p>		Transfer				CallConnected, CallHolding



**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
bxfer	<p>Blind transfers the selected call. It takes these additional optional attributes:</p> <ul style="list-style-type: none"> <li>name: the caller name corresponding to the speed dial number</li> <li>number: the speed dial number, which can be full or simple</li> </ul> <p>Note that bxfer doesn't time out on dialing the target number and doesn't apply digit map on the dialed number.</p> <p>Example:  bxfer;label="To Vmail";  number=sp1(*28)</p>		Blind Transfer				CallConnected, CallHolding
bxfer2	<p>Same as transfer softkey with respect to collecting the target number from the user, but a blind transfer is complete after the target number is collected. In contrast to bxfer, bxfer2 plays dial tone, applies digit map, and would time out as user enters the target number.</p>		Blind Transfer				CallConnected, CallHolding
cbctl	<p>Invokes conference bridge control app to view who's on the bridge or to apply certain control on the selected participants.</p>	Peer isn't a conference bridge.	Bridge Control				CallConnected
park	<p>Parks the selected call against the current extension.</p>		Park				CallConnected, CallHolding
dispcode	<p>Enters disposition code for the selected call.</p>	Disposition Code service not enabled	Disposition Code				CallConnected, CallHolding
trace	<p>Starts a call trace for the selected call.</p>	Call Trace service not enabled.	Trace				CallConnected, CallHolding

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
escalate	Escalates the selected call to a supervisor.	Escalate service not enabled.	Escalate				CallConnected, CallHolding
rec.start	Starts call recording.	Call Recording service not enabled.	Rec.Start				CallConnected, CallHolding
rec.stop	Stops call recording.	Call Recording service not enabled.	Rec.Stop				CallConnected, CallHolding
rec.pause	Pauses call recording	Call Recording service not enabled.	Rec.Pause				CallConnected, CallHolding
rec.resume	Resumes call recording.	Call Recording service not enabled.	Rec.Resume				CallConnected, CallHolding
xfer.now	Completes call transfer operation		Transfer Now				XferRing, XferConnected
conf.now	Starts the conference now. That adds the conference target to the conference by resuming the original call where the conference function was started.		Conference Now				ConfRing, ConfConnected
split	Turns the call to conference target (that hasn't joined the conference yet) into a regular call, at the same call state. This is equivalent to pressing the Cancel key on the phone while the screen is showing an overlay of the call progress with the conference target.		Split				ConfTrying, ConfRing, ConfConnected
split	Breaks up a merged conference call item into individual call items on screen with each conferee and places all conferee calls on hold.	Highlighted call item is NOT a merged conference call.	Split				Conferencing

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
barge	Barges in on a (shared) call that is in progress or holding at another phone.	(Share) Call isn't connected or holding.	Barge In				SCAInUse
monitor	Monitors a (share) call that is in progress on another phone.	(Share) Call isn't connected.	Monitor				SCAInUse
pickup	Picks up a parked call at the highlighted extension	No call parked	Pickup				CallPark ed
mute	Selectively mutes a call.		Mute	UnMute			CallConn ected, CallConn ecting
coach	Requests to coach the call peer, or request to get out of coaching from the call peer		Coach	No Coach			CallConn ected
coachee	Enables Coachee mode on this call, or disable Coachee mode on this call.		Coachee	No Coachee			CallConn ected
nohc	No Hold New Call – Starts a new call without placing the current calls on hold first.		No Hold Call				CallConn ected
divert	Diverts a ringing incoming call to a target number that user enters.		Divert				Ringing
ns.acd	Sets ACD Agent State on the selected SP <i>n</i> service.	Feature Sync Disabled	Signed Out	Availabl e	Unavail able	Wrappi ng Up	NetServi ces
ns.bci	Turns on or off Anonymous Call feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Anony mous	Anony mous			NetServi ces
ns.dnd	Turns on or off Do Not Disturb feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Do Not Disturb	Do Not Disturb			NetServi ces
ns.exec	Turns on or off Executive Call Filtering feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Exec Filter	Exec Filter			NetServi ces
ns.xass	Turns on or off Executive Assistant Divert on the selected SP <i>n</i> service.	Feature Sync Disabled	Exec Assist	Exec Assist			NetServi ces

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
ns.rmoff	Turns on or off Remote Office feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Rm Office	Rm Office			NetServices
ns.simring	Turns on or off Simultaneous Ringing feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Simlt Ring	Simlt Ring			NetServices
ns.cfa	Turns on or off Call Forward All (also known as Unconditional) feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Call Forward	Call Forward			NetServices
ns.cfb	Turns on or off Call Forward Busy feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Cfwd Busy	Cfwd Busy			NetServices
ns.cfna	Turns on or off Call Forward No Answer feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Cfwd No Ans	Cfwd No Ans			NetServices
ns.secclass	Gets or sets Security Calls on the selected SP <i>n</i> service.	Feature Sync Disabled	Sec Class				NetServices
ns.dispcode	Sends a call disposition code on the selected SP <i>n</i> service.	Feature Sync Disabled	Disp Code				NetServices
ns.hotel	Gets or sets Hoteling feature on the selected SP <i>n</i> service.	Feature Sync Disabled	Hoteling				NetServices
ns.bwanw	Gets or sets BroadWorks Anywhere feature on the selected SP <i>n</i> service.	Feature Sync Disabled	BW Anywhere				NetServices
ns.dir	Loads, views, or refreshes the network directory on the selected SP <i>n</i> service.	Feature Sync Disabled	Dir				NetServices
ns.clog	Loads, views, or refreshes the network call history on the selected SP <i>n</i> service.	Feature Sync Disabled	CLog				NetServices
ns.buddy	Loads, views, or refreshes the Buddy List on the selected SP <i>n</i> service.	Feature Sync Disabled	Buddy List				NetServices
blf.call	Calls the monitored extension.		Call				BLFCall

**Assignable Softkeys (continued)**

Softkey ID	Description	Hidden Condition	Default Label	Default Label1	Default Label2	Default Label3	Where
blf.answer	Answers the highlighted call that is ringing on the monitored extension.	<b>X_BlfDirectedCallPickup</b> is disabled for the SP $n$ service.	Answer				BLFCall
blf.pickup	Picks up a call that is currently parked against the monitored extension	<b>X_BlfCallPickup</b> is disabled for the SP $n$ service.	Pick Up				BLFCall
blf.barge	Barges in the highlighted call that is currently active on the monitored extension.	<b>X_BlfBargeIn</b> is disabled for the SP $n$ service.	Barge In				BLFCall
blf.monitor	Barges in to monitor only the highlighted call that is currently active on the monitored extension.	<b>X_BlfBarge</b> is disabled for the SP $n$ service.	Monitor				BLFCall
blf.coach	Barges in to coach (also known as whisper) only the highlighted call that is currently active on the monitored extension.	<b>X_BlfWhisper</b> is disabled for the SP $n$ service.	Coach				BLFCall

## Softkey Set Parameters

The following table lists the softkey set parameters:

**Softkey Set Parameters**

Parameter Group	Parameter	Description
<b>IP Phone – Soft Keys – Softkey Sets</b>	<b>Home</b>	The softkey set shown on the Home Screen. The default is <code>redial,cfa,dnd,missed lines</code>
<b>IP Phone – Soft Keys – Softkey Sets</b>	<b>SCAInUse</b>	The softkey set shown on the Calls app screen when the selected call is a shared Call Appearance on another phone. The default is <code>barge ,monitor ,,newcall</code>
<b>IP Phone – Soft Keys – Soft Key Sets</b>	<b>CallParked</b>	On-hook dialing. Default is <code>switch.line,dial,mode,backspace</code>

## Softkey Set Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>CallConnected</b>	Call is connected. Default is end,hold,conf,transfer,privhold,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume,tomobile
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>CallHolding</b>	Call is holding. Default is end,resume,add2conf,conf,transfer,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume,tomobile
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>Dialtone</b>	Dialtone plays before first digit is entered. Default is redial,phbk,mode,lines
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>Dialing</b>	Dialing a number with at least 1 digit entered. Default is redial,dial,mode,backspace
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>OnDialing</b>	On-hook dialing. Default is switch.line,dial,mode,backspace
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>Ringing</b>	Incoming call ringing. Default is answer,reject
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>CallConnecting</b>	Trying outgoing call, including the case when called party is ringing. Default is end,,newcall
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>ConfTrying</b>	Trying to call the new conferee end
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>ConfRinging</b>	Conferee is ringing. Default is end,,conf.now
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>ConfConnected</b>	Connected with the conferee, including the case when the call is Holding. Default is end,hold,resume,conf.now
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>ConfBridgeControl</b>	Connected with an external conference bridge. Default is eend,deaf,mute
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>Conferencing</b>	A virtual call item representing all the conferee calls in a conference call. All conferee calls by definition are in the Connected state. Default is: end,split,newcall
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>XferTrying</b>	Trying to call the transfer target. Default is end
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>XferRinging</b>	Transfer target is ringing. Default is end,,xfer.now
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>XferConnected</b>	Connected with the transfer target, including the case when the call is Holding. Default is: end,hold,resume,xfer.now

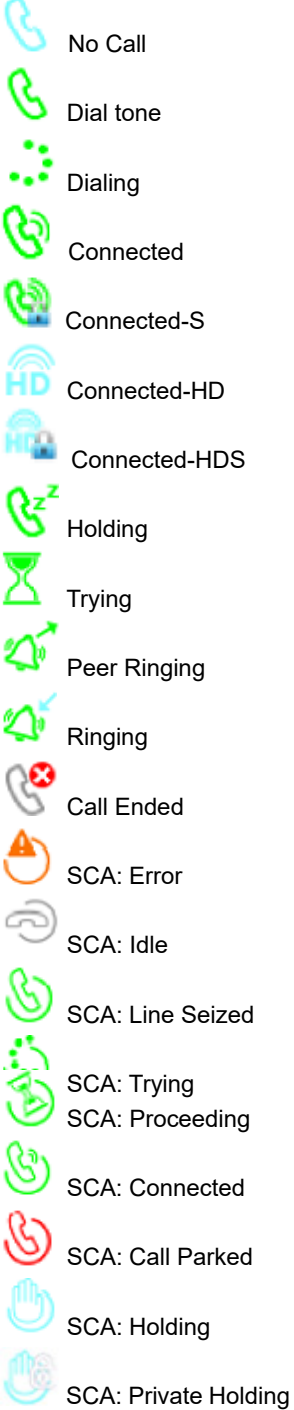
**Softkey Set Parameters (continued)**

Parameter Group	Parameter	Description
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>NetServices</b>	The softkeys at the top level of the Network Services app is where one of the available SP services is highlighted. Default is: ns.bci, ns.dnd, ns.cfa, ns.bac, ns.acd
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>BLFCall</b>	The softkeys to present when user press+hold a BLF key to bring up a list of calls on the monitored extension. Default is: blf.call, blf.answer, blf.pickup, blf.barge, blf.coach, blf.monitor
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>Settings</b>	Settings menu. Default is reboot, reset, fwupdate
<i>IP Phone – Soft Keys – Soft Key Sets</i>	<b>CallError</b>	Outgoing call error encountered, such as called party busy or not found. Default is end, , , newcall

## Feature Keys














You can configure feature keys on the phone or VVX EM50 expansion module to perform one of many different predefined functions with a corresponding multicolor LED that shows the status of the assigned function instance. Side Car Keys, Virtual Line Keys, and Programmable Keys, and are all feature keys that you can configure. Phone administrators can configure each feature key to perform one of the following functions listed in this table:

List of Feature Keys and Their Functions







Feature Key Function	Description	Icon
<p>Call Appearance</p>	<p>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.</p> <p>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.</p> <p>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.</p> <p>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). See the <a href="#">Shared Line and Shared Call Appearances (SCA)</a> section for more details on this topic.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <code>Service</code>: Optional. The service or line with which to bind the key</li> <li>• <code>MaxCalls</code>: Number of calls managed by this call key</li> </ul>	




## List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
Line Monitor	<p>Monitor a Line (a voice service installed on the phone). The Line events that are monitored include:</p> <ul style="list-style-type: none"> <li>• Idle: no calls</li> <li>• Ringing: at least one incoming call</li> <li>• In Use: at least one active call</li> <li>• Holding: at least one call holding</li> </ul> <p>VLKW shows the monitored service name and account user name (usually same as the account DID number or extension). This function must be bound to the specific voice service that it monitors.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <code>Service</code>: Required. The service or line to monitor</li> </ul>	 Idle  Ringing  In Use  Holding
BLF (Busy Lamp Field)	<p>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:</p> <ul style="list-style-type: none"> <li>• Ringing: at least one incoming call</li> <li>• Holding: at least one call holding</li> <li>• Busy: at least one active call</li> <li>• Idle: no calls</li> <li>• Call parked (against the monitored extension)</li> </ul> <p>VLKW shows the bound service name and the monitored extension (or DID number, or account user name). This function must be bound to a specific voice service.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <code>Service</code>: Required. The service that provides the monitoring function</li> <li>• <code>Number</code>: Required. The extension (on the specified service) to monitor. It may contain multiple attributes <ul style="list-style-type: none"> <li>▲ <code>ptt</code> – (no value). Make the call a Push-to-Talk when calling the monitored extension</li> <li>▲ <code>spd</code> – An alternative number to call when calling the monitored extension</li> <li>▲ <code>bx</code> – (no value). Enable one-touch blind transfer behavior</li> </ul> </li> </ul>	 Idle  Busy  Call Parked  Ringing  Offline  Holding
Call Park Monitor	<p>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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <code>Service</code>: Required. The service that provides the parking lot orbit: parking and monitoring function</li> <li>• <code>Number</code>: Required. The parking lot orbit to park and monitor</li> </ul>	 Idle  One Call Ringing  Offline








## List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
Presence Monitor	<p>Monitor the presence/status of one buddy in a Buddy List. It also serves as speed dial to that buddy.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li><b>Service:</b> Required. The service that provides the XMPP service for this function</li> <li><b>Number:</b> 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: <code>abcd-12345@gmail.com</code>, you may specify <code>abcd</code> for the Number field to monitor this buddy, provided there's no other JID in the buddy list that starts with <code>abcd</code>.</li> </ul>	 Online  Offline  Extended Away  Away  Do Not Disturb  Unknown











## List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
Speed Dial	<p>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.</p> <p>Parameters:</p> <p><i>Service</i>: Optional. The suggested service the make the call with</p> <p><i>Number</i>: Required. The number to call. It may include the following attributes:</p> <ul style="list-style-type: none"> <li>• <code>ptt</code> (with no value): make the key a Push-To-Talk key.</li> <li>• <code>bx</code> – (with no value). Enable one-touch blind transfer behavior</li> <li>• <code>send={digit-codes}</code> to automatically dial some digits after the call, where {digit-codes} is a sequence of the following case-sensitive codes: <ul style="list-style-type: none"> <li>▲ 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.</li> <li>▲ S – Pause for 3 seconds.</li> <li>▲ s – Pause for 1 second.</li> <li>▲ U"{prompt}" – Prompt the user to enter one more digit manually on the phone with the given {prompt} shown on the screen, then press <b>OK</b> to continue.</li> <li>▲ A – Wait for the called party to answer before continuing.</li> </ul> </li> </ul> <p>Note that the phones starting executing the first code in {digits} when the call receives early media or when the call is answered, whichever happens first.</p> <p>For example: <code>send=Ass1234U"Enter Passcode"5678</code></p> <p>It's acceptable to have the send attribute specified in a speed dial without a number (for example, <code>Number=;send=1234#</code>). In this case, the speed dial can be used to send out the list of digits on a connected call.</p> <p>Note that if a voice service is specified for the speed dial either in the Service Field or the given number is a full number (one that includes voice service information), the phone doesn't apply digit map before calling the speed dial. Hence any *codes in the number aren't processed and the result may not be desirable. You can include the *codes that the phone needs to process by enclosing them in a pair of [...] at the beginning of the number field. For example: you may set the number field to <code>[*96]SP1(2113);ptt</code>. The phone then interprets *96 locally (to request auto-answer on the called party) and makes the call to 2113 using SP1 service.</p>	 Speed Dial









## List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
Speed Dial (continued)	<p><b>Using Speed Dial to store a Feature Access Code prefix</b></p> <p>There are many applications where a function the Service Provider can perform 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 UI. 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.</p>	 Speed Dial
Do Not Disturb	<p>Turn the <b>Do Not Disturb</b> 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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li><i>Service</i>: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls the phone</li> </ul>	 On  Off
Do Not Ring	<p>Turn the <b>Do Not Ring</b> 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).</p> <p>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.</p> <p>Parameters: None</p>	 On  Off
Block Anonymous Callers	<p>Turn the <b>Block Anonymous Caller</b> 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).</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li><i>Service</i>: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.</li> </ul>	 On  Off











List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
Block Outgoing Caller ID	<p>Turn the <b>Block Caller ID</b> 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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.</li> </ul>	 On  Off
Call Forward All	<p>Turn the <b>Call Forward All Calls</b> 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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Optional. The service to apply this feature on. If no service is specified, the feature applies to all calls on the phone.</li> </ul>	 On  Off
Auto Answer	<p>Turn the <b>Auto Answer (Intercom Calls)</b> feature on or off. Normally this feature is turned on so that incoming intercom calls are answered automatically by the phone on the speakerphone or headset. If this feature is turned off, incoming intercom calls are treated as regular calls and the phone rings normally. This function cannot be bound to any specific voice service. It's applied system-wide to all incoming calls, no matter which service the calls come from.</p> <p>Parameters: None</p>	 On  Off
Call Waiting	<p>Turn the <b>Call Waiting</b> feature on or off. Normally, this feature is enabled and you can accept more incoming calls while already on a call. If this feature is turned off, all incoming calls are rejected as busy when you're on a call. This function cannot be bound to any specific voice service.</p> <p>Parameters: None</p>	 On  Off
Message Status (Monitor Voicemail Status)	<p>Monitor the number of messages in a mailbox. The function must be bound to a 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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Required. The SP service that provides the Voicemail service.</li> <li>• <i>Number</i>: Optional. The number to call (to check voicemail). It may include the following attributes:             <ul style="list-style-type: none"> <li>▲ <i>mbox</i>={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</li> </ul> </li> </ul>	 Messages  No messages









## List of Feature Keys and Their Functions (continued)

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

## List of Feature Keys and Their Functions (continued)







Feature Key Function	Description	Icon
Join/Leave Page Group 4	<p>Join/Leave Page Group 4. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Join/Leave Page Group 5	<p>Join/Leave Page Group 5. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Join/Leave Page Group 6	<p>Join/Leave Page Group 6. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Join/Leave Page Group 7	<p>Join/Leave Page Group 7. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Join/Leave Page Group 8	<p>Join/Leave Page Group 8. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined

## List of Feature Keys and Their Functions (continued)



Feature Key Function	Description	Icon
Join/Leave Page Group 9	<p>Join/Leave Page Group 9. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Join/Leave Page Group 10	<p>Join/Leave Page Group 10. If the user joins the group, then the phone automatically turns on the speaker when anyone in the group starts a page. You can also start the page by pressing down the feature key. This is known as PTT or Push-to-Talk (otherwise you're just listening). If allowed by the phone administrator, you can also "Clamp On" the feature key to talk continuously without needing to hold down the key.</p> <p>This function cannot be bound to any specific voice service. Parameters: None</p>	 Left  Joined
Change ACD Agent State	<p>Change/monitor an ACD (or Call-Center) Agent State to one of the following values:</p> <ul style="list-style-type: none"> <li>• Available (to take new calls)</li> <li>• Unavailable (to take new calls)</li> <li>• Signed Off</li> <li>• Wrapping Up (the last call)</li> </ul> <p>ACD stands for Automated Call Distribution is the primary way a call-center distributes calls among a number of agents working for the call-center. The ACD controller should only send incoming call to an agent whose state is "Available".</p> <p>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.</p> <p>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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <b>Service:</b> Required. The service that provides the ACD Agent function.</li> </ul>	 Signed Off  Available  Wrapping Up  Unavailable



List of Feature Keys and Their Functions (continued)

Feature Key Function	Description	Icon
<p>Guest User Login/Logout</p>	<p>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).</p> <p>This function must be bound to a specific voice service (that supports this feature).</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Required. The service that provides the Guest Login function.</li> </ul>	<p> Guest Logged In</p> <p> Guest Logged Out</p>
<p>Enter Disposition Code for the last call.</p>	<p>Enter a <b>Disposition Code</b> 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).</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Required. The service that provides the Disposition Code function</li> </ul>	<p> Enter Disposition Code</p>
<p>Next Tab</p>	<p>Press this key to switch to the next VLK Tab or cycle back to VLK Tab 1.</p> <p>This function can't bound to any specific voice service.</p> <p>Parameters: None</p> <p>Note that the multiple Line Key Tabs feature can be disabled also by disabling the option <b>User Preferences::LineKeyTabs</b>. In that case the "Next Tab" function does nothing.</p>	<p> Next Tab</p>
<p>Transfer</p>	<p>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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Service</i>: Optional. The suggested service to use to make the call to the transfer target, if the Number parameter is also specified</li> <li>• <i>Number</i>: Optional. The number of the transfer target to call. User aren't prompted to enter the transfer target number if this parameter is specified</li> </ul>	<p> Transfer</p>
<p>Blind Transfer</p>	<p>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.</p> <p>Parameters:</p> <ul style="list-style-type: none"> <li>• <i>Number</i>: 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</li> </ul>	<p> Blind Transfer</p>

**List of Feature Keys and Their Functions (continued)**

Feature Key Function	Description	Icon
Blind Transfer 2	Same as the <b>Transfer</b> function, collecting the target number from the user, but a blind transfer is performed after the target number is collected. In contrast to <b>Blind Transfer</b> , <b>Blind Transfer 2</b> plays dial tone, applies digit map, and would time out as user enters the target number.	 Blind Transfer 2
Action URL	Launch an OBiPhoneXML application at the given URL Parameters: <ul style="list-style-type: none"> <li><b>Number</b>: Required. The URL of the OBiPhoneXML application to run</li> <li><b>Name</b>: A name to identify the application on the phone screen</li> </ul>	 Launch Action URL

## Preassigned Feature Keys (PFKs)

The OBi Edition 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:

- **Transfer** (all phones)

Default **Function** = `Transfer`

Two valid alternatives to this behavior are to change **Function** to `Blind Transfer` or `Blind Transfer 2` when pressing the key would provide corresponding behavior, respectively.

- **Messages** (V VX 350 and V VX 450 only)

Default **Function** = `Message Status`

You should set up the **Number** and **Service** parameters of this key to access voicemail.

- **Hold** (all phones)

Default **Function** = `Hold`

## Feature Key Configuration Parameters

You can configure each feature key independently. The configuration of each feature key comprises of the following set of parameters:

- **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

Not used otherwise.

- **Number:** Required for:

- BLF
- Call Park Monitor
- Presence Monitor
- Speed Dial

Optional for:

- Transfer
- Blind Transfer

- **Name:** Optional for all functions. It's used as a nickname to refer to the entity specified in the **Number** parameter.
- **MaxCalls:** Maximum number of calls to overload on the key. This is only applicable if the function is Call Appearance. Therefore, 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.

It's very common to have multiple feature keys defined with the same function, such as **Call Appearance** and **Speed Dial**. It isn't advisable to have a particular monitor function with the same monitored entity configured on 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.

## Highlights of Feature Key Functions

This section highlights and defines the most commonly used feature key functions.

### Call Keys

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 needs an assigned call key. You need at least one call key to make or receive a call. Each call key can hold exactly one call. Usually, there are at least a few call keys defined on the phone to handle multiple call scenarios, such as call waiting and conference calls.

You can 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.

For incoming calls, the phone automatically assigns the call to an open call key. It first attempts to find a call key that is bound to the service account where the incoming call is on. If the phone can't find an open bound key, it looks for an unbound call key to assign the call to. If the phone can't find any open call keys, the call is rejected as busy.

You can "overload" a call key with as many as 24 calls. You can configure the number of calls to overload on a call key using the `MaxCall` parameter. When the key is overloaded, the LED pattern and the key display shows the states of only one call at a time. This call is referred to as the call-in-focus. The phone automatically selects the call-in-focus. The operation triggered by pressing the call key applies to the call-in-focus only according to its current state.

### Line Monitor Keys

Unlike a line key, a line monitor key monitors the status of a line.

Pressing the line monitor key results in the following behavior:

- Answer an incoming call. If there are multiple calls, the phone answers the oldest one.
- Start the dial tone, if there is call capacity.

### Speed Dial Keys

The Speed Dial function lets users configure a speed dial number from the phone UI. Users can press and hold the **Speed Dial** key until the feature key item displays on the screen. From there, users can input speed dial details. You can configure the service to use for calling with the speed dial. This function supports the Push-to-Talk (PTT) option.

In addition to the speed dial feature keys, there are 99 configurable speed dial numbers that users can access by dialing the corresponding 1-2-digit speed dial code (that is, 1 – 99). These 99 speed dial storages are referred to as the "Speed Dials 99" feature that is entirely independent of the speed dial feature keys discussed here.

### Busy Lamp Field Keys

Users can press a BLF key to monitor the call status of another extension. This feature operates in the context of an SP service. In many cases, the BLF key also acts as a speed dial key to call or transfer a call

to the monitored extension. This key also supports PTT when it's used to call the monitored extension. Note that the call or call transfer to the monitored extension uses the same underlying SP service.

## Call Park Monitor Keys

Call Park Monitor (CPM) is very similar to BLF, except each key monitors a call park orbit instead of another user extension. It only indicates the status whether a call is parked on the orbit or not. Press the key once to park the highlighted call or retrieve the call from the orbit.

## Presence Monitor

This function is used together with a Buddy List. You can configure a feature key to monitor the presence/status of a buddy in a buddy list. You can also use this key as a speed dial to that buddy.

The Buddy List feature operates in the context of an SP service. Pressing the Presence Monitor key to call a buddy uses the same underlying SP service to make the call. The PTT option is not available with this key when calling the buddy.

## Group Page Keys

The phone supports 10 multicast page groups called Page Group 1 through Page Group 10. Set a feature key with the Page Group 1 through Page Group 10 function to use the respective page group. Paging is one way. The phone doesn't play incoming audio when the user is talking. The configuration of each page group has a Push-To-Talk option that you can enable to require users to press the **Page** key to talk. The **Page** key also lets the user join or leave the group with one key press. For example when you don't want to receive an incoming page, you can temporarily leave the group. The LED color also reflects the current group-joining status as a reminder to the user.

## Line Keys and Virtual Line Keys

There are twelve (12) physical Line Keys (LKs) on the VVX 450, six (6) on the VVX 350, four (4) on the VVX 250, and two (2) on the VVX 150. On the VVX 450, the line keys are split half and half to the left and right of the LCD display. On the VVX 350, VVX 250, and VVX 150, the line keys are on the left side of the display.

By means of multiple "tabs" or pages on the VVX 450, VVX 350, and VVX 250, one single physical line key becomes multiple Virtual Line Keys (VLKs), one VLK per physical line key per tab. Tabs display across the top left and right of the screen, with the active tab highlighted. The VVX 150 doesn't support virtual line keys.

These models support four tabs, providing a total of 48 VLKs on the VVX 450, 24 on the VVX 350, and 16 on the VVX 250. The VVX 150 doesn't support tabs.

On the VVX 450, the VLKs are further listed and numbered onscreen as **Left Line Keys** (LLK) for the line keys on the left side of the display, and simply **Line Keys** (LK) for the line keys on the right side of the display. On the VVX 250 and VVX 350, which have only one row of physical line keys on the left, the VLKs are all labeled **Left Line Keys** (LLK).

The virtual line keys are numbered and grouped into tabs as follows:

**Virtual Line Key Labels on VVX 250, VVX 350, and VVX 450**

VLK Groups within Tabs	Tab 1	Tab 2	Tab 3	Tab 4
VVX 250 4 LKs = 16 VLKS	LLK1 – LLK4	LLK5 – LLK8	LLK9 – LLK12	LLK13 – LLK16
VVX 350 6 LKs = 24 VLKS	LLK1 – LLK6	LLK7 – LLK12	LLK13 – LLK18	LLK19 – LLK24
VVX 450 12 LKs = 48 VLKS (2 groups of 6 LKs)	LK1 – LK6 LLK1 – LLK6	LK7 – LK12 LLK7 – LLK12	LK13 – LK18 LLK13 – LLK18	LK19 – LK24 LLK19 – LLK24

## Line Key Window Customization

**Line Key Window dimensions**



A Line Key Window (also known as Screen Tile) is a 160Wx34H-pixel area next to each Line Key on the screen. As shown in the picture above, this window is divided into three nonoverlapping regions: Text Line 1, Text Line 2, and Icon. The information shown in the window depends on the function assigned to the corresponding line key (which is a feature key) and the current state of the assigned function.

You can customize the information shown on Text Line 1 and 2 for the following feature key functions with the parameters in the following table:

- Call Appearance
- Speed Dial
- Busy Lamp Field
- Action URL

**Line Key Window Customization Parameters**

Parameter Group	Parameter	Description
<i>Call Appearance</i>		
<i>IP Phone – Feature Key Customization – Call Appearance</i>	<b>Enable</b>	Enables using customized display for Call Appearance feature keys.

## Line Key Window Customization Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine1	Textual information to show on Text Line 1 area during idle.
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine2	Textual information to show on Text Line 2 area during idle.
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine1InCall	Textual information to show on Text Line 1 area during a call.
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine2InCall	Textual information to show on Text Line 2 area during a call.
<b>Busy Lamp Field</b>		
<i>IP Phone – Feature Key Customization – Busy Lamp Field</i>	Enable	Enables using customized display for Busy Lamp Field feature keys.
<i>IP Phone – Feature Key Customization – Busy Lamp Field</i>	TextLine1	Textual information to show on Text Line 1 area.
<i>IP Phone – Feature Key Customization – Busy Lamp Field</i>	TextLine2	Textual information to show on Text Line 2 area.
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine1InCall	Textual information to show on Text Line 1 area when the monitored extension has one or more calls.
<i>IP Phone – Feature Key Customization – Call Appearance</i>	TextLine2InCall	Textual information to show on Text Line 2 area when the monitored extension has one or more calls.
<b>Speed Dial</b>		
<i>IP Phone – Feature Key Customization – Speed Dial</i>	Enable	Enables using customized display for Speed Dial feature keys.
<i>IP Phone – Feature Key Customization – Speed Dial</i>	TextLine1	Textual information to show on Text Line 1 area.
<i>IP Phone – Feature Key Customization – Speed Dial</i>	TextLine2	Textual information to show on Text Line 2 area.
<b>Action URL</b>		

## Line Key Window Customization Parameters (continued)

Parameter Group	Parameter	Description
<i>IP Phone – Feature Key Customization – Action URL</i>	<b>Enable</b>	Enables using customized display for Action URL feature keys.
<i>IP Phone – Feature Key Customization – Action URL</i>	<b>TextLine1</b>	Textual information to show on Text Line 1 area.
<i>IP Phone – Feature Key Customization – Action URL</i>	<b>TextLine2</b>	Textual information to show on Text Line 2 area.
<b>Group <i>n</i> (<i>n</i> = 1, 2, 3, 4)</b>		
<i>IP Phone – Feature Key Customization – Group <i>n</i></i>	<b>ID</b>	The ID to reference this Group.
<i>IP Phone – Feature Key Customization – Group <i>n</i></i>	<b>TextLine1</b>	Textual information to show on Text Line 1 area.
<i>IP Phone – Feature Key Customization – Group <i>n</i></i>	<b>TextLine2</b>	Textual information to show on Text Line 2 area.
<i>IP Phone – Feature Key Customization – Group <i>n</i></i>	<b>Icon</b>	Internal icon path or HTTP/HTTPS URL.

You can use macros to specify the textual contents for Text Line 1 and Text Line 2 areas. You must precede a macro with a \$, such as \$number or \$(number). The enclosing parenthesis is required when a character that is a legal macro character follows the macro. A legal macro character is one of {a-z A-Z . \_ 0-9}.

The following macros are available.

## Macros for Line Key Window Customizations

Macro	Description	Notes
\$name	The <b>Name</b> field as configured for the feature key.	
\$number	The <b>Number</b> field as configured for the feature key.	
\$service	The default name of the service specified in the <b>Service</b> field for the feature key: SP1, ..., SP6, OBiTALK.	Expands into the word <i>Auto</i> if service field isn't specified.
\$service.displayLabel	The X_DisplayLabel parameter per SP <i>n</i> service or the <b>DisplayLabel</b> parameter per OBiTALK service.	{Blank} if service field isn't specified.
\$service.number	For SP1 – SP6, this is the User ID of the SIP account, which is derived from the <i>AuthUserName</i> or <i>URI</i> parameter of the SP service. For OBiTALK, this is the OBi number of the phone.	{Blank} if service field isn't specified.



**Macros for Line Key Window Customizations (continued)**

Macro	Description	Notes
<code>\$service.displayNumber</code>	The <code>X_DisplayNumber</code> parameter per <code>SPn</code> service or the <code>DisplayNumber</code> parameter per <code>OBiTALK</code> service.	{Blank} if service field isn't specified.
<code>\$func</code>	Short internal name of the function assigned to the key: <ul style="list-style-type: none"> <li>• <code>call</code> for Call Appearance</li> <li>• <code>blf</code> for Busy Lamp Field</li> <li>• <code>spd</code> for Speed Dial</li> <li>• <code>acturl</code> for Action URL</li> </ul>	{Blank} if function isn't specified.
<code>\$funcname</code>	Long internal name of the function assigned to the key: <ul style="list-style-type: none"> <li>• <code>Call</code> for Call Appearance</li> <li>• <code>Busy Lamp</code> for Busy Lamp Field</li> <li>• <code>Speed Dial</code> for Speed Dial</li> <li>• <code>XML APP</code> for Action URL</li> </ul>	{Blank} if function isn't specified.
<code>\$call.service</code> and <code>\$call.service.x</code> where <code>x</code> = number, <code>displayLabel</code> , or <code>displayNumber</code>	Same as <code>\$service</code> and <code>\$service.x</code> applying to the service the call is on.	{Blank} if service isn't defined for the call yet.
<code>\$call.number</code>	Call peer's Number	
<code>\$call.name</code>	Call peer's Name	
<code>\$value</code>	For BLF, this is number of calls currently active on the monitored entity. The value is {blank} for other functions.	
<code>\$blf.ext</code>	The extension of the entity monitored by the feature key.	
<code>@id</code>	Line Key Hard Key ID. Expands into: <ul style="list-style-type: none"> <li>• 1,2 for VVX 150, VVX 250</li> <li>• 1,2,3 for VVX 350, VVX 450</li> </ul>	
<code>@p</code>	Line Key Page. Expands into: <ul style="list-style-type: none"> <li>• 1 for VVX 150, VVX 250</li> <li>• 1,2,3,4 for VVX 350, VVX 450</li> </ul>	
<code>@n</code>	Line Key ID. Expands into: <ul style="list-style-type: none"> <li>• 1,2 for VVX 150</li> <li>• 1 – 16 for VVX 250</li> <li>• 1 – 24 for VVX 350</li> <li>• 1 – 48 for VVX 450</li> </ul>	

**Macros for Line Key Window Customizations (continued)**

Macro	Description	Notes
\$eval	See the <i>Polycom OBiPhoneXML Applications API Guide</i> .	
@text1	The default information displayed on the (Text Line 1) section of the Line Key Window. Applicable in the <code>TextLine1</code> , <code>TextLine2</code> , <code>TextLine1InCall</code> , and <code>TextLine2InCall</code> parameters.	
@text2	The default information displayed on the (Text Line 2) section of the Line Key Window. Applicable in the <code>TextLine1</code> , <code>TextLine2</code> , <code>TextLine1InCall</code> , and <code>TextLine2InCall</code> parameters.	
@icon	The default icon displayed on the (Icon) section of the Line Key Window. Applicable to the <code>Icon</code> parameter.	

## Example

In this example, the information that normally displays on line 1 and line 2 of all Call Appearance Line Keys is reversed. In addition, during a call, "--" is added to the beginning of line 1. Below are the settings for this example.

**Call Appearance Line Key Settings**

Parameter Group	Parameter	Value
Call Appearance		
<b>IP Phone – Feature Key Customization – Call Appearance</b>	<b>Enable</b>	true (that is, the option is checked)
<b>IP Phone – Feature Key Customization – Call Appearance</b>	<b>TextLine1</b>	@text2
<b>IP Phone – Feature Key Customization – Call Appearance</b>	<b>TextLine2</b>	@text1
<b>IP Phone – Feature Key Customization – Call Appearance</b>	<b>TextLine1InCall</b>	-- @text2
<b>IP Phone – Feature Key Customization – Call Appearance</b>	<b>TextLine2InCall</b>	@text1

## Line Key Tabs

By default, the line keys area to the right of the screen has four tabs of VLKs, and only one tab is visible at any time. You can disable this feature using the option **User Preferences::LineKeyTabs**. When disabled, the phone shows no line key tabs in the status bar and only one page of VLKs is available (none on the VVX

150, VLK 1-4 on the VVX 250, and VLK 1-6 on the VVX 350 and VVX 450). The status icons also display in the right corner. To handle more calls in this case, you may enable multiple calls per Call (Appearance) Key as described in the next section.

## Calls App Behavior

The Calls App shows a list of current calls on the phone. Calls controlled under the same call key are grouped together in the list with calls on call keys with a smaller VLK index shown first. New calls are added to the beginning of the call group of the same call key.

The phone can only highlight one call on the Calls App screen, and the corresponding VLKW shows a white bounding box to identify the call key the highlighted call belongs to. As a user navigates through the calls on the screen, the white bounding box moves accordingly.

Normally, the calls display together on the Calls App screen. Hence it's possible to see two calls belonging to the same call key on the same screen. If this isn't desirable, you can disable the option **User Preferences::PackCallsOnDisplay**, which causes the phone to display only the calls belonging to the same call key.

## IP Phone Settings

### Phone Settings

Settings are divided into the following groups:

- Phone Settings
- Line Keys
- Programmable Keys
- Sidecar 1 and Sidecar 2

### DigitMap and OutboundCallRoute

The **DigitMap** controls what number you can dial and applies the given transformation to the dialed number. It can refer to the `DigitMap` parameter values in other parameter groups for better readability and organization. The `OutboundCallRoute` determines which service to use based on the dialed number, after validation and transformation by the **DigitMap**.

### Primary Line

You can select the Primary Line for the Phone and for the AA, respectively, using the **Phone Settings::PrimaryLine** and **Auto Attendant::PrimaryLine** parameters. The primary line is the default line to use when there is no explicitly selected line and no line-selection prefix (line access code) has been dialed. For example, when going off-hook to get Dialtone, the phone tries to allocate a call key that is bound to the primary line for the call, if one is available.

You can make one of the SP Services, OBiTALK or TG1/TG2 as the Primary Line for outbound calls. The Primary Line for the Phone and the Auto Attendant is configured separately. The following list summarizes the choices available for selection as the primary line:

- SP1 Service
- SP2 Service
- SP3 Service
- SP4 Service
- SP5 Service
- SP6 Service
- OBiTALK Service
- Trunk Group 1
- Trunk Group 2

You can select a specific Line to use when making a call explicitly by pressing a call key or line monitor key bound to that line or a softkey corresponding to that line. You can also dial a Line's access code before the destination number. The default service route access codes are defined as:

- \*\* 1 for SP1
- \*\* 2 for SP2
- \*\* 3 for SP3
- \*\* 4 for SP4
- \*\* 9 for OBiTALK

Service route access codes for calling from the Phone can be customized if necessary by modifying **Phone Settings::DigitMap** and **Phone Settings::OutboundCallRoute**. Service route access codes for calling via the Auto Attendant can be customized if necessary by modifying **Auto Attendant::DigitMap** and **Auto Attendant::OutboundCallRoute**.



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.

## Network Directory

The **Enable** option in this group is for the enabling and displaying the Network Directory option on the Main menu. The **VoiceService** option determines which SP service's network directory function to invoke when the Main menu option is selected by the user.



Only one Network Directory option can be shown on the Main menu. The `Name` parameter is reserved for future use.

## Buddy List

The **Enable** option in this group is for the enabling and displaying of the Buddy List option on the Main menu. The **VoiceService** option determines which SP service's buddy list function to invoke when the Main menu option is selected by the user.



Only one Buddy List option can be shown on the Main menu. The `Name` parameter is reserved for future use.

## User Preferences Settings

The **CallForwardUnconditionalFeatureProvider** determines which **CallForwardUnconditional** parameter the Call Forward option under User Preferences should control. Note that this is the same setting the **Call Forward** softkey (in the Home Screen) controls. There is the Phone version and also one version for each voice service. If an *SPn* service is selected and if the **SPn – Network Provided Services::CallForwardAlways** is `true`, then the user preferences and softkey option reflects and controls the setting of the feature at the server side.

The **DoNotDisturbFeatureProvider** determines which **DoNotDisturb** parameter the Do Not Disturb option under User Preferences should control. There is the Phone version and also one version for each voice service. If an *SPn* service is selected and if the **SPn – Network Provided Services::DoNotDisturb** is `true`, then the user preferences option reflects and controls the setting of the feature at the server side.

## Page Groups 1 and 2

**GroupName** is a nickname to refer to the page group; not used anywhere at the moment.

**MulticastAddress** and **MulticastPort** define the multicast address of the group to join, and **TTL** sets the TTL value of the outgoing multicast packets. **ParticipantName** is a name to identify to the group the user of this phone via RTCP messages.

To use a page group effectively, there must be a feature key assigned with the corresponding page group function. You can press the key once to talk to the group, or use PTT to talk if the **PushToTalk** option is also enabled in the page group configuration.

## Line Keys

This group is used for the configuration of the Line Keys as Feature Keys.

There are twelve (12) physical Line Keys (LKs) on the VVX 450, six (6) on the VVX 350, four (4) on the VVX 250, and two (2) on the VVX 150. On the VVX 450, the line keys are split half and half to the left and right of the LCD display. On the VVX 350, VVX 250, and VVX 150, the line keys are on the left side of the display.

# Network

## SIP Registration

You can set up your phones to periodically register with the **ProxyServer** or the **RegistrarServer** by enabling the **SPn Service::X\_RegisterEnable** parameter. **ProxyServer** and **RegistrarServer** could be different, although they're rarely so in practice. **ProxyServer** is a required parameter that you must configure on your phone while **RegistrarServer** is optional and assumed to be the same as the **ProxyServer** if not specified in the configuration.



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

The main purpose of 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. Each phone takes only one local IP address that is either statically assigned in the phone's configuration or dynamically obtained from a local DHCP server. The method to get an IP address assigned is determined by the value of **WAN Settings::AddressingType**.

The **SPn** services for  $n = 1 - 6$  each use a different local contact port for sending and receiving SIP messages (the default is 5060, 5061, 5062, ..., and 5065 respectively). You can configure this port in the **SPn Service::X\_UserAgentPort** parameter. **ProxyServer** and **RegistrarServer** must use the same transport protocol for SIP messages that you can set in the **ProxyServerTransport** parameter.

Your phones support UDP, TCP, and TLS for SIP transport. The default server port is 5060 for UDP/TCP and 5061 for TLS. When using TCP/TLS, the phones initiate a TCP/TLS connection only with the **ProxyServer**. All subsequent SIP messages are exchanged between the phones, and the servers must use the same connection. If for any reason the connection is closed, your phone attempts to re-establish the connection following an exponential back-off retry pattern.



Dynamic address binding through periodic registration isn't strictly 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.

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. Note that your phone also includes the +sip-instance parameter in the Contact header that specifies your phone's MAC address in the UUID. You can suppress this parameter by disabling the **ITSP Profile X – SIP::X\_RegisterIncludeInstance** option.

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. To cope with such a case, tell your phone to try other SIP user agent ports by specifying a comma-separated list of as many as 10 alternative ports in the **SPn Service::X\_UserAgentPorts** parameter. Your phone then cycles through those ports to retry REGISTER until it receives a response from the server.

## Third-Party Registration

You can configure your phone to use a third-party registration that registers it for an Address of Record (AOR) that isn't the same as the account user-id. That is, the user-id in the TO header of the SIP REGISTER request is different from that in the FROM header, which always carries the account user-id.

One application is in the implementation of a shared line using the Bridged Line Appearance (BLR) method. To enable third-party registration, set the user-id to register for in the **SPx Service – SIP Credentials::X\_ShareLineUserID** parameter.

## Registration Period

You can configure the nominal registration Expires header value (implemented as a Contact header parameter value in seconds) used by your phone in REGISTER requests using the **ITSP Profile X – SIP::RegistrationPeriod** parameter. The server determines the actual Expires value. The server may reject the REGISTER request with 423 with a Min-Expires header value (in seconds). If that happens, your phone quickly retries with an Expires header value equal to the Min-Expires header value from the server. When the server accepts the registration, it replies with a 2xx response for the REGISTER and includes an Expires parameter value in the Contact header that matches the Contact your phone uses in the REGISTER request. However, if it isn't found in the Contact, your phone takes the server-supplied Expires value from the Expires header of the 2xx response. If still not found, your phone assumes the server-supplied value is 3600 seconds.

If the server-supplied Expires value is less than the Expires header value used by your phone, it takes the server version to compute the next renewal interval. Otherwise, your phone uses its own Expires header value to do the same.



The server shouldn't supply explicitly or implicitly an Expires value that is larger than what your phone has asked for, as that would be a protocol violation. Your phone, however, ignores such an error.

Your phone computes the next renewal time by subtracting a percentage of the Expires value derived from the 2xx response returned by the server. Use the **ITSP Profile – SIP::X\_RegistrationMargin** parameter to control how the margin to subtract is computed. For example: If `X_RegistrationMargin` is 0 or not specified, the renewal time is half-way before the expiration, if the Expires value is less than 1200s, or 600s before the expiration otherwise. If `X_RegistrationMargin`  $\geq 1$ , it's interpreted as the number of seconds (with any fractional part dropped) before the expiration time to renew registration. If

`X_RegistrationMargin`  $< 1$ , it's interpreted as the fraction of the current expires value to subtract from the expiration time to get the time for the next renewal. For example, if `X_RegistrationMargin` = 0.01 and the Expires value is 300, then the next renewal goes out 3 seconds before the expiration time.

## REGISTER Final Non-2xx Response Handling

When registration encounters an error, your phone can schedule retries based on the type of error. Each recognizable error type is represented by a 3-digit code. Error codes 300 – 699 are the error response codes returned by the server, while 900-999 are used to indicate other errors. The following 9xx error codes are related to registration:

- 900 = Timeout waiting for a response from the server
- 901 = Cannot resolve the server name or the host is not reachable

For 3xx class responses with a valid Contact header, your phone follows the given Contact to retry registration quickly. If a valid Contact is not found, or if the number of consecutive redirections has reached 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 is not present in the response or the value is not 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 handling is by waiting for a certain number of seconds before trying to register again. The number of seconds to wait is determined by the rules specified in the **ITSP Profile X – SIP::RegisterRetryResponseCodes** parameter on the actual error code. The format of this parameter is the same as a digit map. Let's consider the default value of this parameter:

```
(<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. The fixed default value is configured in the **ITSP Profile X – SIP::RegisterRetryInterval** parameter. 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.

## SIP Outbound Proxy Server

An **OutboundProxy** server can be configured on your phone such that all outbound requests are sent via the outbound proxy server instead of directly to the **ProxyServer** or **RegistrarServer**. If the outbound proxy server is listening at a non-standard port, the correct port value must be specified in the `OutboundProxyPort` parameter. The **OutboundProxy** may use a different transport protocol from the **ProxyServer**. The transport protocol to use to communicate with the **OutboundProxy** can be set in the `OutboundProxyTransport` parameters. If `OutboundProxyTransport` is TCP or TLS, your phone initiates a TCP or TLS connection only with the **OutboundProxy**. All subsequent messages exchanged between your phone and the servers **MUST** use the same connection. If for any reason the connection is closed, your phone attempts to re-establish the connection with the **OutboundProxy** following an exponential back-off retry pattern.

Even though your phone only exchanges messages directly with the **OutboundProxy**, the `ProxyServer`, `ProxyServerPort`, and `ProxyServerTransport` parameters are still very much relevant and important since the SIP requests sent by your phone to the server are still formed based on these values, not based on the **OutboundProxy** value. In fact, the **OutboundProxy** value should never appear in the SIP requests generated by your phone (unless the **OutboundProxy** has the same value as the **ProxyServer**).

Some server implementations include the outbound proxy server in a Record-Route header such that your phone should not respect the locally configured **OutboundProxy** value after the initial INVITE is sent for a new call. This behavior can be achieved by enabling the option **ITSP Profile X –**

**SIP::X\_BypassOutboundProxyInCall**. 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.

## DNS Lookup of SIP Servers

When sending out SIP requests to the server, your phone looks up the IP address of the server, using DNS query if the server is specified as a domain name instead of an IP address. If an Outbound Proxy Server is configured, it is used instead of the SIP Proxy Server or SIP Registration Server. The resolution of the server domain name into IP address is performed in the following manner:

- If **ITSP Profile X – SIP::X\_DnsSrvAutoPrefix** is enabled, resolve the name as DNS A Record, DNS SRV Record and as DNS SRV record with a service prefix prepended to the name in parallel by sending 3 queries to each DNS server at the same time. The service prefix to prepend to the name depends on the transport protocol being used; for SIP, `_sip._udp.` for UDP, `_sip._tcp.` for TCP and `_sip._tls.` for TLS. If more than one valid result is returned from the queries, the DNS SRV result for the name with prefix has the highest priority, then the DNS SRV result for the name without the prefix, then the DNS A result.
- Otherwise, resolve the name as a DNS A record and as a DNS SRV Record in parallel by sending 2 queries to each DNS server. If both queries return a valid result, the DNS SRV result is taken over the DNS A result.
- If no valid results are returned, your phone considers the SIP request failed with the error code 901.

If the result from the DNS query is an SRV record, the server port is also taken from that record (the server port value configured on your phone is ignored). Otherwise, the server port is taken from the configured

value. If no value is specified, 5060 is used. We recommend setting the **ProxyServerPort** to 0 (that is, the unspecified value) if DNS SRV lookup is intended for the service.

## NAT Traversal Considerations

If your phone is located behind NAT with respect to the service provider equipment, it can discover the mapped external address corresponding to its local SIP contact address as seen by the server. This may help in some cases where a gateway router's SIP ALG implementation is causing communication problems between your phone and the server. Your phone can discover the mapped external address in one of the following ways:

- From the “received=” and “rport=” parameters of the VIA header of the REGISTER response sent by the server. These two parameters tell your phone its mapped IP address and port number, respectively. This method is used if periodic registration is enabled on your phone.
- From the response to a STUN binding request your phone sent to a STUN server. This method is used by enabling **X\_KeepAliveEnable** and setting the `X_KeepAliveMsgType` parameter to “stun”. The keep-alive messages are sent to the same server where a REGISTER request would be sent to.
- From the value of the **ITSP Profile X – SIP::X\_PublicIPAddress** parameter.

If discovered by one of the methods above, your phone uses the discovered external IP address and port to replace its private address and port when generating SIP requests when the **ITSP Profile X – SIP::X\_DiscoverPublicAddress** option is also enabled. The substitution of private addresses with public addresses applies to the Contact header of any SIP requests and the `c=` line in SDP. If the option **X\_UsePublicIPAddressInVia** is also enabled, the Via address is also substituted. However, this usually isn't necessary.

Your phone can also include an empty `rport` parameter in the Via header of outbound SIP requests if the option **ITSP Profile X – SIP::X\_UseRport** option is enabled. This parameter is sometimes needed to prompt the server to insert an `rport` parameter value in the response. It should also prompt the server to send the response to the port where the request originated from (that is, according to the source port of the IP header of the packet). However, as such behavior on the server is considered standard by many, the empty `rport` parameter has become superfluous in practice.

## Keep Alive Messages

In addition to registration with the server, you can configure your phone to send out periodic keep alive messages on the same network path to keep the NAT pinhole open. Poly recommends that you send the keep alive messages to the same proxy server responsible for registration.

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

Use the following parameters to configure keep alive messages:

## Keep Alive Messages

Parameter Group	Parameter	Description
SPn Service	X_KeepAliveEnable	Enable sending of keep alive message. If set to <code>true</code> , your phone sends periodic keep-alive messages to the destination specified in <b>X_KeepAliveServer</b> and <b>X_KeepAliveServerPort</b> , at the interval specified in <b>X_KeepAliveExpires</b> . The content of this message is the ASCII string <code>keep-alive\r\n</code>
SPn Service	X_KeepAliveExpires	Keep alive period in seconds.
SPn Service	X_KeepAliveMsgType	The type of keep alive messages to send out periodically if <code>keep-alive</code> is enabled. One of the following: <ul style="list-style-type: none"> <li><code>keep-alive</code>: The string <code>keep-alive</code> for SIP transport over UDP, or CRLF (e.g., carriage return and line feed characters) for SIP transport over TCP or TLS.</li> <li><code>empty</code>: A blank line</li> <li><code>stun</code>: A standard STUN binding request. Your phone uses the binding response to form its contact address for REGISTRATION.</li> <li><code>custom</code>: use the value of <b>X_CustomKeepAliveMsg</b></li> <li><code>options</code>: a valid SIP OPTIONS request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled.</li> <li><code>notify</code>: a valid SIP NOTIFY request. No response to this request from the server triggers a proxy failover if proxy redundancy is enabled.</li> </ul>
SPn Service	X_CustomKeepAliveMsg	Defines the custom message to be used when <b>X_KeepAliveMsgType</b> is <code>custom</code> . The value should have the following format: <code>mtd=NOTIFY;event={whatever};user={anyone}</code> Where <ul style="list-style-type: none"> <li><code>NOTIFY</code> may be replaced by any other SIP method, such as <code>PING</code>, the <code>event</code> parameter is optional and is only applicable if the method is <code>NOTIFY</code>.</li> <li>If <code>event</code> is not specified, the 'keep-alive' event is used with <code>NOTIFY</code>.</li> <li>The <code>user</code> parameter is optional. If not specified, the <code>request-uri</code> won't have a <code>userid</code>, and the <code>TO</code> header field uses the same <code>userid</code> as the <code>FROM</code> header, which is the local account <code>userid</code>. If <code>user</code> is specified, it is used as the <code>userid</code> in the <code>Request-URI</code> and <code>TO</code> header.</li> </ul> SIP messages for <code>keep-alive</code> are sent only once without retransmission. Response to the SIP messages are ignored by the phone.

## ***SIP Proxy Server Redundancy and Dual REGISTRATION***

Server Redundancy specifically refers to your phone's capability to:

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

As such, device registration must be enabled to use the server redundancy feature. Other SIP requests, such as INVITE or SUBSCRIBE, are sent to the same server that your phone currently registers with.

If the Outbound Proxy Server is provided, server redundancy is applied to the Outbound Proxy Server instead of the REGISTRATION server. Server redundancy behavior is enabled by enabling the ***ITSP Profile X – SIP::X\_ProxyServerRedundancy*** parameter (which is disabled by default).

Another requirement for using the server redundancy feature is that the underlying server must be configured in your phone as a domain name instead of an IP address. This allows your phone to collect a list of candidate servers based on DNS query. The domain name may be looked up as DNS A record or DNS SRV record. For A records, all the IP addresses returned by the DNS server are considered to have the same priority. For SRV records, the hosts returned by the DNS server can be each assigned a different priority.

After a list of candidate servers is obtained, your phone first looks for a working server according to the stated priority. A working server means one that your phone can successfully register with. This is known as the Primary Server. Subsequently, your phone maintains registration with the primary server the usual way. However, if no working server is found after traversing the entire list, your phone takes a short break and repeats the search in the same order.

While maintaining registration with the Primary Server, your phone continually attempts to fall back to one of the candidate servers that has higher priority than the primary server, if any. The list of candidate servers that your phone is trying to fall back on is known as the primary fallback list, which may be empty.

In addition, your phone can be configured to maintain a secondary registration with a server that has lower or equal priority than the primary server. Secondary registration can be enabled by setting the `X_SecondaryRegistration` parameter to `true`. If `X_ProxyServerRedundancy` is `false`, however, `X_SecondaryRegistration` does not have any effect. If this feature is enabled, as soon as a primary server is found, your phone searches for a working secondary server in the same manner from the list of candidate servers that are of lower or equal priority than the primary server. Similarly, once a secondary server is found, your phone forms a secondary fallback list to continually attempt to fall back on if the list is not empty.

The interval for checking the primary fallback list and the secondary fallback list are configured in the `X_CheckPrimaryFallbackInterval` and `X_CheckSecondaryFallbackInterval` parameters, respectively. These parameters are specified in seconds and the default value is 60 for both.



- Secondary server exists implies primary server exists.
- If the secondary server exists, it immediately becomes the primary server when the current primary server fails. Your phone then starts searching for a new secondary server if the candidate set is not empty.
- The candidate list may change (lengthened, shortened, priority changed, and so forth) on every DNS renewal (based on the entry's TTL). Your phone rearranges the primary and secondary servers and fallback lists accordingly, whichever is applicable.

If the server redundancy feature is disabled, your phone resolves only one IP address from the server's domain name and won't attempt to try other IP addresses if the server is not responding.

## DNS NAPTR Support

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

The following list describes the basic NAPTR feature:

- You can enable 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 (on the corresponding *ITSP Profile x – SIP* web page).  
The phone doesn't fall back to perform regular DNS SRV or DNS A record lookup if NAPTR records aren't available.
- You can use NAPTR lookup whether or not you enable the `X_ProxyServerRedundancy` parameter, although best practices are to enable the `X_ProxyServerRedundancy` parameter with NAPTR lookup.
- If a NAPTR record indicates to look up the host name as an SRV record, the phone considers the host name to be complete and doesn't add a prefix when querying the DNS server for its SRV record, regardless the settings of the `X_DnsSrv` and `X_DnsSrvAutoPrefix` parameters.
- When more than one UDP, TCP, or TLS options are available and all are at the same order and preference, the phone considers UDP the highest priority, then TCP, then TLS.
- With NAPTR enabled, the phone can failover and fallback among UDP/TCP/TLS transports or change to a different transport based on the result of each NAPTR lookup (which is renewed at half of TTL value from the DNS server).

## SIP Privacy

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

If PAID is present, your phone takes the name and number from it. Otherwise, it takes name and number from RPID if it is present, or from the `FROM` header otherwise. RPID, if present, includes the privacy setting desired by the caller. The privacy setting may indicate one of the following options:

- `off` = No privacy requested. Your phone shows name and number.
- `full` = Full privacy requested. Your phone hides both name and number.
- `name` = Name privacy requested. Your phone shows the number but hides the name.
- `uri` = URI privacy requested. Your phone shows the name but hides the number.

Regardless, if PAID exists or not, your phone always takes the privacy setting from the RPID if it is present in the INVITE request.



If the resulting caller name is “Anonymous” (case-insensitive), your phone treats it as if the caller is requesting full privacy.

For outbound calls, the caller’s preferred privacy setting can be stated by phone in an `RPID` header of the outbound `INVITE` request. To enable this behavior, the **ITSP Profile X – SIP::X\_InsertRemotePartyID** parameter must be set to `true`, which is the default value of this parameter. Your phone supports only two outbound caller privacy settings: `privacy=off` or `privacy=full`. The `RPID` header generated by your phone carries the same name and number as the `FROM` header. If outbound caller-ID is blocked, phone sets `privacy=full` in `RPID` and also sets the display name in the `FROM` and `RPID` headers to “Anonymous” for backward compatibility. Your phone won’t insert `PAID` in outbound `INVITE` requests. You can further instruct your phone to use `sip:anonymous@localhost` in the `FROM` header by enabling the option **X\_UseAnonymousFROM** (that is, in this case your phone uses `From: “Anonymous” <sip:anonymous@localhost>`).

Your phone also includes a `Privacy: id` header if **X\_InsertPrivacyHdr** is also enabled.

## STUN and ICE

Your phone supports standard STUN based on RFC3489 and RFC5389 for passing inbound RTP packets to phone when behind NAT. The parameters that control the STUN feature can be found under the section **ITSP Profile X – General**:

- **STUNEnable** – Enable this feature (default is `false`).
- **STUNServer** – The IP address or domain name of the external STUN server to use. STUN feature is disabled if this value is blank, which is the default.
- **X\_STUNServerPort** – The STUN Server’s listening UDP port. Default value 3478 (standard STUN port).

It should be noted that the STUN feature used in this context is only for RTP packets, not SIP signaling packets (which typically do not require STUN). Your phone sends out a STUN binding request right before making or answering a call on SPx. If the request is successful, your phone decodes the mapped external address and port from the binding response and uses them in the `m=` and `c=` lines of its SDP offer or answer sent to the peer device. If the request fails, such as STUN server not found or not responding, the call goes on without using an external address in the SDP.

Standard RTP requires the use of even numbered ports in the `m=` line. If the external port is not an even number, your phone changes the local RTP port, retries STUN, and continues to do this as many as four times or until an even-numbered external port is found. If the fourth trial still results in an odd-numbered external port number, the call goes on without using external address in the SDP.

Your phone supports standard ICE based on RFC5245. ICE is done on a per-call basis to automatically discover which peer address is the best route for sending RTP packets. To enable ICE on your phone, set the **ITSP Profile X – General::X\_ICEEnable** parameter to `yes` (or `true`). The default is `no` (or `false`).

Note that ICE is more effective if STUN is also enabled. However, STUN not a requirement for using ICE on your phone. If STUN is enabled and an external RTP address different from its local address is discovered, your phone offers two ICE candidates in its SDP:

- The local (host) address (highest priority)
- The external (`sflx` or server reflexive) address

Otherwise only the local host candidate is shown in your phone's SDP.



Your phone uses the srflx address in the m= and c= lines of the SDP if STUN is enabled and successful.

If ICE is enabled and the peer's SDP has more than one candidate, your phone sends STUN requests to each peer candidate from its local RTP port. As soon as it receives a response from the highest priority candidate, your phone concludes ICE and uses this candidate to communicate with the peer subsequently. Otherwise your phone allows as long as 5 seconds to wait for a response from all candidates and selects the highest priority one that responds. Once ICE is completed successfully, your phone further applies the symmetric RTP concept to determine the peer's RTP address (that is, sends to the address where the peer's RTP packets are coming from).

## ***ITSP Driven Distinctive Ringing***

Your phone offers 10 ring tones and 10 call waiting tone patterns in each ring profile. These patterns are numbered from 1 to 10. Each pattern also comes with a configurable name. You can assign a different default ring tone to each trunk on your phone.

An ITSP can instruct your phone which ring tone 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 ring tone name or call waiting tone name in the ring tone profile that matches the Alert-Info URI. Ring tone names are not case sensitive when the phone compares them. If the phone finds a match, your phone plays the corresponding ring tone or call waiting tone. Otherwise, your phone plays the default ring tone.

## ***RTP Statistics – the X-RTP-Stat Header***

When ending an established call, your phone 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 reported fields are:

- PS = Number of Packets Sent
- PR = Number of Packets Received
- OS = Number of bytes sent
- OR = Number of bytes received
- PL = Number of packets lost
- JI = Jitter in milliseconds
- LA = Decode latency or jitter buffer size in milliseconds
- DU = Call duration in seconds
- EN = Last Encoder Used
- DE = Last Decoder Used

For example:

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

To enable the X-RTP-Stat feature, the *ITSP Profile X – SIP::X\_InsertRTPStats* parameter must be set to `true`.

## RTCP

Your phone supports RTCP (RFC 3550) and RTCP-XR (RFC3611) with MOS statistics for VQ reporting.

## Media Loopback Service

Your phone supports the media loopback draft as described in draft-music-media-loopback-13.txt. Your phone supports the following media loopback features:

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

When acting as a loopback mirror, your phone always sends primer packets so that incoming packets can get through NAT or a firewall. The media loopback feature is controlled by the following parameters (under the **Phone Settings – Calling Features** section):

- `AcceptMediaLoopback` – Enable your phone to accept incoming calls that request media loopback. Default is YES.
- `MediaLoopbackAnswerDelay` – The delay in milliseconds before your phone answers a media loopback call. Default is 0.
- `MediaLoopbackMaxDuration` – The maximum duration to allow for an incoming media loopback call. Default is 0, which means the duration is unlimited.



Your phone rejects an incoming media loopback call if:

- Phone is off hook.
- Phone is ringing.
- One or more calls are on hold.

Your phone terminates an inbound media loopback call already in progress when:

- Phone is off-hook.
- Phone is ringing.

To make an outgoing loopback call, dial one of the following star codes before dialing the target number:

- \*03 – Make a Media Loopback Call.
- \*04 – Make a RTP Packet Loopback Call.



Outbound Media Loopback Call is not subject to a call duration limit. It lasts until you hang up or until the called number ends the call.



## A SIP/SP Configuration Example

The following table details a configuration example where the ITSP Profile to use is B with two SP services, SP1 and SP3, both pointing to the same ITSP Profile.

### Configuration Example

Parameter Group	Parameter	Example Value	Notes
<i>ITSP Profile B – General</i>	<b>SignalingProtocol</b>	SIP	Standard default value is SIP
<i>ITSP Profile B – SIP</i>	<b>ProxyServer</b>	sip.phonepower.com	
<i>ITSP Profile B – SIP</i>	<b>OutboundProxy</b>	sbc.phonepower.com	
<i>ITSP Profile B – SIP</i>	<b>OutboundProxy</b>		
<i>SP1 Service</i>	<b>X_ServProvProfile</b>	B	
<i>SP1 Service – Credentials</i>	<b>AuthUserName</b>	3409991003	
<i>SP1 Service – Credentials</i>	<b>AuthPassword</b>	i9cik#dkL	
<i>SP1 Service – Credentials</i>	<b>X_MyExtension</b>	1003	
<i>SP3 Service</i>	<b>X_ServProvProfile</b>	B	
<i>SP3 Service – Credentials</i>	<b>AuthUserName</b>	3409991005	
<i>SP3 Service – Credentials</i>	<b>AuthPassword</b>	c98dfa74{	
<i>SP3 Service – Credentials</i>	<b>X_MyExtension</b>	1005	
<i>Phone Settings</i>	<b>PrimaryLine</b>	SP3	

# Call Routing

---

Call Routing is the process by which your phone sets up a call bridge or an endpoint call based on such information as the trunk on which the call originates, the caller's number, the called number, and so on. Call Routing Rules are parameters used to instruct your phone how to route calls. A call may transform into a call bridge or an endpoint call after being routed by your phone according to the given routing rules.

Every call has to originate from somewhere. From your phone's perspective, calls originating from the trunk side are considered Inbound Calls, while calls originating from an endpoint are Outbound Calls. The call routing rule syntaxes for inbound calls and outbound calls are slightly different and are explained below. Call Routing Rule configuration relies heavily on digit maps. If you are not familiar with how digit maps work yet, please read the [Digit Map Configuration](#) section in this document first.

## Basic Incoming Call Handling

For each incoming call from a specific trunk, your phone handles the call in the following order:

- Ignore/reject the call if the trunk is disabled.
- Forward the call if native per-line.
- Call Forward Unconditional feature is enabled on the service.
- Apply Busy Treatment to the call if native per-line Do Not Disturb feature is enabled on the service.
- Apply Busy Treatment to the call if the number of existing calls on the trunk exceeds the limit set for the service.
- Apply the rules in the `InboundCallRoute` parameter of the service to determine where to send the call. A common destination for an incoming call is `ph` (to ring your phone).

Note the following:

- Busy Treatment refers to whether to reject/ignore the call or apply native per-line Call Forward On Busy if the feature is enabled on the service.
- When it comes to reject/ignore a call, the decision whether to reject or ignore is based on the service.
- You can configure **InboundCallRoute** to enable your phone for complex call handling. For example:
  - Ring the phone, the AA, and one or more cell phone numbers via SPn simultaneously. Whoever answers first takes the call.
  - If the caller number ends in 4281234 or 3357, ring the AA and a cell phone number simultaneously.

## *Inbound Call Route Configuration*

Every trunk has a corresponding **InboundCallRoute** in your phone's configuration. It is a comma-separated list of rules where each rule is also surrounded by a pair of curly brackets `{ }`. No extra white spaces are

allowed. These rules tell your phone how to handle an inbound call, such as sending it to the Phone (and ringing the attached phone(s)), sending it to the Auto Attendant for further routing (interactively with the caller), or making another call on a specific trunk to bridge with this call.

The general format is:

```
InboundCallRoute := rule OR {rule},{rule},...
```

The curly brackets may be omitted if there is only one rule in the route. The OR operator is NOT part of the parameter syntax. It is used here to separate alternative values only.

A rule has the following format:

### Inbound Call Route Structure

Inbound Call Route Structure				
{rule}, {rule}, {rule}, {rule}				
rule =				
{peer-list		:	terminal-list}	
peer-list =				
{(peering), (peering), (peering),		:	(terminal), (terminal)}	
peering =			terminal =	
{(caller-list	>	callee-list)	:	PHx OR AAx OR LIx(arg) OR SPx(arg) OR PPx(arg)}
caller-list =		callee-list =	arg =	
{caller caller caller	>	callee callee}	cid	> target
caller =		caller =	cid =	target =
number OR (digit-map) OR ? or @ (? = anonymous, @ = any #)		number OR (digit-map) OR @	(digit-map) OR spoofed-caller -number (the outbound CallerID)	(digit-map) OR number-to-call



- The `terminal-list` can be empty, which means to block this call. The preceding `:` cannot be omitted. As many as 4 terminals may be specified in the list. The listed terminals are called/rung by your phone simultaneously. This operation is called forking the call. A terminal may be a trunk or an endpoint.
- Abbreviated terminal names are case-insensitive.
- The `number` and `number-to-call` are literal strings, such as 14089991234.
- The `digit-map` is just any proper digit map, such as (1xxx|xx.). Make sure to include the enclosing parentheses
- The `spoofed-caller-number` is a literal string, such as 14081112233, to be used as the caller number for making a new call on the specified trunk
- (Mlabel) is a named digit map, where `label` is the abbreviated name of any terminal that has a digit map defined: SP1-SP6, PP, PH, or AA.
- `$1` is an internal variable containing the value of the caller number of this inbound call, after any digit map transformation in the matched caller object of the matched peering object in the peering-list.
- `$2` is an internal variable containing the called number of this inbound call, after any digit map transformation in the matched `callee` object of the matched peering object in the peering-list.

More notes on peering-list and peering objects:

- The `peering-list` is optional in **InboundCallRoute**. If `peering-list` is empty, the succeeding `:` can be omitted also. An empty peering-list implies a single peering object whose caller object list matches any caller number. That is, the following **InboundCallRoute** values are all equivalent:
  - ^ ph
  - ^ {ph}
  - ^ {:ph}
  - ^ {?|@>@:ph}
- The `callee-list` in a peering object can be empty. It implies the callee object @, meaning any called number. The preceding `>` can be omitted if `callee-list` is empty.
- The `caller-list` in a peering object can be empty. It implies the caller-list @|?, meaning any caller number including `anonymous`. The succeeding `>` cannot be omitted if `caller-list` is empty but not the `callee-list`.

More notes on the `arg`, `cid`, and `target` objects:

- The `cid` object inside an `arg` object is optional. If omitted, it implies no caller-ID spoofing when making the call on the specified trunk. The succeeding `>` can be omitted if `cid` is omitted.
- The `target` object inside an `arg` object is optional. If omitted, it implies the `target` `$2`, which means to call the original called number after applying any necessary digit map transformation implied by the rule. The preceding `>` cannot be omitted if `target` is omitted but `cid` is not.
- The `arg` object is optional. If omitted, it implies the `arg` with the `target` `$2` and no `cid`. If `arg` is omitted, the succeeding parentheses ( ) can be omitted also.

An inbound call matches a rule if its caller-number/callee-number matches one of the peering objects of the rule. Peering objects are tested in the order left to right, and the first matched peering object wins. Rules are also checked in the order left to right, and the first matched rule wins. Therefore, it is important that you place the more specific rules first in the **InboundCallRoute** if multiple rules can potentially match the same inbound call.

**InboundCallRoute** examples:

- `ph OR {ph} OR {:ph} OR {@|?>@:ph}` (all equivalent)  
It says: Ring the phone (only) for all incoming calls. This is the default `InboundCallRoute` for all trunks.
- `{14081223330|15103313456:ph,aa},{(1800xx.|1888xx.)|?:},{ph}`  
It says: Ring the phone and AA for calls coming from 1 408 122 3330 or 1 510 331 3456, block all 800, 888, and anonymous calls, and ring just the phone for all other calls.
- `{(x.4081113333|x.4152224444):aa},{ph}`  
It says: Ring the AA for calls coming from any number that ends with 408 111 3333 or 415 222 4444, and ring the phone for all other calls. Be sure to include the enclosing parentheses in this example since “x.” is a digit map specific syntax.
- `{200123456:aa},{sp1(14083335678)}`  
It says: Ring the AA for calls coming from 200123456. For all any other call, bridge it by calling 1 408 333 5678 using SP1 Service.

## Basic Outgoing Call Handling

When the trunk receives a number to call, it makes the call if the trunk is enabled, running, and under full capacity. Otherwise, the trunk fails to send the outgoing call.

Although there’s a `DigitMap` parameter available per service, the trunk doesn’t use it to validate the number to call. The validation completes at a higher level before the system routes the call attempt to the trunk for execution. The phone uses the per-line `DigitMap` as a reference in other `DigitMap` parameters (usually in **Phone Settings::DigitMap**) and in trunk selection from within a trunk group.

## Outbound Call Route Configuration

Every endpoint has an `OutboundCallRoute` parameter in your phone’s configuration. It tells your phone where to send the call when the endpoint attempts to make a call. Endpoints may call each other or an outside number using one of the trunks. The `OutboundCallRoute` syntaxes are almost identical to those of the `InboundCallRoute` parameter. The differences are mainly in the implied value when an optional field is omitted, there are no caller objects, and one and only one terminal object per terminal-list is present in an `OutboundCallRoute` parameter. Forking is not supported when routing outbound calls.

The general format is:

```
OutboundCallRoute := rule OR {rule},{rule},...
```



The curly brackets may be omitted if there is only one rule in the route. The **OR** operator is NOT part of the parameter syntax. It is used here to separate alternative values only.

A rule has the following format:

## Outbound Call Route Structure

Outbound Call Route Structure	
<code>{rule}, {rule}, {rule}, {rule}</code>	
<code>rule =</code>	
<code>{callee-list</code>	<code>: terminal}</code>
<code>callee-list =</code>	<code>terminal =</code>
<code>{callee callee</code>	<code>: PHx, AAx, SPx, LIx, OR PPx}</code>
<code>callee =</code>	
<code>number OR (digit-map) OR @</code>	<code>(@ = any number)</code>



- A terminal may be a trunk or another endpoint.
- Abbreviated terminal names are case-insensitive.
- The `number` and `number-to-call` are literal strings, such as 14089991234.
- The `digit-map` is just any proper digit map, such as (1xxx|xx.). Make sure to include the enclosing parentheses.
- The `spoofed-caller-number` is a literal string, such as 14081112233, to be used as the caller number for making a new call on the specified trunk.
- (Mlabel) is a named digit map where label is the abbreviated name of any terminal that has a digit map defined: SP1-SP6, PP, PH, or AA.
- \$2 is an internal variable containing the called number of this outbound call, after any digit map transformation in the matched callee object.
- The `callee-list` can be empty, which implies the single callee object @, which means any called number. The succeeding ':' can be omitted also when `callee-list` is empty.

More notes on the *arg*, *cid*, and *target* objects:

- The *cid* object inside an *arg* object is optional. If omitted, it implies no caller-ID spoofing when making the call on the specified trunk. The succeeding '>' can be omitted if *cid* is omitted.
- The *target* object inside an *arg* object is optional. If omitted, it implies the *target* \$2, which means to call the original called number after applying any necessary digit map transformation implied by the rule. The preceding '>' cannot be omitted if *target* is omitted but not the *cid*.
- The *arg* object is optional. If omitted, it implies the *arg* with the *target* \$2 and no *cid*.

An outbound call matches a rule if its called number matches one of the callee objects of the rule. Callee objects are tested in the order left to right, and the first matched callee wins. Rules are also checked in the order left to right, and the first matched rule wins. Therefore, it is important that you place the more specific rules first in the `OutboundCallRoute` parameter if multiple rules can potentially match the same outbound call.

Note that every endpoint also has a digit map defined. The user-dialed number is completely processed with the endpoint's digit map first before it is passed to the `OutboundCallRoute` parameter for routing decision. Therefore the number used for matching call routing rules has already incurred the transformations, if any, implied by the digit map. Remember this fact when crafting your own `OutboundCallRoute` parameter.

`OutboundCallRoute` examples:

- `sp1 OR {SP1} OR {:SP1} OR {@:Sp1}` (all equivalent).

This rule says: Make all calls using SP1 Service, without any caller-id spoofing or digit transformation.

- `{**0:aa}, {***:aa2}, {(Mpli):pli}, {(<**1:>(Msp1)):sp1}, {(<**2:>(Msp2)):sp2}, {(<**9:>(Mpp)):pp}`

**This is the default *Phone Settings*::OutboundCallRoute** parameter. It says:

- Dial `**0` to invoke AA.
- Dial `***` to invoke the local phone configuration IVR (also known as AA2).
- `(Mpli)` and `pli` are substituted with the abbreviated name of the **PrimaryLine** value.
- Use SP1 Service to call all numbers that start with `**1` and subsequent digits matching SP1 Service's `DigitMap` parameter. Remove the `**1` prefix from the resulting number before making the call.
- Use SP2 Service to call all numbers that start with `**2` and subsequent digits matching SP2 Service's `DigitMap` parameter. Remove the `**2` prefix from the resulting number before making the call.
- Use the OBiTALK Service to call all numbers that start with `**9` and subsequent digits matching OBiTALK Services `DigitMap` parameter. Remove the `**9` prefix from the resulting number before making the call.

## Voice Gateways and Trunk Groups

This section describes voice gateways and trunk groups, and includes details on accessing both.

### Voice Gateways

A gateway is another phone that enables two stage dialing. Incoming OBiTALK callers to call the gateway first with a normal OBiTALK call, get the AA, and then dial the target number. For authentication, the AA may ask the user to enter a PIN before establishing the second call.

You can also configure a gateway with one-stage or direct dialing so that the caller can dial the target number directly without going through the AA. Because a user can't enter a PIN when direct dialing, you can configure an optional user ID and password so the device can automatically authenticate with the gateway.

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

For example, you can add the rule `{(1xxx xxx xxxx):vg2}` in ***Phone Settings*::OutboundCallRoute** to enable the device dial out using VGs when the caller dials any 11-digit number starting with 1. On the gateway side, you can add the corresponding rule `{>(1 xxx xxx xxxx):sp1}` in the ***OBiTALK Service*::InboundCallRoute** to make the call on its SP1 trunk. You can change the last rule to `{(290 333 100|200 444 101)>(1 xxx xxx xxxx):sp1}` if you want to limit the gateway to allow just the two stated caller numbers to make such calls.

You can also configure a gateway with a SIP URL as the access number over one of the SP trunks. For example, you can set the gateway access number as `SP1(some-sip-server.mydomain.com)` or `SP2(192.168.15.111:5062)`.

Note the following when using an SP trunk to access a (SIP) gateway:

- The device doesn't use the outbound proxy, ICE, or STUN regardless of the settings on the SP trunk.
- The device uses only its local address as the **SIP Contact** and ignores any netted address discovered by the device.
- The device uses the gateway's SIP URL to form the FROM header of the outbound INVITE.
- The device uses the gateway's **AuthUserID** and **AuthPassword** for authentication.
- The device applies the symmetric RTP concept.

## Trunk Groups

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

Availability of trunk is based on the following criteria:

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

You can configure as many as four trunk groups on your phone. You can reference a trunk group and its associated digit map using the short name  $TGn$  and (Mtgn) and reference them in other digit maps and call routing rules so that the system can route calls to a particular trunk group.

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



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

## Trunks, Endpoints, and Terminals

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.

The following matrix shows the possible call bridge connections.

### Supported Two-Way Call Bridges

	SP1 Service	SP2 Service	SP3 Service	SP4 Service	SP5 Service	SP6 Service	OBiTALK Service
SP1 Service	yes	yes	yes	yes	yes	yes	yes
SP2 Service	yes	yes	yes	yes	yes	yes	yes
SP3 Service	yes	yes	yes	yes	yes	yes	yes
SP4 Service	yes	yes	yes	yes	yes	yes	yes
SP5 Service	yes	yes	yes	yes	yes	yes	yes



**Supported Two-Way Call Bridges (continued)**

	SP1 Service	SP2 Service	SP3 Service	SP4 Service	SP5 Service	SP6 Service	OBI TALK Service
SP6 Service	yes	yes	yes	yes	yes	yes	yes
OBI TALK Service	yes	yes	yes	yes	yes	yes	yes

Each supported service is also referred to as a trunk, a traditional telco term for 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 OBI TALK Service

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

The phone (PH) and the AA are the entities in your phone where calls can terminate (start or end there), whereas trunks rely on the corresponding service providers to terminate the call. In this document, the phone and the AA are referred to as endpoints. 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.

The following matrix shows the possible call connections between the endpoints and the trunks:

**Supported Endpoint Calls**

	Any Trunk	PH1	AA
Any Trunk	n/a	yes	yes
PH	yes	no	yes
AA	yes	yes	no

# Third-Party Servers

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This section provides information on configuring phones and features with third-party servers.

## BroadSoft

This section shows you how to configure Poly OBi Edition devices 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**

<b>Display Caption (Default)</b>	<b>Action URL</b>	<b>Summary</b>
Net Services	Phone://netsrv	A collection of settings of features the service provider offers. The settings are organized 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.

## Net Services App

BroadSoft offers a set of user features that you can access through a few APIs that correspond to an SP service on the phone. You can find these features under **Network Services per SP** service on the phone and the settings under the **Net Services** App on the Main menu.

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

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

The softkey set on this screen is configurable with these options:

- BCI
- DND
- CFA

- ACD

You can select the SP service to view or change all the available settings for the SP service.

The ordering and the list of items to include in the second-level menu are configurable with the **NetServicesMenu1** parameter.

## Net Dir App




The usage depends on the app provider. There are two options:

- BroadSoft Directory
- LDAP

## Call Recording Controls

(SIP/BroadSoft.) Your phone supports the call recording functions available with a BroadSoft application server, by providing the controls for call recording during a call. You can enable this feature on a per SP service basis using the **SPn Service – Network Provided Services::CallRecording** parameter. When you enable this feature, the phone shows one of the following call recording states in the call items of the Calls App:

### Call Recording Controls

Call Recording State	Description	Available Softkey Options	Icon
off	Recording has been turned off	Rec.Start	
on	Recording has been turned on	Rec.Stop, Rec.Pause	
paused	Recording has been turned on but paused at the moment	Rec.Resume	
na	Feature is not enabled	—	

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

## BroadSoft Call Center Features

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

### Disposition Code

A disposition code can be entered by an agent for the current call that is still ongoing or for the last call that has just ended. For the first case, the agent selects the **Dispose Code** softkey that is available when the

call is in the connected state. The agent then enters the code and submits it while talking to the caller. For the latter case, the agent can press the feature key that has been assigned the `Disposition Code` function right after the call, then enter and submit the code.

To use this feature on the phone, you must enable the option ***SPn Service – Network Provided Services::DispositionCode***. The option only applies to calls on the same SP service.

## ***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 is executed entirely on the softswitch after invocation from the phone. To use this function on the phone, you must enable the option ***SPn Service – Network Provided Services::CallTrace***. The option only applies to calls on the same SP service.

To start a call trace outside the context of a call, the administrator can define a speed dial feature key with the following parameters:

- `Function = Speed Dial`
- `Number = customer-originated-trace`
- `Server = The SP Service to invoke the Call Trace operation with, such as SP3`

## ***Escalation***

The agent can escalate the current call to a specific supervisor or the predefined default supervisor by selecting the **Escalate** softkey that is available when the call is in the Connected State. After pressing the key, the agent has a chance to enter a specific supervisor's extension, or skip that to use the default supervisor's extension. To use this feature on the phone, you must enable the option ***SPn Service – Network Provided Services::Escalation***. The option only applies to calls on the same SP service.

## ***Call Center Information***

When the softswitch sends an incoming call to the phone, it may include some basic information about the call center where the call is coming from. Such information, if available, is displayed by the phone with the Ring Alert message for the incoming call. The following information can be displayed:

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

This behavior as described above does not have any configuration.

In addition, the phone can proactively subscribe to the status of the call center that a SP service belongs to. This feature is enabled with the option ***SPn Service – Network Provided Services::CallCenter***. The call center status information can be viewed at any time on the phone by going through the Net Services App (from the phone's main menu).



## BroadSoft Guest Login/Logout (Hoteling)

(SIP only.) This feature is also known as Hoteling (consult BroadSoft documentation on how to administer this feature on the server). The phone may be set up to be used temporarily by a guest, such as a visiting employee or temp worker in a hot desk environment. To use this feature on the phone, the option **SPn Service – Network Provided Services::Hoteling** must be enabled. If enabled, the phone starts a subscription to the x-broadworks-hoteling event package in the context of a SP service, right after the first successful registration with the SIP Proxy Server. The expires value of this subscription can be set with the **ITSP Profile – SIP – Feature Configuration::X\_BWHotelingSubscribeExpires** parameter.

There are two ways to access the guest login/logout function:

- Within the **Net Services** App, select the SP service, and then select the Hoteling option
- Define a feature key with the function `Hoteling` (but no more than one hoteling feature key per SP service). The LED and the icon of the key reflect the current guest login state, as shown in the following table:

### Hoteling Settings

Guest Login State	Description	Icon	LED
Guest Logged In	A guest has successfully login. The login extension is shown on the screen		Solid green
Guest Logged Out	No guest login		Off
Protocol error	Problem reaching the server or subscribing to the service		Off

When trying to login, the guest is prompted to enter the guest extension first. After submitting a valid guest extension, the guest is prompted to enter the corresponding password. Once the password is accepted by the server, the phone screen shows the guest extension in the **Call Appearance** and **Line Monitor** keys that are bound to the SP service. The guest logs out by pressing the **Hoteling** feature key again once, or the server logs out the guest remotely. The screen updates automatically according to the updated guest login state.

## BroadSoft AS-Feature-Event Features

The AS-Feature is a collection of network-provided features available on a BroadSoft application server. You can view and change the settings from the phone UI. These network-provided features are configured and executed in the context of a single SP service.

To allow any of the network-provided features listed below to be viewable and changeable from the phone, you must enable the option **SPn Service – Calling Fetures::X\_ASFeatureEventSubscribe** and must also enable the individual network-provided feature whichever you want to allow users to access from the phone.



The features themselves are executed entirely on the server and the settings of the features are stored on the server. The phone displays the values of the settings as stored on the server (not the ones entered and submitted by user, which may or may not be acceptable by the server).

The as-feature is based on the SIP subscribe/notify framework. The expires value of the subscription dialog (initiated by the phone per SP service with the feature enabled) can be set using the **ITSP Profile X-Feature Configuration::X\_ASFeatureEventSubscribeExpires** parameter. On each subscribe request sent by the phone, the server returns a NOTIFY that refreshes the current settings of all the softswitch-provided features listed below (whichever is enabled or made available on the server). When a setting is changed, the server also updates the phone with a NOTIFY that specifies the latest settings of just the affected features.

Some of the network-provided features can be accessed from the phone via special feature key functions (such as Do Not Disturb) or special softkeys (such as Call Forward All), while all enabled network-provided features can be accessed via the Net Services App of the Main menu.

## Call Forward All

CallForwardAll is synonymous to CallForwardUnconditional in this document. This feature, if enabled, lets the softswitch forward all calls to the configured forwarding number unconditionally. The functionality provided by this feature is similar to that of the CallForwardUnconditional feature provided natively by the phone (per line). The administrator is advised to disable the native version when using the network-provided version to avoid ambiguity. To make the setting of this network-provided service viewable and changeable from the phone UI, you must enable the option **SPn Service – Network Provided**

**Services::CallForwardAlways**. In addition to going through the Net Services app, the following methods of access are available:

- Feature Key: Define a feature key with the function Call Forward. Set the Service parameter of the key to the SP service that provides this feature.
- Softkey: The **Call Forward** softkey on the home screen of the phone can be mapped to the network provided CallForwardAll service on a specific SP service, by setting the **Phone Settings – User Preferences Settings::CallForwardUnconditionalFeatureProvider** parameter to equal to that SP service.
- User Preferences – Call Forward Setting: As a side effect of enabling the softkey option in the last item, the Call Forward Setting under User Preferences app on the UI is also pointing to the same network provided **CallForwardAlways** feature.

You can enable or disable **CallForwardAlways** as well as setting a forwarding number. The settings are submitted and stored on the server.

## Call Forward Busy

To make the setting of this network-provided service viewable and changeable from the phone, you must enable the option **SPn Service – Network Provided Services::CallForwardBusy**. The functionality provided by this feature is similar to that of the CallForwardOnBusy feature that is available natively on the phone (per line). To use the version provided by the softswitch, the administrator is advised to disable the native version to avoid ambiguity.

The only way of access from the phone is by going through the Net Services app. The phone allows you to get and change the setting of this feature on the server. The setting is submitted to the server with a subscribe request and stored on the server.

## Call Forward No Answer

To make the setting of this network-provided service viewable and changeable from the phone, you must enable the option **SPn Service – Network Provided Services::CallForwardNoAnswer**. This feature is similar to the `CallForwardOnNoAnswer` feature that is available natively on the phone (per line). To use the version that is provided by the softswitch, the administrator is advised to disable the native version to avoid ambiguity.

The only way of access from the phone is by going through the Net Services app. You can enable or disable the feature, set the forwarding number, and specify the number of rings before forwarding the call. These settings are submitted to the server with a subscribe request and stored on the server.

## Do Not Disturb

To make the setting of this network provided service viewable and changeable from the phone, you must enable the option **SPn Service – Network Provided Services::DoNotDisturb**. The functionality provided by this feature is similar to that of the `DoNotDisturb` feature that is available natively on the phone (per line). To use the version that is provided by the softswitch, the administrator is advised to disable the native version to avoid ambiguity.

In addition to going through the Net Services app, the setting can also be accessed via:

- Feature Key: Define a feature key with the function `Do Not Disturb`. Set the Service parameter of the key to the SP service that provides this feature.

You can enable or disable this feature from the UI. The setting is submitted to the server with a subscribe request and stored on the server.

## ACD Agent State

ACD stands for Automated Call Distribution and is the primary way a call center distributes calls among a number of agents. Normally the ACD controller only sends a new call to an agent who is in the “Available” state. An agent typically “Signs On” when they arrive at work and then “Signs Off” when done for the day (or taking a very long break). Agents may at any time tag themselves as “Unavailable” when taking a break, or “Wrapping Up” when filing paperwork for the last call before taking another call. Through as-event subscription, your phone allows an agent to sign on, sign off, or change their availability states from the phone UI. To make this feature available on the phone, you must enable the **SPn Service – Network Provided Services::ACDAgent** option.

You can invoke this feature in two ways:

- Press the feature key with the function `ACD Agent`
- Within the Net Services App, select the SP service, and then select the **ACD Sign On/Off** item

With the ACD feature key, the agent can sign on by pressing the key once.

You can sign on or off and change your state from the phone UI. You can set your state to one of the following values:

- Available (to take new calls)
- Unavailable (to take new calls)
- Signed On
- Signed Off
- Wrapping Up (the last call)

While Signed Off, press the key once to sign on and become Available. While Available, press the key once to become Unavailable. You must also enter one of the valid unavailable reason codes (such as 11) that are defined by the Call Center administrator. While Unavailable or Wrapping Up, you can press the key once to become Available again.



You cannot change your state to Signed Off or Wrapping Up directly by pressing the feature key. To change to these states, you 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 for agents.

## Security Classification

To make the Security Classification setting viewable and changeable from the phone, you must enable the option **SPn Service – Network Provided Services::SecurityClass**. With that the phone lets you view or set the security levels for your extension. The currently available security levels you are allowed to choose are presented to you on the screen. In addition to invoking this function by going through the Net Services App, the administrator can also define a feature key with the function `Security Class` as a shortcut to launch this function.

## Executive Call Filter

To make the setting of the Executive Call Filter option viewable and changeable from the phone, you must enable the option **SPn Service – Network Provided Services::Executive**. In addition to invoking the function through the Net Services App, the administrator can define a feature key with the function `Exec Filter On/Off` as a shortcut to turn this setting on or off. The LED color also reflects the current on or off status.

## Executive Assistant

By enabling the option **SPn Service – Network Provided Services::ExecutiveAssistant**, the phone makes the following settings of this feature available to 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.

In addition to invoking the function through the Net Services App, the administrator can define a feature key with the function `Exec Assistant` as a shortcut to launch this option. The LED color also reflects the current **Divert** on or off status.



## Call Recording Settings

The phone can extract the call recording settings from the as-event notifications to help determine if and which call recording controls to present to you during a call, provided the **SPn Service – Network Provided Services::CallRecording** is also enabled.

## BroadSoft XSI Features

This is a collection of features that is provided with a BroadSoft XSI Application Server. The phone makes XSI Features available per SP/SIP service. This allows the configuration of as many as six independent sets of XSI services per phone, one per SP service. To execute any of the features listed in this section, the following parameters are required:

### BroadSoft XSI Features

Parameter Group	Parameter	Description
<i>ITSP Profile X – SIP</i>	<b>X_XsiServer</b>	The XSI server hostname or IP address. Phone attempts to resolve the hostname as DNS A Record only. DNS SRV lookup isn't supported here.
<i>ITSP Profile X – SIP</i>	<b>X_XsiServerPort</b>	The server port. If not specified (or 0), the default port is used (80 for HTTP or 443 for HTTPS).
<i>ITSP Profile X – SIP</i>	<b>X_XsiServerScheme</b>	Must be HTTP or HTTPS.
<b>SPn Service – SIP Credentials</b>	<b>X_XsiUserName</b>	The user name to authenticate to the XSI server with. If not specified (blank), the phone forms the user name as: <code>{sip-userid}@{sip-domain}</code> where <code>{sip-userid}</code> is the SIP Account User ID that is used for SIP Registration on the same SP service, and <code>{sip-domain}</code> is the domain name that is used for SIP Registration on the same SP service.
<b>SPn Service – SIP Credentials</b>	<b>X_XsiPassword</b>	The password to authenticate to the XSI server with. If not specified (blank), the same password for SIP authentication on the same SP service is used.

Some of the XSI features can be accessed from the phone by launching dedicated apps (such as Network Directories) or via special feature key functions (such as Do Not Disturb) or special softkeys (such as Call Forward All). In addition, all enabled XSI services can also be accessed under the Net Services app of the Main menu.



- **CallForwardAll**, **CallForwardBusy**, **CallForwardNoAnswer**, and **DoNotDisturb** can also be accessed using the **AS-Feature-Sync** method. If **AS-Feature-Sync** is enabled, then it is used instead of XSI to access these features.
- Except for network directories and call logs, the XSI server needs to notify the phone when settings have changed on the server. This notification is done using an event channel over HTTP. The method for sending HTTP messages is to use Comet.

## Network Directories

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

### BroadSoft Hosted PBX Platform

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

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

Please consult your BroadSoft documentation on how to setup and manage these directories on the server side.

To access this service from the phone, you must enable the option **SPn Service – Network Provided Services::Directory**. You can invoke the Network Directory service of a specific SP service from the phone by launching the Net Dir app from the Main menu of the phone UI. Which SP service's Network Directories service is invoked is controlled by the **Phone Settings – Network Directory::VoiceService** parameter. In order for the Net Dir item to show on the phone's Main menu, both **Phone Settings – Network Directory::Enable** and **SPn Service – Network Provided Services::Directory** of the corresponding SP service must be enabled. You can access the Network Directories on other SP services through the Net Services App.

It should be noted that all the directory data are stored entirely on the server and are downloaded once when the network directories function is invoked on the phone. After initial invocation, you can refresh the data by pressing the **Refresh** or **Refresh All** softkey. A Search function lets you enter a search string of a name pattern (such as OBi\*) and submit it to the server. The server then returns a list of entries matching the search criteria to display on the phone screen.

### Network Directory Softkey Options

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.

You can disable the Search option by setting the **SPn Service – Network Directory Setup::EnableSearch** to `false`. This replaces the **Search** softkey with the **Refresh** softkey, which reloads the results from the URL.

### Other PBX Platforms

A service provider can host a Group directory similar to the format used in BroadWorks on the network. The URL to access this directory is configured in the: **SPn Service – Network Directory Setup::URL** parameter. For example:

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

The returned document should be an XML (properly escaped) as shown in the example below:

```
<?xml version='1.0' encoding='utf-8' ?>
<Group>
  <startIndex>1</startIndex>
  <numberOfRecords>2</numberOfRecords>
  <totalAvailableRecords>2</totalAvailableRecords>
  <groupDirectory>
    <directoryDetails>
      <name>Samuel Samson</name>
      <firstName>Samuel</firstName>
      <lastName>Samson</lastName>
      <userId>ssamson</userId>
      <number>14089991234</number>
      <extension>104</extension>
      <department>Marketing</department>
      <groupId>Management</groupId>
      <impId>ssamson@abcd-wallpaper.com</impId>
      <additionalDetails>
        <emailAddress>sam.samson@abcd-wallpaper.com</emailAddress>
        <mobile>14089998899</mobile>
      </additionalDetails>
    </directoryDetails>
    <directoryDetails>
      <name>John J. Smith</name>
      <firstName>John</firstName>
      <lastName>Smith</lastName>
      <userId>john</userId>
      <number>14087771123</number>
      <extension>103</extension>
      <department>Sales</department>
      <groupId>Staff</groupId>
      <impId>jjs@abcd-wallpaper.com</impId>
      <additionalDetails>
        <emailAddress>john.smith@abcd-wallpaper.com</emailAddress>
        <mobile>16509991802</mobile>
      </additionalDetails>
    </directoryDetails>
  </groupDirectory>
</Group>
```

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

- **results** Number of results to return by the server. This is merely a suggestion to the server which may return a different number of results. The phone won't reject the result if the number of results returned is different from the request and 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:!4xuIKK12y@mypbx.com/group-dir.php?user=jsmith>

## Replace the Built-In Phone Book with a Network Directory

You can replace the built-in phone book with a Network Directory, such that when you press the **phbk** softkey or select **Contacts** from the Main menu, your phone launches the corresponding network directory instead of the built-in (local) phone book. Below is the configuration for this:

**User Preferences – Phone Book Settings::ActionURL** = phone://netdir

**Phone Settings – Network Directory::VoiceService** = SPn

(where n = 1, 2, 3, .. )

## Network Call Logs

The network call logs consist of four logs: All, Missed, Received, and Outgoing. The log data are stored entirely on the server and are downloaded to the phone when you invoke this function from the phone. Please consult BroadSoft on how to manage these call logs on the server side.

To make this function available on the phone, you must enable the option **SPn Service – Network Provided Services::CallLogs**. There is no specialized app, feature key functions or softkey options to launch network call logs. You can only invoke this function by going through the Net Services app.

If the Buddy List is also enabled and available under the same SP service, the phone also displays the presence icon in the network directory if it can be found in the buddy list.

## BroadWorks Anywhere

You can view and change the following settings of this feature from the phone UI:

- 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 (Note: The last location cannot be removed via XSI).
- Edit a location's **Number** or **Description** attributes.

To make this function available on the phone, you must enable the option **SPn Service – Network Provided Services::BroadWorksAnywhere**. There is no specialized app, feature key functions or softkey options to launch this function. You can only invoke this function by going through the Net Services App.

## Remote Office

You can view and change the following settings of this feature from the phone UI:

- Enable or disable this feature.
- Change the Remote Phone Number (Note: The number cannot be removed via XSI).

To make this function available on the phone, you must enable the option **SPn Service – Network Provided Services::RemoteOffice**. There is no specialized app, feature key functions or softkey options to launch this function. You can only invoke this function by going through the Net Services App.

## Simultaneous Ring

You can view and change the following settings of this feature from the phone UI:

- 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 (Note: The last location cannot be removed via XSI).
- Change the **Phone Number** attribute of a location.
- Turn the **Answer confirmation required** option on or off for each location.

To make this function available on the phone, you must enable the **SPn Service – Network Provided Services::SimultaneousRing** option. There is no specialized app, feature key functions, or softkey options to launch this function. You can only invoke this function by going through the Net Services App.

## Call Forward Always

To make this function available on the phone, you must enable **SPn Service – Network Provided Services::CallForwardAlways**. This feature is similar to the **CallForwardUnconditional** feature that is available natively on the phone (per service). To use the version that is provided by the softswitch, the administrator should disable the corresponding service on the phone to avoid ambiguity.

In addition to going through the Net Services app, the following methods of access are available:

- Feature Key: Define a feature key with the function Call Forward. Set the Service parameter of the key to the SP service that provides this feature.
- Softkey: The Call Forward softkey on the home screen of the phone can be mapped to the network provided **CallForwardAll** service on a specific SP service, by setting the **Phone Settings – User Preferences Settings::CallForwardUnconditionalFeatureProvider** parameter to equal to that SP service.
- User Preferences – Call Forward Setting: As a side effect of enabling the softkey option in the last item, the Call Forward Setting under the User Preferences app on the UI points to the same network-provided **CallForwardAlways** feature.

You can enable or disable **CallForwardAlways** as well as set a forwarding number. The settings are stored on the server.

## Call Forward Busy

To make this function available on the phone, you must enable n **SPn Service – Network Provided Services::CallForwardBusy**. This feature is similar to the **CallForwardOnBusy** 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 to avoid ambiguity.

Use the Net Services app to access the feature on the server. The phone lets you access and set this feature and store feature settings on the server.

## Call Forward No Answer

To make this function available on the phone, you must enable **SPn Service – Network Provided Services::CallForwardNoAnswer**. This feature is similar to the **CallForwardOnNoAnswer** feature that is available natively on the phone (per service). To use the version that provided by the softswitch, disable the corresponding service on the phone to avoid ambiguity.

Use the Net Services app to access the feature on the server. You can enable and disable the feature, set the forwarding number, set the number of rings before forwarding the call and store feature settings on the server.

## Anonymous Call

To make this function available on the phone, enable **SPn Service – Network Provided Services::AnonymousCall**. 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 UI, and the server stores the feature settings.

## Do Not Disturb

To make this function available on the phone, enable **SPn Service – Network Provided Services::DoNotDisturb**. This feature is similar to the **Do Not Disturb** feature available natively on the phone (per line). To use the version provided by the softswitch, disable the corresponding feature on the phone to avoid ambiguity.

Use the Net Services app to access the feature and define a feature key with the function `Do Not Disturb`. Set the Service parameter of the key to the SP service that provides this feature to users who can enable or disable this feature from the UI. The server stores the feature settings.

## Real-Time Voice Transcription

VVX business IP phones, OBi Edition, can stream live phone audio (in the form of PCM samples) to your server-based natural language processing (NLP) service. Depending on your NLP service, you can save phone recordings for training or quality control, produce live conversation analytics data, and monitor live conversation transcriptions.

## Set Up Real-Time Voice Transcription

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

### To set up real-time voice transcription:

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

**Voice Transcription Parameters**

Parameters	Value
<b>URL</b>	Transcription service URL.
<b>Key</b>	Transcription service authentication key The authentication key is provided by your transcription service provider.
<b>Secret</b>	Secret for authentication. The secret is provided by your transcription service provider.
<b>Config</b>	Configuration string.

- 2 Select **Submit**.

## ***Enable Real-Time Voice Transcription***

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

**To enable real-time voice transcription:**

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

# Device Web Page and Configuration Parameter Reference

This reference section lists all the configuration and status parameters. Configuration parameters can be modified. Status parameters are read-only.

## Status Parameters

The Status parameters web pages show read-only values for certain parameters on your phone. They include these pages:

- [System Status Web Page](#)
- [Call Status Web Page](#)
- [Call History Web Page](#)
- [SP Services Stats Web Page](#)

## System Status Web Page

System Status is available on the phone built-in portal and on the OBITALK device management portal, under the web page with the same title.

### System Status Settings

Parameter	Description	Example Value
<i>WAN Status (DeviceInfo.Network.Status.WAN.)</i>		
<b>AddressingType</b>	Method currently used by the phone to get an IP address assignment.	DHCP
<b>IPAddress</b>	IP address currently assigned to the phone when using static IP addressing.	192.168.15.165
<b>SubnetMask</b>	Subnet mask to use when using static IP addressing.	255.255.255.0
<b>DefaultGateway</b>	Gateway to use when using static IP addressing.	192.168.15.1
<b>DNSServer1</b>	URL for domain name server 1 when using static IP addressing.	8.4.4.4
<b>DNSServer2</b>	URL for domain name server 2 when using static IP addressing.	4.2.2.2
<b>MACAddress</b>	MAC address installed on the phone.	9CADEF90004E



## System Status Settings (continued)

Parameter	Description	Example Value
<b>LLDP-MEDStatus</b>	Enables LLDP media endpoint discovery for improved network connections.	Enabled
<b>CDPStatus</b>	CDP discovery status.	Disabled
<b>802_1XStatus</b>	802_1X authentication status.	HELD
<b>SWPortLinkStatus</b>	Status of the switch port (the port connecting the phone to the Internet).	Link:Y; Spd:100F; Auto:N
<b>PCPortLinkStatus</b>	Status of the PC port (the port connecting the phone to an auxiliary PC).	Link:N; Spd:100F; Auto:Y
<b>Stats</b>	Ethernet statistics.	P/E: Rx=82276/0; Tx=5538/0

*WiFi Status (DeviceInfo.)*

<b>AddressingType</b>	The method to assign an IP address to this interface. Choice of: <ul style="list-style-type: none"> <li>• DHCP</li> <li>• Static</li> </ul>	DHCP
<b>IPAddress</b>	The IP address to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>SubnetMask</b>	The subnet mask to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>DefaultGateway</b>	The default gateway to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>DNSServer1</b>	An additional DNS Server to use in addition to the ones received from DHCP.	None
<b>DNSServer2</b>	An additional DNS Server to use in addition to the ones received from DHCP.	None
<b>MACAddress</b>	The physical address of this interface.	9CADEF90004E
<b>Stats</b>	WiFi statistics.	P/E: Rx=0/0; Tx=0/0
<b>Platform</b>	WiFi platform.	OBiWiFi 5G

*Product Information (DeviceInfo.)*

<b>ModelName</b>	Your phone's model name.	VVX 450
<b>MACAddress</b>	Your phone's MAC address.	9CADEF90004E
<b>SerialNumber</b>	Your phone's serial number.	88H01NA00ZXV
<b>OBiNumber</b>	Your phone's OBi number, a value that uniquely identifies your phone to Poly and to other OBi devices.	552 860 300
<b>PartNumber</b>	Your phone's part number.	3111-48840-001
<b>HardwareVersion</b>	Your phone's hardware version.	1.1

**System Status Settings (continued)**

Parameter	Description	Example Value
<b>SoftwareVersion</b>	Your phone's installed software version. This value changes with a firmware update or downgrade.	6.4.1.10856
<b>SystemTime</b>	Shows the current time on the system.	15:32:35 1/29/2019, Wednesday
<b>UpTime</b>	With last Reboot Reason in parentheses. See the <a href="#">Reboot Reason Codes</a> table following this table for the reboot reason codes.	20 Days 5:04:13 (2)
<b>CertificatesStatus</b>	Indicates if a device certificate is installed on the phone.	Installed
<b>CustomizationStatus</b>	Indicates if this phone is a customized unit.	Generic
<b>BackgroundUpdate</b>	Indicates if background firmware update is enabled on this phone.	Enabled

**SPn Service Status (VoiceService.1.VoiceProfile.1.Line.n), n = 1 – 6**

<b>Status</b>	Registration status of this service. If there are problems with the registration or authentication, the SIP 4xx – 6xx error code and error message are displayed here. This is very useful information for troubleshooting issues with SIP-based services.	Registered (server=192.168.15.118; expire in 39s)
<b>PrimaryProxyServer</b>	IP address of the current Primary Proxy Server if proxy server redundancy is enabled on this service.	10.100.123.234
<b>SecondaryProxyServer</b>	IP address of the current Secondary Proxy Server if proxy server redundancy and secondary registration are both enabled on this service.	10.100.234.123
<b>CallState</b>	Describes the state of an active call on this service.	0 Active Calls

**OBiTALK Service Status (VoiceService.1.X\_P2P.1.Stats.)**

<b>Status</b>	Connection status with the OBiTALK network.	Normal (User Mode)
<b>CallState</b>	Describes the state of an active call on OBiTALK.	0 Active Calls

## Reboot Reason Codes

When your phone reboots, it provides one of the following reboot reason codes:

**Reboot Reason Code Descriptions**

Reason Code	Description
1	Reboot on power cycle.
2	Operating system reboot.
3	Reboot after firmware update via provisioning or phone (**6).

**Reboot Reason Code Descriptions (continued)**

Reason Code	Description
4	Reboot after new profile invoked.
5	Reboot after parameter value change or firmware change, and invoked via device web page.
6	Reboot after factory reset.
7	New Profile Invoked AND profile URL changed.
8	Reboot from SIP NOTIFY (reserved).
9	Reboot from phone IVR.
10	Reboot from webpage — No change in parameter value or firmware.
11	Reboot during OBiTALK signup.
12	Reboot during OBiTALK signup.
13	Reboot after DHCP server offers IP, GW-IP, and/or Netmask different from what the phone is currently using.
16	Reboot after DHCP RENEW NAK received.
18	Reboot after WAN IP setting(s) changed.
30	Reboot from Phone UI — No change in parameter value.
31	Reboot from Phone UI — After parameter value change.

## Call Status Web Page

The Call Status page shows a number of running call statistics and state parameters for each active call currently in progress. Call Status of each call is only available during the lifetime of the call. It is removed as soon as the call is ended.

The following information and statistics are shown for each call:

**Call Status Descriptions**

Status	Description
Terminal ID	A short name to identify each call terminal: Phone, SP1, SP2, SP3, SP4, SP5, SP6, OBiTALK.
State	Call State.
Peer Name	Call Peer's Name.
Peer Number	Call Peer's Number.
Start Time	Starting time of the call.
Duration	Duration of the call.
Peer RTP Address	The peer address and port where RTP packets are sent to.

**Call Status Descriptions (continued)**

Status	Description
Local RTP Address	The local address and port where RTP packets are sent from.
RTP Transport	The transport used for RTP (UDP, TCP, or SSL).
Audio Codec	The audio encoder and decoder being used for this call.
RTP Packetization (ms)	The transmitted and received packet sizes in milliseconds.
RTP Packet Count	Total number of RTP packets transmitted and received thus far.
RTP Byte Count	Total number of RTP bytes transmitted and received thus far.
Peer Clock Differential Rate	Clock difference between this phone and the peer in ppm (parts per million).
Packets in Jitter Buffer	Number of packets in the Jitter Buffer.
Packets Out-of-Order	Number of received RTP packets that are out of order.
Packets (10ms) Interpolated	Number of frames interpolated.
Packets Lost	Number of incoming RTP packets assumed lost.
Packet Loss Rate	Amount of incoming RTP packets assumed lost rate in percent.
Packet Drop Rate	Amount of incoming RTP packets dropped in percent.
Jitter Buffer Length	Size of the current jitter buffer in milliseconds.
Received Interarrival Jitter	Average measured network jitter in the received direction in milliseconds.
DTMF Digits Received	Number of DTMF digits received.
Jitter Buffer Underruns	Amount of jitter buffer underruns during the call.
Jitter Buffer Overruns	Amount of jitter buffer overruns during the call.
Sequence number discontinuities	Number of times a discontinuity is encountered in the sequence number of the incoming RTP stream.
Round-Trip Propagation Delay	A measurement of the round trip propagation delay of RTP packets.
End System Delay	Contribution to the overall end-to-end system delay by this end-point in milliseconds as defined in RFC3611.
Peer End System Delay	Contribution to the overall end-to-end system delay by the peer end-point in milliseconds as defined in RFC3611.
MOS-LQ	The estimated mean opinion score for listening quality.
MOS-SQ	The estimated mean opinion score for conversational quality.

For each entry on the call status page, the following buttons are available:

- **Remove:** This button is available for all calls. Pressing this button ends that call.
- **Record:** Pressing this button lets you record the current conversation in an audio (.au) file.

## Call History Web Page

This page shows the calls in the call history, in order of most recent to oldest.

The following buttons are available:

- **Remove All:** Pressing this button erases the entire call history.
- **Save All:** Pressing this button saves the call history to the `callhistory.xml` file.

The **Call History** page isn't available on the OBiTALK.com website.

## SP Services Stats Web Page

This page shows the following information for each SP $n$  service, where  $n = 1 - 6$ .

### SP Services Settings

Parameter	Description	Default Setting
<i>Reset Statistics (VoiceService.1.VoiceProfile.1.Line.n.Stats.), n = 1 - 6</i>		
<b>ResetStatistics</b>	Check this option and press <b>Submit</b> to reset the statistics for this SP Service.	NA
<i>RTP Statistics (VoiceService.1.VoiceProfile.1.Line.n.Stats.), n = 1 - 6</i>		
<b>PacketsSent</b>	Total RTP packets sent on this line.	NA
<b>PacketsReceived</b>	Total RTP packets received on this line.	NA
<b>BytesSent</b>	RTP payload bytes sent for this line.	NA
<b>BytesReceived</b>	RTP payload bytes received for this line.	NA
<b>PacketsLost</b>	Number of RTP packets lost on this line.	NA
<b>Overruns</b>	Number of times a jitter buffer overrun was received on this line.	NA
<b>Underruns</b>	Number of times a jitter buffer underrun was received on this line.	NA

## OBiWiFi Configuration Parameters

The OBiWiFi Configuration parameters web pages show Wi-Fi parameters on your phone. They include these pages:

- [WiFi Settings Web Page](#)
- [WAN Settings Web Page](#)

## WiFi Settings Web Page

This page shows all the Wi-Fi setup parameters and status:

## Wi-Fi Settings

Parameter	Description	Default Setting
<b>Basic Settings (DeviceInfo.WiFi.Basic.)</b>		
<b>Enable</b>	Enables OBiWiFi feature. You must have an OBiWiFi dongle attached to the phone to use this feature.	True
<b>PreferredAccessPoint</b>	Indicates which access point to use when more than one remembered access point are in range. Select from the list: <ul style="list-style-type: none"> <li>• None</li> <li>• Access Point 1</li> <li>• Access Point 2</li> <li>• ...</li> <li>• Access Point 20.</li> </ul> This value is automatically populated with the last access point that the phone user chose to connect explicitly from the native phone web page.	None
<b>ShowAccessPointPassword</b>	Check this box and press <b>Submit</b> to show all the access point passwords in (unmasked) plain text (no reboot required). The passwords are masked again following a reboot of the device.	False
<b>ScanResultWaitDuration</b>	Specifies the WiFi Scan time, from 3 to 30 seconds. This parameter can also be changed from the Phone interface under the <b>Settings/WiFi Settings</b> menu.	6
<b>Internet Settings (DeviceInfo.WiFi.)</b>		
<b>AddressingType</b>	The method to assign an IP address to this interface. Choice of: <ul style="list-style-type: none"> <li>• DHCP</li> <li>• Static.</li> </ul>	DHCP
<b>IPAddress</b>	The IP address to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>SubnetMask</b>	The subnet mask to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>DefaultGateway</b>	The default gateway to use if <b>AddressingType</b> = <i>Static</i> .	None
<b>DNSServer1</b>	An additional DNS Server to use in addition to the ones received from DHCP.	None
<b>DNSServer2</b>	An additional DNS Server to use in addition to the ones received from DHCP.	None
<b>802_1XMode</b>	Port-based network access control provides an authentication mechanism to attach to a LAN. Choice of: <ul style="list-style-type: none"> <li>• Disable</li> <li>• MDS</li> <li>• TLS</li> <li>• TTLS-MSCHAPv2</li> <li>• PEAP-MSCHAPv2</li> </ul>	Disable

**Wi-Fi Settings (continued)**

Parameter	Description	Default Setting
<b>802_1XIdentity</b>	User name for 802.1X authentication.	None
<b>802_1XPassword</b>	Password for 802.1X authentication.	None
<b>802_1XAnonymousID</b>	If blank, the anonymous identity isn't used in authentication.	None
<b>802_1XTLSecurityProfile</b>	The TLS platform profile to use for device certification.	1
<b>Access Point <i>n</i> (DeviceInfo.WiFi.AP.n.), <i>n</i> = 1, 2, ..., 20</b>		
<b>SSID</b>	SSID of the access point.	None
<b>Password</b>	Password or pass-phrase based on the authentication method used by the access point. For WPA, the pass-phrase should be no more than 64 characters. For WEP, the password should be in one of the four formats: 10 HEX digits, 26 HEX digits, 5 ASCII characters, or 13 ASCII characters. The HEX digits can be upper or lower case.	None
<b>SecurityEnabled</b>	This read-only parameter indicates if the access point has security enabled or not.	No

## WiFi Access Points Web Page

The **WiFi Access Points** page offers a familiar user interface to let you scan for access points in your vicinity. You can click on one of the available access points to connect to a WiFi network. If the access point requires authentication but the phone does not have valid credentials, a page prompts you to enter a password or pass-phrase and press **Connect** to continue.

If your access point does not show up as a listed device on this page, for example, its SSID isn't broadcast. You can enter its SSID and security credentials manually by clicking the **Add a Network** link. The **Manage Networks** link takes you back to the **Wi-Fi Settings** page. The **Scan For Networks** link reloads this page so you can rescan for access points in your vicinity.

## System Management Parameters

The System Management parameters web pages show network parameters on your phone. They include these pages:

- [WAN Settings Web Page](#)
- [Auto Provisioning Web Page](#)
- [Device Admin Web Page](#)
- [Device Update Web Page](#)

## WAN Settings Web Page

The **WAN Settings** page lists the Ethernet settings for your phone.

## WAN Settings

Parameter	Description	Default Setting
<i>Internet Settings (DeviceInfo.WAN.)</i>		
<b>AddressingType</b>	The method used for assigning IP address, subnet mask, default gateway, and so forth, to the device. Choice of: <ul style="list-style-type: none"> <li><b>DHCP</b>: IP address, subnet mask, and default gateway are assigned by DHCP Server.</li> <li><b>Static</b>: IP address, subnet mask, and default gateway are taken from the manually configured values.</li> </ul>	DHCP
<b>IPAddress</b>	The IP address assigned to the phone when <b>AddressingType</b> is set to <i>Static</i> .	None
<b>SubnetMask</b>	The subnet mask to use when <b>AddressingType</b> is set to <i>Static</i> .	None
<b>DefaultGateway</b>	The default gateway IP address assigned to the phone when <b>AddressingType</b> is set to <i>Static</i> .	None
<b>DNSServer1</b>	IP address of the first DNS server to use, in addition to the ones obtained from the DHCP server when DHCP is also enabled. If <b>AddressingType</b> is set to <i>Static</i> , the phone only uses <b>DNSServer1</b> and <b>DNSServer2</b> for DNS lookup. It tries as many as 5 DNS servers when attempting to resolve a domain name. <b>DNSServer1</b> and <b>DNSServer2</b> are tried first, whichever is specified, and then the ones obtained from the DHCP server if available.	None
<b>DNSServer2</b>	IP address of the second DNS server to use, in addition to the ones obtained from the DHCP server when DHCP is also enabled. If <b>AddressingType</b> is set to <i>Static</i> , the phone only uses <b>DNSServer1</b> and <b>DNSServer2</b> for DNS lookup. It tries as many as 5 DNS servers when attempting to resolve a domain name. <b>DNSServer1</b> and <b>DNSServer2</b> are tried first, whichever is specified, and then the ones obtained from the DHCP server if available.	None
<b>VLANEnable</b>	Enables VLAN operation. When enabled, your phone allows only packets with the <b>VLANID</b> to pass, and inserts an 802.1Q header according to <b>VLANID</b> and <b>VLANPriority</b> to all outbound packets.	False
<b>VLANID</b>	Valid range is 0 – 4094 (4095 is reserved). 0 means VLAN is disabled and egress packets aren't tagged by the device. This setting applies to all packets sent by the device.	0
<b>VLANPriority</b>	Valid choices are 0 – 7. This setting applies to all packets sent by the device.	0



## WAN Settings (continued)

Parameter	Description	Default Setting
<b>VLANDiscovery</b>	Enables taking VLAN setting from DHCP options. Choice of: <ul style="list-style-type: none"> <li>Disabled</li> <li>Fixed</li> <li>Custom</li> </ul>	Disabled
<b>VLANDiscoveryOption</b>	Specifies which DHCP option to use for VLAN discovery.	129
<b>802_1XMode</b>	Authentication is enabled or disabled with the selected mode. Choice of: <ul style="list-style-type: none"> <li>Disable</li> <li>MD5</li> <li>TLS</li> <li>TTLS-MSCHAPv2</li> <li>PEAP-MSCHAPv2</li> </ul>	Disable
<b>802_1XIdentity</b>	Identity.	None
<b>802_1XPassword</b>	Password for EAP-MD5, EAP-TTLS Private key, and EAP-TTLS/MSCHAPv2 modes.	None
<b>802_1XAnonymousID</b>	Anonymous ID. If empty, anonymous ID isn't used in authentication.	None
<b>802_1XTLSecurityProfile</b>	Security profile for 802.1X authentication. Choices are 1 or 2.	1
<b>LLDP-MED</b>	Enables LLDP-MED discovery.	True
<b>LLDP-MEDExclusivePeriod</b>	Number of seconds for LLDP-MED discovering Network Policy exclusively before the IP is established according to <b>AddressType</b> . Choice of: <ul style="list-style-type: none"> <li>0</li> <li>1</li> <li>2</li> <li>5</li> <li>10</li> <li>20</li> <li>30</li> <li>60</li> </ul>	5
<b>LLDP-MEDAssetID</b>	Customizable <b>AssetID</b> to be included in Inventory Management TLV. The default is the OBi number.	§DSN
<b>CDP</b>	Enable Cisco Discovery Protocol (CDP).	True
<b>OpenSSLCiphers</b>	<b>OpenSSLCiphers</b> provides support for all SSL/TLS connections. An empty value tells the phone to use the DEFAULT ciphers. A valid value must start with DEFAULT: or HIGH:.	Empty

**WAN Settings (continued)**

Parameter	Description	Default Setting
<b>Switch Port</b>		
<b>Name</b>	SW	
<b>Speed</b>	<p>Link speed selection mode for the port that connects your phone to the internet. This port must be connected for your phone to work. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto</li> <li>• 100 Full</li> <li>• 100 Half</li> <li>• 10 Full</li> <li>• 10 Half</li> <li>• 1000 Full</li> <li>• Disabled</li> </ul> <p>Poly recommends using only the <code>Auto</code> setting for this parameter.</p>	Auto
<b>PC Port</b>		
<b>Name</b>	PC	
<b>Speed</b>	<p>Link speed selection mode for the port that connects your phone to your PC. This port can be left disconnected. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto</li> <li>• 100 Full</li> <li>• 100 Half</li> <li>• 10 Full</li> <li>• 10 Half</li> <li>• 1000 Full</li> <li>• Disabled</li> </ul> <p>Poly recommends using only the <code>Auto</code> or <code>Disabled</code> settings for this parameter.</p>	Auto
<b>Local Time (DeviceInfo.Time.)</b>		
<b>CurrentLocalTime</b>	Current local date and time of the phone (read-only).	
<b>CurrentNTPServer1</b>	Hostname or IP address of the current first NTP server.	None
<b>CurrentNTPServer2</b>	Hostname or IP address of the current second NTP server.	None
<b>Time Service Settings (DeviceInfo.Time.)</b>		
<b>NTPServer1</b>	Hostname or IP address of the first NTP server.	ntp.polycom.com
<b>NTPServer2</b>	Hostname or IP address of the second NTP server.	None
<b>NTPServerTimeCache</b>	Cache of the latest accumulated NTP time.	None

## WAN Settings (continued)

Parameter	Description	Default Setting
<b>LocalTimeZone</b>	<p>Local time zone. Choice of:</p> <ul style="list-style-type: none"> <li>• GMT-12:00 (Int'l Dateline West)</li> <li>• GMT-11:00 (Samoa)</li> <li>• GMT-10:00 (Hawaii)</li> <li>• GMT-09:00 (Alaska)</li> <li>• GMT-08:00 (Pacific Time)</li> <li>• GMT-07:00 (Mountain Time)</li> <li>• GMT-06:00 (Central Time)</li> <li>• GMT-05:00 (Eastern Time)</li> <li>• GMT-04:00 (Atlantic Time)</li> <li>• GMT-03:30 (Newfoundland)</li> <li>• GMT-03:00 (Buenos Aires, Greenland)</li> <li>• GMT-02:00 (Mid-Atlantic)</li> <li>• GMT-01:00</li> <li>• GMT+00:00 (London, Lisbon)</li> <li>• GMT+01:00 (Rome, Paris, Madrid)</li> <li>• GMT+02:00 (Athens, Cairo)</li> <li>• GMT+03:00 (Moscow, Baghdad)</li> <li>• GMT+04:00 (Abu Dhabi)</li> <li>• GMT+04:30 (Kabul)</li> <li>• GMT+05:00 (Islamabad, Karachi)</li> <li>• GMT+05:30 (New Delhi)</li> <li>• GMT+05:45 (Kathmandu)</li> <li>• GMT+06:00</li> <li>• GMT+07:00 (Bangkok, Jakarta)</li> <li>• GMT+08:00 (Beijing, HK, Singapore)</li> <li>• GMT+09:00 (Tokyo, Seoul)</li> <li>• GMT+10:00 (Sydney, Guam)</li> <li>• GMT+11:00 (Solomon Is.)</li> <li>• GMT+12:00 (Fiji, Auckland)</li> </ul>	GMT-08:00
<b>DaylightSavingTimeEnable</b>	Enables Daylight Saving Time on the phone.	True
<b>DaylightSavingTimeStart</b>	<p>Daylight Saving Time Start Date. Format: month/day/weekday/hh:mm:ss, where month=1-12, day=±(1-31), weekday=0,1-7 (0=special, 1=Monday, 7=Sunday), hh=0-23, mm=0-59, ss=0-59.</p> <p>If weekday=0, daylight saving starts on the given month/day. Otherwise, it starts on the weekday on or after the given month/day if day &gt; 0, or on the weekday on or before the last day of the given month+day+1. Note that day = -1 is equivalent to the last day of the month.</p> <p>:ss can be omitted if its value is 0. :mm:ss can be omitted if mm and ss are both 0.</p>	3/8/7/2

**WAN Settings (continued)**

Parameter	Description	Default Setting
<b>DaylightSavingTimeEnd</b>	Daylight Saving Time End Date. Same format as Start Date.	11/1/7/2
<b>DaylightSavingTimeDiff</b>	Amount of time to add to current time during Daylight Saving Time. Format: [-]hh:mm:ss. :ss can be omitted if it is 0. :mm:ss can be omitted if both are 0.	1

***DHCP Client Settings (X\_DHCPClient.)***

<b>Hostname</b>	Device Hostname to be sent to server.	\$DM
<b>ExtraOptions</b>	Comma-separated list of extra DHCP options to be requested. Choice of: <ul style="list-style-type: none"> <li>• 42</li> <li>• 66</li> <li>• 150</li> <li>• 159</li> <li>• 160</li> <li>• 161</li> </ul>	66, 42

***DNS Control (X\_DNSControl.)***

<b>DNSQueryOrder</b>	Controls the order in querying DNS Servers. Choice of: <ul style="list-style-type: none"> <li>• DNS Server1, DNS Server 2, DHCP Offered DNS Servers</li> <li>• DHCP Offered DNS Servers, DNS Server1, DNS Server 2</li> <li>• DNS Server1, DNS Server 2</li> </ul>	DNS Server1, DNS Server 2, DHCP Offered DNS Servers
<b>DNSQueryDelay</b>	Controls the delay (in seconds) before trying the next DNS server when resolving a domain name. Choice of: 0, 1, 2, 3, 4, and 5.	2

## WAN Settings (continued)

Parameter	Description	Default Setting
<i>Local DNS Records (X_LocalDNSRec.)</i>		
<b>N</b>  where N = 1 – 32	<p>One of 32 Local DNS Records (numbered 1 – 32). Each record is a mini script of either of the following formats:</p> <ul style="list-style-type: none"> <li>Name=A, A, A, . . .</li> <li>Name=R, R, R, . . .</li> </ul> <p>where Name represents the domain name to be resolved locally, and has the format <code>prefix+domain</code> (such as <code>machine.sip+obihai.com</code>). Everything after '+' is considered as the domain to be appended to the host field in each R on the right hand side. '+' is optional. If missing, the full domain must be used in every R.</p> <p>A represents an A record, which is just an IP address, such as 192.168.12.17.</p> <p>R represents an SRV record and has the format: {host:port,pri,wt} where</p> <ul style="list-style-type: none"> <li>host is a hostname with or without domain part (such as xyz, xyz.abc.com.). A dot (.) at the end of host indicates it is a complete hostname that does not require the domain to be appended.</li> <li>port is a port number (such as 5060). port is optional. The default to use is based on the protocol (5060 for SIP, 80 for HTTP, and so forth).</li> <li>pri is the priority. Valid value is 0 (highest) – 65535 (lowest).</li> <li>wt is the weight. Valid value is 0 (lowest) – 65535 (highest). wt is optional. 1 is the default if not specified. pri is optional only if wt isn't specified. 1 is the default if not specified.</li> </ul> <p>The enclosing curly brackets { } are also optional if there is only one R, or if no comma appears inside the R.</p> <p>Examples:</p> <pre>_sip._udp+obihai.com=abc,xyz,pqr:5080,{mm m,2},{super.abc.com.}  abc.obihai.com=192.168.15.118,192.168.15. 108</pre> <p>Note: If the A record of a given hostname cannot be found in any of the local DNS records, the phone attempts to resolve it using external DNS queries. Any change applied to a local DNS record needs a reboot to take effect.</p>	None

## Auto Provisioning Web Page

The **Auto Provisioning** web page shows all the parameters related to remote provisioning of the device, as shown in the following table. Provisioning is an important topic for deployment by service providers. Please see the *Polycom OBi Edition Deployment Guide* for details on phone provisioning.

## Auto Provisioning Settings

Parameter	Description	Default Setting
<b>System Info(X_DeviceManagement.SystemInfo.)</b>		
<b>X_EnableTR69</b>	Enables TR69 provisioning.	False
<b>X_Enable3CXPNP</b>	Enables 3CX Plug-and-Play support. This option lets the phone broadcast a SIP Subscribe to the Event: ua-profile on each reboot. With this enabled, the phone also listens to SIP Notify of the same Event to extract from it the provisioning URL to request configuration, but only if the current <b>ITSP Provisioning – ConfigURL</b> parameter is still set to the default value (default flag on the device native web page is checked).	False
<b>Auto Firmware Update (X_DeviceManagement.FirmwareUpdate.)</b>		
<b>Method</b>	<p>Current operational method of auto firmware updating. Choice of:</p> <ul style="list-style-type: none"> <li>• <b>Disabled</b> = Do not check for firmware upgrade from <b>FirmwareURL</b>.</li> <li>• <b>System Start</b> = Check for firmware upgrade from <b>FirmwareURL</b> just once on system start</li> <li>• <b>Periodically</b> = Check for firmware upgrade from <b>FirmwareURL</b> on system start, and then periodically at the interval specified in the Interval parameter</li> <li>• <b>Time Of Day</b> = Download from <b>ConfigURL</b> once a day at the time specified in the <b>TimeofDay</b> parameter.</li> </ul> <p>Note: First firmware upgrade check on system start is performed after a random delay of 0 to 30 seconds.</p>	Disabled
<b>Interval</b>	When <b>Method</b> is set to <b>Periodically</b> , this is the number of seconds between each checking of firmware upgrade check from <b>FirmwareURL</b> . If value is 0, the phone checks once only on system start, the equivalent to setting <b>Method</b> to <b>System Start</b> .	0
<b>TimeofDay</b>	<p>The time during the day at which to execute <b>ConfigURL</b> is the <b>Method</b> is set to <b>Time Of Day</b>. It must be specified in the format: hh:mm[+rr] where:</p> <ul style="list-style-type: none"> <li>• hh: 0 - 23</li> <li>• mm: 0 - 59</li> <li>• +rr: (optional) maximum range of random delay in minutes from 0 - 360. If not specified, the default is 30 minutes. Setting rr = 0 removes the random delay.</li> </ul>	00:00+30
<b>RandomDelayRange</b>	The random time delay range (in seconds) inserted before the first attempt only.	30
<b>FirmwareURL</b>	URL of firmware package. URL must include scheme. Supported schemes are http:// and tftp://.	None
<b>TLSSecurityProfile</b>	Security profile when using HTTPS. Choices are 1 or 2.	1

## Auto Provisioning Settings (continued)

Parameter	Description	Default Setting
<b>DnsLookupType</b>	The type of DNS record to query. Choice of: <ul style="list-style-type: none"> <li>A Record Only</li> <li>SRV Record Only</li> <li>Try Both</li> </ul>	A Record Only
<b>DnsSrvPrefix</b>	Prepends service prefix to domain name when querying DNS. Choice of: <ul style="list-style-type: none"> <li>No Prefix</li> <li>With Prefix</li> <li>Try Both</li> </ul>	No Prefix
<b>Username</b>	Optional user name for authentication if URL scheme is http:// or https://.	None
<b>Password</b>	Optional password for authentication if URL scheme is http:// or https://.	None
<b>RangeDownload</b>	If HTTP/HTTPS is specified in <b>FirmwareURL</b> , your phone tries to download a portion of the image that needs to be updated using HTTP range header, instead of the entire image, unless the server doesn't support the range header.	False
<b>Suspend</b>	Suspends firmware update until canceled.	False
<b>Precheck</b>	Performs firmware precheck.	True

*ITSP Provisioning (X\_DeviceManagement.ITSPProvisioning.)*

<b>Method</b>	Current operational method of Provisioning. Choice of: <ul style="list-style-type: none"> <li>Disabled = Do not download from <b>ConfigURL</b></li> <li>System Start = Download from <b>ConfigURL</b> just once on system start</li> <li>Periodically = Download from <b>ConfigURL</b> on system start, and then periodically at the interval specified in the Interval parameter</li> <li>Time Of Day = Download from <b>ConfigURL</b> once a day at the time specified in the <b>TimeOfDay</b> parameter</li> </ul> <p>Note: First download on system start is performed after a random delay of 30 to 90 seconds. If there is a firmware update scheduled at the beginning. Or a random delay of 10 to 70 seconds.</p>	System Start
<b>Interval</b>	When <b>Method</b> is set to <i>Periodically</i> , this is the number of seconds between download from <b>ConfigURL</b> . If value is 0, the phone downloads once only on system start, the equivalent to setting <b>Method</b> to <i>System Start</i> .	0

## Auto Provisioning Settings (continued)

Parameter	Description	Default Setting
<b>TimeofDay</b>	The time during the day at which to execute <b>ConfigURL</b> is the <b>Method</b> is set to <code>Time Of Day</code> . It must be specified in the format: <code>hh:mm[+rrr]</code> where: <ul style="list-style-type: none"> <li>• <code>hh</code>: 0 – 23</li> <li>• <code>mm</code>: 0 – 59</li> <li>• <code>+rrr</code>: (optional) maximum range of random delay in minutes from 0 – 360. If not specified, the default is 30 minutes. Setting <code>rr = 0</code> removes the random delay.</li> </ul>	00:00+30
<b>ConfigURL</b>	URL of config file.	<code>tftp://\$DHCP OPT66 /\$DM.xml;\$DHCP OPT66;</code> Note: Default value is blank for OBiTALK provisioning.
<b>DnsLookupType</b>	The type of DNS record to query. Choice of: <ul style="list-style-type: none"> <li>• A Record Only</li> <li>• SRV Record Only</li> <li>• Try Both</li> </ul>	A Record Only
<b>DnsSrvPrefix</b>	Prepends service prefix to domain name when querying DNS. Choice of: <ul style="list-style-type: none"> <li>• No Prefix</li> <li>• With Prefix</li> <li>• Try Both</li> </ul>	No Prefix
<b>Override</b>	Defines the local settings that can be overridden by this parameter. Choice of: <ul style="list-style-type: none"> <li>• All</li> <li>• All except user settings</li> </ul>	All
<b>GPRM0 to GPRM7</b>	Non-volatile generic parameters that can be referenced in other parameters, such as <b>ConfigURL</b> .	None
<b>TPRM0 to TPRM3</b>	Temporary variables used in scripts for <b>ConfigURL</b> . Please refer to the <i>Polycom OBi Edition Deployment Guide</i> for examples on how to use these variables.	None
<b>OBiTALK Provisioning (X_DeviceManagement.Provisioning.)</b>		
<b>ZeroTouch</b>	Enables ZeroTouch provisioning.	True



## Auto Provisioning Settings (continued)

Parameter	Description	Default Setting
<b>Method</b>	<p>Current operational method of provisioning. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>Disabled</code> = Do not download from <b>ConfigURL</b></li> <li>• <code>System Start</code> = Download from <b>ConfigURL</b> just once on system start</li> <li>• <code>Periodically</code> = Download from <b>ConfigURL</b> on system start, and then periodically at the interval specified in the <code>Interval</code> parameter</li> <li>• <code>Time Of Day</code> = Download from <b>ConfigURL</b> once a day at the time specified in the <code>TimeofDay</code> parameter</li> </ul> <p>Note: First download on system start is performed after a random delay of 30 – 90 seconds. If there is a firmware update scheduled at the beginning. Or a random delay of 10 – 70 seconds.</p>	Disabled
<b>Interval</b>	<p>When <b>Method</b> is set to <code>Periodically</code>, this is the number of seconds between download from <b>ConfigURL</b>. If value is 0, the phone downloads once only on system start, the equivalent to setting <b>Method</b> to <code>System Start</code>.</p>	0
<b>TimeofDay</b>	<p>The time during the day at which to execute <b>ConfigURL</b> if the <b>Method</b> is set to <code>Time Of Day</code>. It must be specified in the format: <code>hh:mm[+rr]</code> where:</p> <ul style="list-style-type: none"> <li>• <code>hh</code>: 0 – 23</li> <li>• <code>mm</code>: 0 – 59</li> <li>• <code>+rr</code>: (optional) maximum range of random delay in minutes from 0 – 360. If not specified, the default is 30 minutes. Setting <code>rr = 0</code> removes the random delay.</li> </ul>	00:00+30
<b>ConfigURL</b>	URL of config file.	None
<b>DnsLookupType</b>	<p>The type of DNS record to query. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>A Record Only</code></li> <li>• <code>SRV Record Only</code></li> <li>• <code>Try Both</code></li> </ul>	A Record Only
<b>DnsSrvPrefix</b>	<p>Prepends service prefix to domain name when querying DNS. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>No Prefix</code></li> <li>• <code>With Prefix</code></li> <li>• <code>Try Both</code></li> </ul>	No Prefix
<b>Override</b>	<p>Defines the local settings that can be overridden by this parameter. Choice of:</p> <ul style="list-style-type: none"> <li>• <code>All</code></li> <li>• <code>All except user settings</code></li> </ul>	All
<b>GPRM0 to GPRM7</b>	Non-volatile generic parameters that can be referenced in other parameters, such as <b>ConfigURL</b> .	None

## Auto Provisioning Settings (continued)

Parameter	Description	Default Setting
<b>TPRM0 to TPRM3</b>	Temporary variables used in scripts for <b>ConfigURL</b> . Please refer to the <i>Polycom OBi Edition Deployment Guide</i> for examples on how to use these variables	None
<b>Phone Customization Data Package (X_DeviceManagement.ITSPCustomizationUpdate)</b>		
<b>Note that this parameter group name does not end with a period.</b>		
<b>Method</b>	<p>Current operational method of provisioning. Choice of:</p> <ul style="list-style-type: none"> <li>Disabled = Do not download from <b>ConfigURL</b></li> <li>System Start = Download from <b>ConfigURL</b> just once on system start</li> <li>Periodically = Download from <b>ConfigURL</b> on system start, and then periodically at the interval specified in the Interval parameter</li> <li>Time Of Day = Download from <b>ConfigURL</b> once a day at the time specified in the TimeofDay parameter</li> </ul> <p>Note: The first download on system start is performed after a random delay of 30 to 90 seconds if there is a firmware update scheduled at the beginning, else a random delay of 10 to 70 seconds.</p>	Disabled
<b>Interval</b>	When <b>Method</b> is set to <i>Periodically</i> , this is the number of seconds between download from <b>ConfigURL</b> . If value is 0, the phone downloads once only on system start, the equivalent to setting <b>Method</b> to <i>System Start</i> .	0
<b>TimeofDay</b>	<p>The time during the day at which to execute <b>ConfigURL</b> is the <b>Method</b> is set to <i>Time Of Day</i>. It must be specified in the format: hh:mm[+rr] where:</p> <ul style="list-style-type: none"> <li>hh: 0 – 23</li> <li>mm: 0 – 59</li> <li>+rr: (optional) maximum range of random delay in minutes from 0 – 360. If not specified, the default is 30 minutes. Setting rr = 0 removes the random delay.</li> </ul>	00:00+30
<b>DownloadURL</b>	URL of file to be downloaded.	None
<b>MD5Checksum</b>	MD5 checksum of the downloaded file.	None
<b>Incremental</b>	When disabled, old data is erased before new update is installed.	False
<b>DnsLookupType</b>	<p>The type of DNS record to query. Choice of:</p> <ul style="list-style-type: none"> <li>A Record Only</li> <li>SRV Record Only</li> <li>Try Both</li> </ul>	A Record Only

**Auto Provisioning Settings (continued)**

Parameter	Description	Default Setting
<b>DnsSrvPrefix</b>	Prepends service prefix to domain name when querying DNS. Choice of: <ul style="list-style-type: none"> <li>No Prefix</li> <li>With Prefix</li> <li>Try Both</li> </ul>	No Prefix
<b>Username</b>	Optional user name for authentication if URL scheme is http:// or https://	None
<b>Password</b>	Optional password for authentication if URL scheme is http:// or https://.	None
<b>User Defined Macro n (X_DeviceManagement.X_UserDefinedMacro.n.), n = 0, 1, 2, 3 (\$UDM0 – \$UDM3)</b>		
<b>Value</b>	Any plain text, or reference to another parameter's full TR-104 name prepended by a '\$'	None
<b>ExpandIn</b>	A comma-separated list of parameters that are allowed to use this macro expansion. Each parameter must be specified using its full TR-104 name. Specify ANY to allow it in all parameters.	None
<b>SyntaxCheckResult</b>	This read-only status value regards the syntax of the UDM. Pass means that this UDM is valid. Otherwise, it shows the syntax error detected by the phone either in the <b>Value</b> or <b>ExpandIn</b> parameters of the UDM.	Disabled
<b>Statistics Reporting (X_DeviceManagement.Monitoring.)</b>		
<b>SyncCQM</b>	Enables sending anonymous call quality reports to PDMS-SP in JSON format. Caller-ID information is excluded.	Disabled

## \$MACRO Expansion Supported by the Phone

**Supported Macros**

Macro Name	Description	Where It Can Be Used
MAC	Phone MAC address in upper case, such as 9CADEF000000.	ANY
MACC	Phone MAC address in upper case with colons, such as 9C:AD:EF:00:00:00.	ANY
mac	Phone MAC address lower case, such as 9c:ad:ef:00:00:00.	ANY
macc	Phone MAC address lower case, with colons, such as 9c:ad:ef:00:00:00.	ANY

**Supported Macros (continued)**

Macro Name	Description	Where It Can Be Used
FWV	Firmware version, such as 1.0.3.1626.	ANY
HWV	Hardware version, such as 2.8.	ANY
IPA	Phone current IP address, such as 192.168.15.100.	ANY
DM	Phone model name, such as VVX 350.	ANY
DMN	Phone model number, such as 350.	ANY
OBN	Phone OBi number, such as 200 123 456.	ANY
DSN	Phone serial number, such as 88B01NA00000.	ANY
GPRMn n=0–7	Value of <b>Auto Provisioning</b> ::GPRMn.	<b>Auto Provisioning</b> ::ConfigURL, <b>Auto Firmware Update</b> ::FirmwareURL
TPRMn n=0-3	Value of <b>Auto Provisioning</b> ::TPRMn.	<b>Auto Provisioning</b> ::ConfigURL, <b>Auto Firmware Update</b> ::FirmwareURL
UDMn, n=0-3	Value of <b>User-Defined Macro n</b> ::Value.	Value of <b>User-Defined Macro n</b> ::ExpandIn
DHCPOPT66	Value of DHCP option 66 assigned by server.	ANY
DHCPOPT150	Value of DHCP option 150 assigned by server.	ANY
DHCPOPT159	Value of DHCP option 159 assigned by server.	ANY
DHCPOPT160	Value of DHCP option 160 assigned by server.	ANY
DHCPOPT161	Value of DHCP option 161 assigned by server.	ANY

## Device Admin Web Page

The **Device Admin** web page includes the following configuration parameters.

**Device Admin Settings**

Parameter	Description	Default Setting
<b>Web Server (X_DeviceManagement.WebServer.)</b>		
Port	Web server port number.	80

Device Admin Settings (continued)

Parameter	Description	Default Setting
<b>PasswordMinimumLength</b>	Specifies the minimum length for both the admin and user passwords.	0
<b>AdminPassword</b>	Administrator password, case sensitive.	admin
<b>AdminPin</b>	Administrator PIN to enter the updater menu, case sensitive.	
<b>UserPassword</b>	User password, case sensitive.	user
<b>LCDScreenShot</b>	Enables creating LCD screenshot from the option in Device Update page.	False
<b>UserMenuDeviceAdmin</b>		True
<b>UserMenuDeviceUpdate</b>		True

*Syslog (X\_DeviceManagement.Syslog.)*

<b>Server</b>	IP address of the Syslog Server where the phone sends syslog debug messages to. If the value is blank, syslog is disabled.	None
<b>Port</b>	Syslog Server Port Number.	514
<b>Level</b>	Syslog Message Level.	7
<b>TAG</b>	An arbitrary string to add to the beginning of each syslog message.	None
<b>ReportingEnable</b>	Buffers syslog locally and uploads it periodically to a server according to <b>ReportingInterval</b> and <b>ReportingURL</b> .	False
<b>ReportingInterval</b>	Periodic buffered syslog upload interval in seconds. Range is 30 seconds to 120 seconds.	60
<b>ReportingURL</b>	URL to upload locally buffered syslog.	None
<b>ReportingUTCTimeStamp</b>	Include UTC time stamp (TR69 ISO 8601) in each message.	True
<b>LevelDspObi</b>	Log level for OBi/DSP interface. Choice of: <ul style="list-style-type: none"> <li>• Debug</li> <li>• Event 1</li> <li>• Event 2</li> <li>• Event 3</li> <li>• Minor Error</li> <li>• Major Error</li> <li>• Fatal Error</li> </ul>	Event 3

## Device Admin Settings (continued)

Parameter	Description	Default Setting
<b>LevelAvm</b>	Log level for AVM. Choice of: <ul style="list-style-type: none"> <li>• Debug</li> <li>• Event 1</li> <li>• Event 2</li> <li>• Event 3</li> <li>• Minor Error</li> <li>• Major Error</li> <li>• Fatal Error</li> </ul>	Minor Error
<b>LevelSps</b>	Log level for SSPS. Choice of: <ul style="list-style-type: none"> <li>• Debug</li> <li>• Event 1</li> <li>• Event 2</li> <li>• Event 3</li> <li>• Minor Error</li> <li>• Major Error</li> <li>• Fatal Error</li> </ul>	Minor Error
<b>LevelDrvCodec</b>	Log level for DrvCodec. Choice of: <ul style="list-style-type: none"> <li>• Debug</li> <li>• Event 1</li> <li>• Event 2</li> <li>• Event 3</li> <li>• Minor Error</li> <li>• Major Error</li> <li>• Fatal Error</li> </ul>	Minor Error
<b>LevelUsb</b>	Log level for USB. Choice of: <ul style="list-style-type: none"> <li>• Debug</li> <li>• Event 1</li> <li>• Event 2</li> <li>• Event 3</li> <li>• Minor Error</li> <li>• Major Error</li> <li>• Fatal Error</li> </ul>	Minor Error
<b>HTTP Client (X_DeviceManagement.HTTPClient.)</b>		
<b>UserAgent</b>	The value of User-Agent request-header field.	\$DM
<b>TimeOut</b>	HTTP request timeout setting in seconds. This value shall be greater than 60 seconds.	600
<b>ProxyServer</b>	Host name or IP address of the HTTP proxy server.	None

## Device Admin Settings (continued)

Parameter	Description	Default Setting
<b>ProxyServerPort</b>	Destination port to connect to the HTTP proxy server. Range = [0:65535]. Do not choose a port at random.	80
<b>ProxyAuthUsername</b>	User name for proxy authentication.	None
<b>ProxyAuthPassword</b>	Password for proxy authentication.	None
<b>BypassProxyServerForLocalAddresses</b>	Enables the option to bypass the HTTP Proxy server when requests are sent to addresses in the same subnet.	False
<b>BypassProxyForSubnets</b>	List of intranet subnets that can bypass the proxy server. For example: 10.10.10.0/24, 192.168.0.0/16	None
<b>External Port Security (X_DeviceManagement.X_PortSecurity.)</b>		
<b>USB</b>	Locks the USB port so that the phone does not react to any USB device connected to it.	False
<b>PCPort</b>	Locks the PC port so that the phone does not allow any network traffic in and out of that port.	False
<b>Remote PCAP Server (X_DeviceManagement.X_RPCAPD.)</b>		
<b>Enable</b>	Enables PCAP (Packet Capture) Server function on the phone.	False
<b>Port</b>	PCAP Server port number.	2002
<b>Client</b>	List of clients allowed to connect to this server. An empty list means everyone is allowed.	None
<b>Packet Capture (X_DeviceManagement.X_PacketCapture)</b>		
<b>On</b>	Enables test.	False
<b>Status</b>	Status of capturing.	Test Result Not Available
<b>Interface</b>	The name of the interface from which to capture. Choice of: <ul style="list-style-type: none"> <li>• Primary</li> <li>• eth0</li> <li>• wlan0</li> </ul>	Primary
<b>Storage</b>	The name of the interface to listen to. Choice of: <ul style="list-style-type: none"> <li>• Internal Storage</li> <li>• OBiTalk Designated</li> </ul>	Internal Storage
<b>RestartCaptureOnReboot</b>		False
<b>PromiscuousMode</b>		True
<b>WebAccessExcluded</b>		True

Device Admin Settings (continued)

Parameter	Description	Default Setting
<b>PostponeFirmwareUpdate</b>		True
<b>Platform CA n (X_DeviceManagement.PlatformCACert.n.), n = 1, 2</b>		
<b>DownloadURL</b>	URL to download certificate.	None
<b>MD5Checksum</b>	MD5 checksum of the certificate file to be downloaded. Failure to provide this causes the phone to try to download the same file on every reboot or restart.	None
<b>CommonName</b>	The Common Name set in the installed certificate. Read-only status field.	None
<b>FingerPrint</b>	SHA1 fingerprint of the installed certificates.	None
<b>Obsolete</b>	When set to true, the certificate is deleted from the phone. Also, the certificate downloading process is ignored.	False
<b>Custom DeviceCert n (X_DeviceManagement.CustomDeviceCert.n.), n = 1, 2</b>		
<b>DownloadURL</b>	URL to download certificate.	None
<b>MD5Checksum</b>	MD5 checksum of the downloaded certificate.	None
<b>CommonName</b>	The Common Name set in the installed certificate. Read-only status field.	None
<b>FingerPrint</b>	SHA1 fingerprint of the installed certificates.	None
<b>Obsolete</b>	When set to true, the device is deleted from the phone. Also, the certificate downloading process is ignored.	False
<b>TLSPlatform Profile n (X_DeviceManagement.TLSPlatform.n.), n = 1, 2</b>		
<b>CipherSuite</b>	The cipher suite to use in a TLS profile (the encryption algorithms to support in establishing a TLS connection according to the TLS profile specification configured on the phone).	None
<b>CACertList</b>	The CA Certificate List to use in a TLS profile. Choice of: <ul style="list-style-type: none"> <li>• Default</li> <li>• Default+P1</li> <li>• Default+P2</li> <li>• All</li> <li>• Platform1</li> <li>• Platform2</li> <li>• Platform1+2</li> </ul>	Default
<b>DeviceCert</b>	The Device Certificate List to use in a TLS profile. Choice of: <ul style="list-style-type: none"> <li>• Polycom</li> <li>• Custom1</li> <li>• Custom2</li> </ul>	Polycom



## Device Update Web Page

The Device Update web page offers the following functions:

- [Firmware Update](#)
- [Backup AA User Prompts](#)
- [Screenshot](#)
- [Screenshot: Side Car](#)
- [PCAP Capture Result](#)
- [Backup Configuration](#)
- [Restore Configuration](#)
- [Reset Configuration](#)

## Firmware Update

You can upgrade the firmware for your phone from the native phone web page. The firmware file with which you want to upgrade the phone must be stored locally on a computer from which you can access with a web browser.

### Follow these steps to upgrade:

- 1 Select the **System Management – Device Update** menu on the side panel of the web page.
- 2 Click the **Browse** button in the **Firmware Update** section of the page. In a file browser window, select the firmware file.
- 3 Click the **Update** button to start the upgrade process.

The process takes about 30 seconds to complete.



Don't disconnect the power from the device during this procedure. If the new firmware is upgraded successfully, the device reboots automatically to start running the new firmware. Otherwise, the web page shows an error message explaining why the upgrade failed.

## Possible Error Messages on Firmware Update Failure

The following table lists the possible error messages encountered when a firmware upgrade fails.

### Error Messages for Firmware Update Failure

Error Message	Description	Suggested Solution
Firmware Package Checksum Error	A corrupted firmware package file was used for the update.	Check the file and / or re-download the firmware package and try again.
System Is Busy	The phone is busy because one of the phone services is in an active call or device provisioning is in progress.	Try to update again later.
Firmware Is Not Modified	The phone is already running the same firmware as the one selected for update.	No need to upgrade.

## Backup AA User Prompts

Click the **Backup** button to start backing up the AA user prompts. A popup window asks what you want to do with the `backupaa.dat` file.

## Screenshot

Click the **Snap** button to capture a screenshot of the current desktop image on your phone. This saves a file named `screenshot.bmp` to your PC. The default save location is your downloads folder, unless otherwise specified.



Enable **LCDScreenShot** parameter on the **Device Admin** web page to show the **Screenshot** function.

## Screenshot: Side Car

Click the **Snap 1** button to capture a screenshot of Side Car 1, and click the **Snap 2** button to capture a screenshot of Side Car 2.



Enable **LCDScreenShot** parameter on the **Device Admin** web page to show the **Screenshot: Side Car** function.

## PCAP Capture Result

If PCAP is enabled on your phone, click this **Extract** button to extract the PCAP capture results. This saves a file named `trace.pcap` to your PC. The default save location is your downloads folder, unless otherwise specified.

To enable PCAP, set the **X\_DeviceManagement.X\_RPCAPD.Enable** parameter to `True`. Before extracting PCAP capture data, set this parameter to `False`.

## Backup Configuration

The current configuration of the phone can be backed up and stored as a file in XML format at a user specified location. The default name of the file is `backupxxxxxxxxxxxx.xml`, where `xxxxxxxxxxxx` represents the MAC address of your phone.

When backing up a device's configuration, you can select one the following options before clicking **Backup**.

### Backup Options

Option	Description	Default Setting
Incl. Running Status	If checked, the values of all status parameters are included in backup file. Otherwise, status parameters are excluded from the backup.	No
Incl. Default Value	If checked, the default values of parameters are included in the backup file. Otherwise, default values are excluded from the backup.	No

**Backup Options (continued)**

Option	Description	Default Setting
Use OBi Version	If not checked, the backup file uses XML tags that are compliant with the TR-104 standard. Otherwise, the backup file is stored in an OBi proprietary format where the XML tags aren't compliant with TR-104, but the file size is smaller and the file is more readable.	No
Encrypt Content	If checked, the configuration data is encrypted.	No

When the file browser window opens, you can change the filename and choose the location to save the backup file.



Different web browsers may handle this differently. If the operation is blocked due to the security setting of the web browser, you should change the security setting temporarily to allow this operation to complete.

## Restore Configuration

When restoring the configuration to a previous backup copy, you need to specify the backup file you want to restore to by clicking **Browse** in the **Restore Configuration** section of the web page. Then, select the **Restore** button to start the process. The phone reboots automatically after the restoration is complete.



All passwords and PINs are excluded from the backup file. Hence, they aren't available to restore. Call history is excluded from the backup, but can be saved as an XML formatted file separately from the Call History web page.

## Reset Configuration

The **Reset Configuration** function resets the phone to its factory default condition. Call history and various statistical information is removed at the same time. Resetting the device configuration should be used with extreme caution as the operation cannot be undone.

**Reset Configuration Options**

Option	Description	Default Setting
User Data	When checked, clicking <b>Reset</b> restores all your user data to the factory defaults.	Yes
Voice Configuration	When checked, clicking <b>Reset</b> restores all your voice configuration parameters to the factory defaults.	Yes
Networking	When checked, clicking <b>Reset</b> restores all your networking settings to the factory defaults.	Yes

To do this, click **Reset** in the **Reset Configuration** section. A confirmation window pops up. On confirmation, the phone resets its configuration to its factory default condition. The phone reboots automatically when factory reset completes.

## Service Providers Parameters

The Service Providers parameters web pages show parameters for the provisioned service providers on your phone. They include these sets of pages for each of the six ITSP Profiles A through F :

- [General Web Page](#)
- [SIP Web Page](#)
- [RTP Web Page](#)

## ITSP Profile X (X = A, B, C, D, E, F)

ITSP profiles represent profiles for the service providers. Voice service profiles, described in the next set of web pages, represent the profiles that bind your phone to the service providers.

### General Web Page

The following configuration parameters are available on this page.

#### General Settings

Parameter	Description	Default Setting
<i>General ITSP Settings (VoiceService.1.VoiceProfile.k.), k = 1 – 6 for X = A – F, respectively</i>		
<b>Name</b>	Human-readable string to identify the profile instance. Maximum length = 127 characters.	None
<b>SignalingProtocol</b>	Choose among the following list of signaling protocols for this ITSP: <ul style="list-style-type: none"> <li>• SIP</li> </ul>	SIP
<b>DTMFMethod</b>	Method to pass DTMF digits to peer device. Choice of: <ul style="list-style-type: none"> <li>• Inband - DTMF tone are sent as inband audio signal</li> <li>• RFC2833 - DTMF tone events are relayed per RFC2833</li> <li>• SIPInfo - DTMF tones are relayed with SIP INFO request</li> <li>• Auto - Method to use based on call setup negotiation (either Inband or RFC2833 can be negotiated)</li> </ul>	Auto
<b>InbandDTMFVolume</b>	DTMF tone volume when sending inband DTMF. Valid values are –24dB to 0dB in 3dB steps.	–15dB
<b>X_UseFixedDurationRFC2833DTMF</b>	When relaying DTMF digit events on this trunk using RFC2833, the RFC2833 RTP packets normally keep streaming for as long as the digit is pressed. With this option set to True, the phone sends only one RTP digit event packet with a fixed duration of 150 ms regardless how long the digit has been pressed.	False
<b>DigitMap</b>	A digit map to restrict the numbers that be dialed or called with this service. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of digit map syntaxes. Maximum Length = 511 characters.	(1xxxxxxxxxx   <1> [2-9]xxxxx xxxx 011xx. xx.)

## General Settings (continued)

Parameter	Description	Default Setting
<b>STUNEnable</b>	Enables phone to send a STUN binding request for its RTP port prior to every call.	False
<b>STUNServer</b>	IP address of domain name of the STUN Server to use.	None
<b>X_STUNServerPort</b>	UDP listen port of the STUN Server.	3478
<b>X_ICEEnable</b>	Enables phone to use ICE algorithm to find the best peer RTP address to forward RTP traffic for every call.	False
<b>X_EarlyICEEnable</b>	Enables starting ICE upon 18x response when possible for outbound calls.	False
<b>X_EarlyICEEnableIn</b>	Enables sending 183 with SDP and start ICE on INVITE when possible for inbound calls.	False
<b>X_ICEExpires</b>	ICE Expires in milliseconds for outgoing calls.	2000
<b>X_ICEExpiresIn</b>	ICE Expires in milliseconds for incoming calls.	10
<b>X_IgnoreSTUNCheckError</b>	Ignores STUN check error when a call is connected.	False
<b>X_SymmetricRTPEnable</b>	Enables phone to apply symmetric RTP behavior on every call. That is, send RTP to peer at the address where incoming RTP packets are received from.	False

**Service Provider Info (VoiceService.1.VoiceProfile.k.ServiceProviderInfo.), k = 1 – 6 for X = A – F, respectively**

<b>Name</b>	Human-readable string identifying this service provider. Maximum length = 127 characters.	None
<b>URL</b>	Website of this service provider. Maximum length = 127 characters.	None
<b>ContactPhoneNumber</b>	Phone number to contact this service provider. Maximum length = 31 characters.	None
<b>EmailAddress</b>	Email address to contact this service provider. Maximum length = 127 characters.	None

## SIP Web Page

The following configuration parameters are available on this page.

### SIP Settings

Parameter	Description	Default Setting
<b>SIP (VoiceService.1.VoiceProfile.k.SIP), k = 1 – 6 for X = A – F, respectively</b>		
<b>ProxyServer</b>	Host name or IP address of the SIP proxy server.	None

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>ProxyServerPort</b>	Destination port to connect to the SIP server.	5060
<b>ProxyServerTransport</b>	Transport protocol to connect to SIP server. Choice of: <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> </ul>	UDP
<b>RegistrarServer</b>	Hostname or IP address of the SIP registrar. If a value is specified, the phone sends REGISTER to the given server. Otherwise, REGISTER is sent to <b>ProxyServer</b>	None
<b>RegistrarServerPort</b>	Destination port to connect to SIP registrar.	5060
<b>X_XsiServer</b>	Host name or IP address of the BroadSoft XSI Application Server.	False
<b>X_XsiServerPort</b>	Listening port of the BroadSoft XSI Application Server. If not specified or 0, the default port is used (80 for HPTP and 443 for HTTPS). Range = [0:65535].	0
<b>X_XsiServerScheme</b>	Scheme to access the BroadSoft XSI Application Server. Choice of: <ul style="list-style-type: none"> <li>• HTTP</li> <li>• HTTPS</li> </ul>	HTTP
<b>UserAgentDomain</b>	CPE domain string. If empty, the phone uses <b>ProxyServer</b> as its own domain to form its AOR (Address Of Record) or Public Address when constructing SIP messages (for example, in the FROM header of outbound SIP Requests). Note: If <b>SPx Service::URI</b> is specified, additional rules applied in forming the AOR. See description of the <b>URI</b> parameter for more details and examples.	False
<b>OutboundProxy</b>	Host name or IP address of the outbound proxy. Outbound proxying is disabled when this parameter is blank.	None
<b>OutboundProxyPort</b>	Destination port to be used in connecting to the outbound proxy.	5060
<b>X_OutboundProxyTransport</b>	A different SIP transport can be used by the <b>OutboundProxy</b> . Choice of: <ul style="list-style-type: none"> <li>• UDP</li> <li>• TCP</li> <li>• TLS</li> <li>• Follow ProxyServerTransport</li> </ul>	Follow ProxyServerTransport
<b>X_TLSecurityProfile</b>	Security profile to use for SIP TLS connection.	False
<b>X_UserAgentContactFollowProxyServerTransport</b>	If enabled, the user agent should use a Contact and Via transport that agrees with <b>ProxyServerTransport</b> .	False

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_BypassOutboundProxyInCall</b>	Bypasses the outbound proxy after initial call setup.	False
<b>RegistrationPeriod</b>	Nominal interval between device register in seconds.	60
<b>X_RegistrationMargin</b>	Specifies the margin to renew SIP registration before it expires. The default is to renew at half-time before the next expiration if the expires value is less than 1200 seconds, or 600 seconds before expiration otherwise. You can specify here the number of seconds before expiration to renew registration explicitly (such as 10 or 20), or as a fraction of the current register expires value (such as 0.1 or 0.25).	None
<b>TimerT1</b>	Value of SIP timer T1 in ms.	500
<b>TimerT2</b>	Value of SIP timer T2 in ms.	4000
<b>TimerT4</b>	Value of SIP timer T4 in ms.	5000
<b>TimerA</b>	Value of SIP timer A in ms.	500
<b>TimerB</b>	Value of SIP timer B in ms.	32000
<b>TimerD</b>	Value of SIP timer D in ms.	32000
<b>TimerE</b>	Value of SIP timer E in ms.	500
<b>TimerF</b>	Value of SIP timer F in ms.	32000
<b>TimerG</b>	Value of SIP timer G in ms.	500
<b>TimerH</b>	Value of SIP timer H in ms.	32000
<b>TimerI</b>	Value of SIP timer I in ms.	5000
<b>TimerJ</b>	Value of SIP timer J in ms.	32000
<b>TimerK</b>	Value of SIP timer K in ms.	5000
<b>InviteExpires</b>	Invite request Expires header value in seconds.	60
<b>ReInviteExpires</b>	Re-invite Expires header value in seconds.	10
<b>RegisterMinExpires</b>	Register Min-Expires header value in seconds (not used at the moment).	15
<b>RegisterRetryInterval</b>	Register retry interval in seconds.	30
<b>X_RegisterRetryResponseCodes</b>	A set of rules to control how many seconds to wait before retrying register after a specific failure response. The rules are specified with a digit map string. The value after the letter <i>w</i> designates the number of seconds to wait. A range can be specified with two numbers with a – in between such that a random delay within that range is used.	(<40[17]:w120> <40[34]:w120> <99[01]:w120-200> [4-9]xx)

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_RegisterIncludeInstance</b>	Enables or Disables inclusion of instance parameter in the Contact of REGISTER requests.	True
<b>DSCPMark</b>	Diffserv code outgoing SIP packets. Valid range: [0-63].	26
<b>X_SpoofCallerID</b>	Allows outbound Caller ID spoofing. If set to <code>True</code> , the phone attempts to set the caller ID name and user ID field in the FROM header to that of a remote caller in the case of a bridged call (from another trunk, such as PSTN Line or another SP Service). Otherwise, the phone always provides its own account information to form the FROM header. Note that most service providers don't allow originating a call if the FROM header field doesn't match the account credentials. Enable this option only if you are sure that the service provider allows it.	False
<b>X_UseRefer</b>	Enables using SIP REFER for call transfer. If disabled, the phone bridges the call instead when performing a call transfer, which consumes some resources on the phone.	True
<b>X_ReferAOR</b>	Enables using the target's AOR in Refer-To header of SIP REFER. If disabled, the target's Contact is used instead.	True
<b>X_HoldReferee</b>	Holds the Referee before a blind transfer if the call isn't placed on hold. This allows reconnecting with the Referee if the blind call transfer fails.	False
<b>X_Use302ToCallForward</b>	Enables the 302 response to INVITE for call forward. If disabled, the phone bridges the call legs instead when forwarding a call, and consumes some resources on the device.	True
<b>X_UserAgentName</b>	If a value is specified, the phone includes a User-Agent header in all SIP Requests or a Server header in all SIP responses that contains exactly the given value.	OBIHAI / \${DM} - \${FWV}
<b>X_ProcessDateHeader</b>	Enables the phone to decode the DATE header sent by the ITSP in a 200 response to its REGISTER. The DATE header specifies the current GMT time and the phone can use this to adjust its local time and date without relying on NTP.	True
<b>X_InsertRemotePartyID</b>	Enables the phone to include a Remote-Party-ID header in its outbound SIP INVITE to indicate to the ITSP the caller's preferred privacy setting (either full or none).	True
<b>X_EnforcePAssertedIdentity</b>	Takes Caller ID from P-Asserted-Identity header only.	False
<b>X_InsertPPreferredIdentity</b>	Inserts P-Preferred-Identity header in all outbound INVITE.	False



## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_InsertPAccessNetworkInfo</b>	Inserts P-Access-Network-Info header in REGISTER and INVITE requests.	False
<b>X_InsertPrivacyHdr</b>	Inserts a Privacy:id header in INVITE for anonymous calls.	False
<b>X_UseAnonymousFROM</b>	Enables using <code>sip:anonymous@localhost</code> in FROM header of SIP INVITE when attempting to make an anonymous call.	False
<b>X_SessionRefresh</b>	Enables session refresh signaling (with SIP Re-INVITE) during a connected call. This allows the phone to detect if the connection with the peer is broken abnormally so it can release the call. Disable this option if the ITSP does not support Re-INVITE sent from the client device.	False
<b>X_SessionTimer</b>	Enables standard session timer behavior based on RFC4028.	False
<b>X_SessionExpires</b>	Session Expires before value. If session refresh is enabled, the phone refreshes half-time before the session expires.	20
<b>X_AccessList</b>	A comma-separated list of IP addresses such that the phone only accepts SIP requests coming from one of the given addresses. If the list is empty, the phone accepts SIP requests from any IP address.	False
<b>X_InsertRTPStats</b>	Enables the phone to include a X-RTP-Stat header in a BYE request or 200 response to BYE request at the end of an established call. This header contains a summary of RTP statistics collected during the call.	True
<b>X_MWISubscribe</b>	Enables the phone to SUBSCRIBE to the message-summary event package to support MWI and VMWI service. Note that the phone handles NOTIFY of this event package regardless of whether <b>MWISubscribe</b> is enabled.	False
<b>X_MWISubscribeURI</b>	Blank implies to use the same URL as REGISTER for the TO and FROM header as well as the Request-URI Otherwise, if the URI does not contain '@', it is user as the user ID field in TO/FROM header as well as the Request-URI, which are otherwise same as REGISTER If the URI contains '@', it is used in the TO and FROM header as the phone forms the Request-URI of SUBSCRIBE the same way as the TO header, with an additional port number.	False
<b>X_MWISubscribeExpires</b>	Periodic interval to renew SUBSCRIBE.	3600
<b>X_RegSubscribe</b>	Enables subscription to the reg event package.	False

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_RegSubscribeExpires</b>	Reg event subscription expires value.	3761
<b>X_NoNonceAuth</b>	Enables sending Authorization header without being challenged first.	False
<b>X_ProxyServerRedundancy</b>	Enables proxy redundancy feature on the phone. To use this feature, phone registration must be enabled and the SIP Registration Server or Outbound Proxy Server must be configured as a domain name.	False
<b>X_SecondaryRegistration</b>	Enables phone to register with a secondary server in addition to the primary server. <b>X_ProxyServerRedundancy</b> must be enabled for this parameter to take effect	False
<b>X_CheckPrimaryFallbackInterval</b>	Interval in seconds at which the phone should check the primary fallback list of candidate servers.	60
<b>X_CheckSecondaryFallbackInterval</b>	Interval in seconds at which the phone should check the secondary fallback list of candidate servers.	60
<b>X_ProxyFailoverResponseCodes</b>	A list of failure response codes specified in the form of a digit map string to trigger proxy failover. If only one digit map is specified, it applies to REGISTER and INVITE requests. If two digit maps are provided (separated by a comma), the first one applies to REGISTER and the second to INVITE.	([5-9]xx)
<b>X_InviteFailoverWaitRegisterTimer</b>	Maximum time (in milliseconds) to wait for successful register failover to retry INVITE after a failure.	32000
<b>X_ProxyRequire</b>	If this parameter isn't blank, the phone includes a Proxy-Require header stating the value of this parameter in all SIP requests sent to the ITSP.	None
<b>X_MaxForward</b>	Value for the Max-Forward header in all SIP requests sent by the phone.	70
<b>X_AcceptLanguage</b>	If this parameter isn't blank, the phone includes an Accept-Language header stating the value of this parameter in all SIP requests sent to the ITSP.	None
<b>X_DnsSrv</b>	Enables DNS SRV Lookup for the Proxy Server or the Outbound Proxy Server.	False
<b>X_DnsSrvAutoPrefix</b>	Lets the phone automatically prepend a standard prefix to the domain name when querying DNS Server to resolve the ProxyServer or OutboundProxy name as a SRV record. The standard prefix is: <ul style="list-style-type: none"> <li>• <code>_sip._udp.</code> for SIP over UDP</li> <li>• <code>_sip._tcp.</code> For SIP over TCP</li> <li>• <code>_sip._tls.</code> for SIP over TLS</li> </ul>	False
<b>X_DnsNAPTR</b>	Enables DNS/NAPTR lookup for the Proxy Server or for the Outbound Proxy Server.	False

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_Support100rel</b>	Turns on support for RFC3262 (reliable provisional SIP responses). If enabled, the phone announces this support in a SIP Supported header, and requires a caller to use this option if the caller also supports this feature.	False
<b>X_UserEqPhone</b>	Enables inserting <code>user=phone</code> parameter in INVITE Request-URI.	False
<b>X_UseTelURI</b>	Enables using <code>tel:</code> in outbound SIP Request-URI and TO-URL.	False
<b>X_DiscoverPublicAddress</b>	Enables the phone to use the public IP address and port it has discovered as its SIP Contact address.	False
<b>X_UsePublicAddressInVia</b>	Use the public IP address and port discovered by the phone as the address in VIA header.	False
<b>X_PublicIPAddress</b>	A static public IPv4 address, if specified, is used by the phone to form its SIP Contact address.	None
<b>X_UseRport</b>	Enables the phone to insert a blank <code>rport</code> parameter in the VIA header for outbound SIP messages. This option should be turned off if you are using port forwarding on the external router to route inbound SIP messages to the phone.	True
<b>X_UseCompactHeader</b>	Enables using compact format SIP headers.	False
<b>X_IncludeMessageHash</b>	Includes a MD5 hash of all the SIP headers in an X-MD5-Hash header. A hash of the SDP is also included in an x-md5-hash SDP attribute.	False
<b>X_EchoServer</b>	A server that can echo back SIP messages to the device.	None
<b>X_EchoServerPort</b>	The echo server listening port.	5060
<b>X_EnableRFC2543CallHold</b>	Enables the phone to recognize call hold signaling as used in RFC2543.	False
<b>X_VerifyServerDomain</b>	Enables verification of server domain against its certificate on a SSL/TLS connection.	False
<b>X_RejectKeyResponseCode</b>	SIP Response Code and Phrase to inbound INVITE when you press the <b>Reject</b> key.	None
<b>X_SupportOutbound</b>	Enables support for SIP Outbound (RFC 5626).	False
<b>X_SupportPath</b>	Enables support for the SIP header path.	False
<b>X_SupportServiceRoute</b>	Enables support for the SIP header Service-Route.	False
<b>X_SupportPAssociatedURI</b>	Enables support for the SIP header P-Associated-URI.	False

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_Sticky18x</b>	Enables the behavior of ignoring further 18x responses without SDP upon receiving the first 18x with SDP to INVITE in the context of playing a local ringback tone.	False
<b>Feature Configuration (VoiceService.1.VoiceProfile.k.SIP), k = 1 – 6 for X = A – F, respectively</b>		
<b>X_CallParkMethod</b>	Select the method to use to park a call. Choice of: <ul style="list-style-type: none"> <li>Feature Code</li> <li>REFER</li> </ul>	Feature Code
<b>X_DirectedCallPickup Method</b>	Select the method to use for directed call pickup. Choice of: <ul style="list-style-type: none"> <li>Feature Code</li> <li>INVITE+Replaces</li> </ul>	Feature Code
<b>X_AutoAnswerMethod</b>	Method to signal to the called phone to auto-answer the call. Choice of: <ul style="list-style-type: none"> <li>Call-Info: Inserts <b>answer-after=0</b> parameter in a Call-Info header in the INVITE request.</li> <li>Alert-Info: Inserts <b>info=alert-autoanswer;delay=0</b> parameter in an Alert-Info header in the INVITE request.</li> </ul>	Call-Info
<b>X_ShareLineMethod</b>	Select the signaling method for share line operation. Choice of: <ul style="list-style-type: none"> <li>call-info</li> <li>dialog;sla</li> <li>dialog;ma</li> </ul>	call-info
<b>X_ShareLineStateAgent</b>	A State Agent user ID that the phone initiates a subscription to, if specified.	False
<b>X_DialogStateUpdate Filter</b>	Comma-separated list of unwanted states to not notify the state agent.	ringing,early,proceeding
<b>X_DialogSubscribeExpires</b>	BLA Dialog Subscription Renewal interval in seconds. Set to 0 to disable subscription renewal.	3600
<b>X_IgnoreDialogInfoVersion</b>	Ignore the version attribute in dialog-info tag to work around implementation that does not update version.	False
<b>X_BLFSubscribeExpires</b>	BLF Subscription Expires value, in seconds. Set to 0 to disable subscription renewal.	3600
<b>X_CallInfoSubscribeExpires</b>	CallInfo (SCA) Subscription Renewal interval in seconds. Set to 0 to disable subscription renewal.	3600
<b>X_BWCallParkSubscribeExpires</b>	x-broadworks-callpark Subscription Renewal interval in seconds. Set to 0 to disable subscription renewal.	3600
<b>X_BWCallCenterSubscribeExpires</b>	x-broadworks-callcenter Subscription Renewal interval in seconds. Set to 0 to disable subscription renewal.	3600

## SIP Settings (continued)

Parameter	Description	Default Setting
<b>X_BWHotelingSubscribeExpires</b>	x-broadworks-hoteling Subscription Renewal interval in seconds. Set X_KeepAliveMsgType to 0 to disable subscription renewal.	3600
<b>X_ASFeatureEventSubscribeExpires</b>	(BroadSoft) as-feature-event Subscription Renewal interval in seconds. Set to 0 to disable subscription renewal.	3600
<b>X_LineSeizeSubscribeExpires</b>	Line-seize event subscription renewal interval in seconds.	15
<b>Feature Codes (VoiceService.1.VoiceProfile.k.X_FeatureCode.), k = 1 – 6 for X = A – F, respectively</b>		
<b>DirectedCallPickup</b>	Code to invoke Directed Call Pickup feature on the ITSP.	*97
<b>CallPickup</b>	Code to retrieve a parked call from the ITSP.	*88
<b>BargeIn</b>	Code to barge-in an active call on another extension.	*33
<b>Whisper</b>	Code to whisper (or coach) another extension on an active call.	*48
<b>Monitor</b>	Code to monitor (or coach) another extension on an active call.	*47
<b>Park</b>	Code to park a call on the ITSP.	*68

## RTP Web Page

The following configuration parameters are available on this page.

## RTP Settings

Parameter	Description	Default Setting
<b>RTP (VoiceService.1.VoiceProfile.k.RTP), k = 1 – 6 for X = A – F, respectively</b>		
<b>LocalPortMin</b>	Base of port range for tx/rx RTP with this SP.	16600
<b>LocalPortMax</b>	Top of port range for tx/rx RTP with this SP.	16798
<b>KeepAliveInterval</b>	Interval in seconds between sending keep alive packet on an RTP channel that is currently in idle (due to call hold for instance). RTP keepalive is disabled if the value of this parameter is set to 0.	0
<b>DSCPMark</b>	Diffserv code for outgoing RTP packets with this SP.	46
<b>X_RefreshSession</b>	Allow incoming RTP packets to refresh session.	False

**RTP Settings (continued)**

Parameter	Description	Default Setting
<b>X_SymmetricMedia</b>	If incoming payload type changes unannounced, after 10 packets with the new payload type, decoding switches to the new format. If symmetric media is enabled, outgoing packets are also in the new format.	False
<b><i>RTCP (VoiceService.1.VoiceProfile.2.RTP.RTCP), k = 1 – 6 for X = A – F, respectively</i></b>		
<b>Enable</b>	Enables RTCP operation.	False
<b>TxRepeatInterval</b>	Interval in milliseconds between RTCP transmissions.	10000
<b>LocalCName</b>	The local CNAME to use in RTCP message. By default, the phone uses <code>account-userid@local-ip-address</code> as the local CNAME.	None
<b>X_RTCPMux</b>	Enables RTCP-Mux operation (send and receive RTCP on the same local port as the corresponding RTP).	False
<b>X_VqPublishEnable</b>	Enables Voice Quality (VQ) report sent to the proxy server using Publish method.	False
<b>X_VqPublishUrl</b>	A user name or URL to send VQ report using Publish method.	False
<b>X_VqPublishInterval</b>	Interval in seconds between VQ reports. 0 or {blank} disables periodic reports.	0
<b><i>Jitter Buffer (VoiceService.1.VoiceProfile.k.RTP.JIB.), k = 1 – 6 for X = A – F, respectively</i></b>		
<b>Adaptive</b>	Enables jitter buffer adaptation.	True
<b>MaximumSize</b>	Maximum jitter buffer size in ms.	250
<b>SetPoint</b>	Initial play out delay in ms.	60
<b>Target</b>	Target play out delay in ms.	20
<b>AdaptationSlope</b>	Maximum adaptation slope in samples per 10 ms.	16

## Voice Services

The Voice Services parameters web pages show parameters for the voice services that are bound to the service providers for your phone. They include these sets of pages:

- [SPn Service Web Page \(n = 1, 2, 3, 4, 5, 6\)](#)
- [OBiTALK Service Web Page](#)
- [Auto Attendant Web Page](#)
- [Gateways and Trunk Groups Web Page](#)

## ***SP<sub>n</sub> Service Web Page (n = 1, 2, 3, 4, 5, 6)***

The following configuration parameters are available on this page.

### **SP<sub>n</sub> Services Settings**

<b>Parameter</b>	<b>Description</b>	<b>Default Setting</b>
<b><i>SP<sub>n</sub> Service (VoiceService.1.VoiceProfile.1.Line.n.), n = 1 – 6</i></b>		
<b>Enable</b>	Enables this line.	True
<b>X_DisplayLabel</b>	A label that represents this service on the phone screen. Special characters must include a closing character. For example, an open curly bracket must also include a closing curly bracket.	None
<b>X_DisplayNumber</b>	A number that represents this service on the phone screen. Special characters must include a closing character. For example, an open curly bracket must also include a closing curly bracket.	None
<b>X_ServProvProfile</b>	Select a Service Provider profile for this service. Choices are A, B, C, D, E, or F.	A
<b>X_RingProfile</b>	Selects a Ring Profile to ring the phone for incoming calls on this service that are routed to it. The ringing pattern is taken from the given profile. Choices are A or B.	A
<b>X_CodecProfile</b>	Selects a Codec Profile for all calls on this service. Choices are A or B.	A
<b>X_InboundCallRoute</b>	Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the phone. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify this parameter.	ph
<b>X_RegisterEnable</b>	Enables registration for this line. If set to True, the phone sends periodic SIP REGISTER to the service provider according to the settings in the ITSP Profile. Otherwise, the phone doesn't send any SIP REGISTER for the service.	True
<b>X_AcceptSipFromRegistrar Only</b>	Accept SIP packets coming from the current registrar IP address only.	False
<b>X_NoRegNoCall</b>	Enables this option to disallow incoming and outgoing calls if registration with the service provider isn't successful.	False
<b>X_KeepAliveEnable</b>	Enables sending keep alive message. If set to True, the phone sends periodic keep-alive messages to the same server where a REGISTER request would be sent. The content of this message is the ASCII string <code>keep-alive\r\n</code> .	False
<b>X_KeepAliveExpires</b>	Keep alive period in seconds.	15

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>X_KeepAliveExpiresFollow Server</b>	Follows the keep-alive expires value from server in the VIA-header keep parameter.	True
<b>X_KeepAliveMsgType</b>	The type of keep alive messages to send out periodically if keep-alive is enabled. Choice of: <ul style="list-style-type: none"> <li>• <code>keep-alive</code>: The string <code>keep-alive</code></li> <li>• <code>empty</code>: A blank line</li> <li>• <code>stun</code>: A standard STUN binding request. The phone uses the binding response to form its contact address for REGISTRATION</li> <li>• <code>custom</code>: use the value of <b>X_CustomKeepAliveMsg</b></li> <li>• <code>options</code>: A SIP OPTIONS message</li> <li>• <code>notify</code>: A SIP NOTIFY message</li> </ul>	<code>keep-alive</code>
<b>X_CustomKeepAliveMsg</b>	Defines the custom message to be used when <b>X_KeepAliveMsgType</b> is <code>custom</code> . The value has the following format: <code>mtd=NOTIFY;event=&lt;whatever&gt;;user=&lt;anyone&gt;</code> Where <ul style="list-style-type: none"> <li>• <code>NOTIFY</code> can be replaced by any other SIP method, such as <code>PING</code></li> <li>• <code>event</code> is optional and is only applicable if method is <code>NOTIFY</code>. If <code>event</code> isn't specified, the 'keep-alive' event is used with <code>NOTIFY</code></li> <li>• <code>user</code> is optional. If not specified, the request-uri won't have a user ID, and the TO header field uses the same user ID as the FROM header, which is the local account user ID. If <code>user</code> is specified, it's used as the user ID in the Request-URI and TO header.</li> </ul> SIP messages for keep-alive are sent only once without retransmission. Responses to the SIP messages are ignored by the phone.	None
<b>X_UserAgentPort</b>	UDP port where the phone sends and listens for SIP messages.	5060
<b>X_UserAgentPorts</b>	A comma-separated list of as many as 10 alternative user agent ports to use when there is no response received from the SIP Registrar.	None
<b>DirectoryNumber</b>	Directory number associated with this service.	None
<b>X_DefaultRing</b>	Default ring pattern number to ring the phone for incoming calls on this trunk that are routed to the phone according to the <b>InboundCallRoute</b> of this service. The ring pattern is taken from the selected Ring Profile. Valid choices are 1–10.	1
<b>X_RingtoneFile</b>	Default ring tone to use for incoming calls on this trunk.	None



**SPn Services Settings (continued)**

Parameter	Description	Default Setting
<b>X_SatelliteMode</b>	Enables satellite mode on this trunk. In this mode, you must explicitly sign on (using * code) to receive phone calls on this trunk. The SIP REGISTER sent by the phone to the ITSP on this trunk indicates if you want to sign on (and therefore takes over the incoming calls for this account). This feature is only applicable if the service is provided by an OBiPLUS system.	False
<b>X_AcceptResync</b>	Controls whether to accept a SIP NOTIFY request with event=resync to trigger a reboot of the phone so it can download new firmware or configuration upon boot up. Choice of: <ul style="list-style-type: none"> <li>no (don't accept resync trigger)</li> <li>yes with authentication (accept after challenging the sender)</li> <li>yes without authentication (accept without challenging the sender)</li> </ul>	yes without authentication
<b>Debug Options (VoiceService.1.VoiceProfile.1.Line.n.), n = 1 – 6</b>		
<b>X_SipDebugOption</b>	Enables sending of SIP signaling debug information to the syslog server (if one is configured on the device). Choice of: <ul style="list-style-type: none"> <li>Disable (do not send SIP signaling debug information)</li> <li>Log All Messages (log all messages)</li> <li>Log All Except REGISTER Messages</li> </ul>	Disable
<b>X_SipDebugExclusion</b>	Comma-separated list of SIP methods to exclude from the debug log.	None
<b>Credentials (VoiceService.1.VoiceProfile.1.Line.n.SIP), n = 1 – 6</b>		
<b>AuthUserName</b>	The User ID to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the phone is challenged by the UAS with a 401 or 407 response.	None
<b>AuthPassword</b>	The password (corresponding to <b>AuthUserName</b> ) to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the phone is challenged by the UAS with a 401 or 407 response.	None
<b>DTLSPeerCertAltName</b>	Alternate name for verifying peer's DTLS certificate.	None

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>URI</b>	<p>This parameter affects the way the AOR is formed by the phone in outbound SIP Requests. The AOR has the format: <code>user@domain</code>.</p> <p>If the value of <b>URI</b> is empty, the phone gets the user portion of its AOR from the <b>AuthUserName</b>, and the domain portion the value of ITSP Profile's <b>UserAgentDomain</b> if it isn't empty, or that of the <b>ProxyServer</b> otherwise.</p> <p>If the value of <b>URI</b> isn't empty and does not contain "@", it is used as the user portion of the AOR while the domain portion is formed the usual way.</p> <p>If the value of <b>URI</b> contains @, it is interpreted as a full AOR and phone takes it as the AOR as is.</p> <p>Some examples:</p> <ul style="list-style-type: none"> <li>Let <b>ProxyServer</b> = <code>sip.myitisp.com</code>, <b>AuthUserName</b> = <code>4089991123</code>, <b>URI</b>=[empty], <b>UserAgentDomain</b>=[empty], then AOR = <code>4089991123@sip.myitisp.com</code></li> <li>Change <b>UserAgentDomain</b> to <code>users.myitisp.com</code>, then AOR = <code>4089991123@users.myitisp.com</code></li> <li>Change <b>URI</b> to <code>bobdylan</code>, then AOR = <code>bobdylan@users.myitisp.com</code></li> <li>Change <b>URI</b> to <code>bobdylan@superusers.myitisp.com</code>, then AOR = <code>bobdylan@superusers.myitisp.com</code></li> </ul> <p>Note: In all cases, the phone uses <b>AuthUserName</b> and <b>AuthUserPassword</b> to compute authorization if challenged by a 401 or 407 response.</p>	None
<b>X_MyExtension</b>	An extension assigned to this line for inter-office calling and feature invocation.	None
<b>X_XsiUserName</b>	User name to authenticate a BroadSoft XSI Application Server. If not specified, <code>sip-userid@sip-proxy</code> is used.	None
<b>X_XsiPassword</b>	Password to authenticate a BroadSoft XSI Application Server. If not specified, the SIP password is used.	None
<b>X_XmppDomain</b>	Domain name of a BroadSoft XMPP Service.	None
<b>X_XmppDomainUseSRV</b>	Enables using SRV record to resolve XMPP domain.	False
<b>X_XmppUserName</b>	User name to authenticate to the XMPP Server.	None
<b>X_XmppAuthWithUserName</b>	Use the <b>XmppUserName</b> as-is for authentication without appending the @domain.	False
<b>X_XmppPassword</b>	Password to authenticate a BroadSoft XMPP Application Server.	None

**SPn Services Settings (continued)**

Parameter	Description	Default Setting
<b>X_ContactUserID</b>	An alternative user ID to be used in Contact header. Enter <code>Random</code> to let the phone generate a random one.	None
<b>X_EnforceRequestUserID</b>	Enforce incoming INVITE request user ID to match <b>AuthUserName</b> or <b>X_ContactUserID</b> .	False
<b>Share Line Features (VoiceService.1.VoiceProfile.1.Line.n.ShareLineFeatures.), n = 1 – 6</b>		
<b>X_ShareLine</b>	Check this option if this is a shared line.	False
<b>X_ShareLineUserID</b>	The (third-party) user ID to register for this line, if different from the account user ID.	None
<b>X_ShareLineBargeIn</b>	Check this if the share line supports barge-in function.	False
<b>X_ShareLineWhisper</b>	Check this if the share line supports whisper function.	False
<b>X_ShareLineMonitor</b>	Check this if the share line supports monitor function.	False
<b>X_ShareLinePrivateHold</b>	Check this if the share line supports private hold function.	False
<b>X_ShareLineCallPickup</b>	Check this if the share line supports call pickup function.	False
<b>BLF Features (VoiceService.1.VoiceProfile.1.Line.n.BLFFeatures.), n = 1 – 6</b>		
<b>X_BifBargeIn</b>	Check this if the BLF feature supports barge-in function.	False
<b>X_BifCallPickup</b>	Check this if the BLF feature supports call pickup function.	False
<b>X_BifDirectedCallPickup</b>	Check this if the BLF feature supports directed call pickup function.	False
<b>X_BifWhisper</b>	Check this if the BLF feature supports whisper (or coach) function.	False
<b>X_BifMonitor</b>	Check this if the BLF feature supports monitor function.	False
<b>Calling Features (VoiceService.1.VoiceProfile.1.Line.n.CallingFeatures.), n = 1 – 6</b>		
<b>CallerIDName</b>	Display name to identify the subscriber. The display name field is usually inserted in a FROM header in outbound SIP requests (such as INVITE) for the purpose of displaying a Caller ID Name on the recipient's device.	None
<b>MaxSessions</b>	The maximum number of simultaneous calls that can be established on this service.	4
<b>CallForwardUnconditional Enable</b>	Enables call forwarding of all calls unconditionally by the phone. If <b>CallForwardUnconditionalNumber</b> is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the phone using a Star Code.	False

SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>CallForwardUnconditionalNumber</b>	Directory number to forward all incoming calls on this service unconditionally. Maximum Length is 127 characters. Note: You can set this parameter from the phone using a Star Code.	None
<b>CallForwardOnBusyEnable</b>	Enables call forwarding of all incoming calls when the phone is busy. If <b>CallForwardOnBusyNumber</b> is blank, this parameter is treated as if it has been set to False. The phone is considered busy if one of the following conditions holds: <ul style="list-style-type: none"> <li>This service already reaches the limit of simultaneous calls as specified in <b>MaxSessions</b></li> <li>DND (Do Not Disturb) Service is enabled on this service if the call is routed to the phone where the phone is in a busy state (such as ringing, dialing, playing reorder, or already having 2 calls in progress)</li> </ul> Note: You can set this parameter from the phone using a Star Code.	False
<b>CallForwardOnBusyNumber</b>	Directory number to forward all incoming calls on this service when the phone is busy. Maximum Length is 127 characters. Note: You can set this parameter from the phone using a Star Code.	None
<b>CallForwardOnNoAnswerEnable</b>	Enables call forwarding of all incoming calls when the call isn't answered after a period as specified in <b>CallForwardOnNoAnswerRingCount</b> . If <b>CallForwardOnNoAnswerNumber</b> is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the phone using a Star Code.	False
<b>CallForwardOnNoAnswerNumber</b>	Directory number to forward all incoming calls when the call isn't answered after a period specified in <b>CallForwardNoAnswerRingCount</b> . Note: You can set this parameter from the phone using a Star Code.	None
<b>CallForwardOnNoAnswerRingCount</b>	Number of rings to be considered by the phone as no answer to an incoming call. Note: 1 ring is approximately 6 seconds.	2
<b>X_BlockedCallers</b>	A comma-separated list of as many as 10 caller numbers to block from calling this service.	None
<b>X_MailboxID</b>	The mailbox ID to subscribe MWI with.	None
<b>X_CheckVoiceMailNumber</b>	The number to call to check voicemail.	None
<b>X_CheckVoiceMailWithCallKey</b>	Enables checking voicemail by pressing and holding down a call key that is bound to this SP service, provided <b>X_CheckVoiceMailNumber</b> is specified.	False

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>MWIEnable</b>	Enables MWI Signal (stutter dial tone) generation. If enabled, any SP voice service enabled on the phone that has MWI Service enabled triggers the generation of stutter dial tone if there are new voicemails for the subscriber on the service provider's voicemail system.	True
<b>X_VMWIEnable</b>	Enables Visual Message Waiting Indication for this service on the phone. It is specified as a bit mask, such that bit 0 = Phone 1, bit 1 = phone 2, and so forth.	255
<b>X_MessageCenterEnable</b>	Shows this Message Status under <b>Messages</b> menu of phone UI.	True
<b>X_CallFeaturePreferences Menu</b>	Shows Call Feature Preferences under <b>Preferences</b> menu of phone UI.	False
<b>MessageWaiting</b>	This state indicates if there are any new messages for this subscriber on the service provider's voicemail system.	False
<b>MessageCount</b>	Displays count of new messages, in format new/old (urgent – new/urgent – old).	None
<b>AnonymousCallBlockEnable</b>	Enables blocking of Anonymous Calls on this service. Anonymous calls are rejected with a SIP 486 (Busy) response and Call Forward On Busy service isn't applied. Note: You can set this parameter from the phone using a Star Code.	False
<b>AnonymousCallEnable</b>	Enables masking of Caller-ID information for all outgoing calls. If enabled, the called party sees the call as coming from an anonymous caller. Note: You can set this parameter from the phone using a Star Code.	False
<b>DoNotDisturbEnable</b>	Enables Do Not Disturb Service. If enabled, all incoming calls on this service are treated as if the phone is busy. Note: You can set this parameter from the phone using a Star Code.	False
<b>X_BridgedOutboundCallMaxDuration</b>	Limit on the call duration in seconds for all outbound calls that are bridged from the same or another trunk. A blank or 0 value implies the call duration isn't limited.	None
<b>X_AcceptDialogSubscription</b>	Enables the phone to accept SUBSCRIBE to this trunk's dialog event package.	False
<b>X_SkipCallScreening</b>	Automatically skip call screening on answering incoming call by dialing out some digit during the first 2 seconds.	True

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>X_S RTP</b>	<p>Enables SRTP. Choose one of:</p> <ul style="list-style-type: none"> <li><code>Disable SRTP</code> = Do not use SRTP for all calls. The call fails if the peer insists on using SRTP only.</li> <li><code>Use SRTP Only</code> = Require all calls to use SRTP. The call fails if the peer does not support SRTP.</li> <li><code>Use SRTP When Possible</code> = Use SRTP for a call if the peer supports SRTP. Otherwise, fall back to use regular unencrypted SRTP.</li> </ul>	<code>Disable SRTP</code>
<b>X_ASFeatureEventSubscribe</b>	Enables subscription to the as-feature-event package.	False
<b>X_ShowDiversionHistory</b>	Enables showing the diversion history of a call on the Calls screen.	True
<b>X_RTPRecordURL</b>	URL to upload RTP recording.	None
<b>X_UrlDialingIPAddress</b>	If enabled, dialing or calling an IP address bypasses the outbound proxy.	True
<b>X_UrlDialingDomainName</b>	If enabled, dialing or calling a domain name bypasses the outbound proxy. For example, if you have a conference bridge number SP1(bridge@some-domain.com), you may need to set this option to False if you still need the call to this bridge to go through the outbound proxy.	True
<b>X_IncomingCallMustUseBoundCallKey</b>	<p>Controls whether incoming call must be assigned to a bound call key only or can be assigned to a floating (unbound) call key if bound call keys aren't available.</p> <p>--OR--</p> <p>If enabled, the phone must assign an incoming to a Line Key with Function = Call Appearance and Service the same as the one the call is from. Otherwise, the phone may also assign the call to a Line Key with Function = Call Appearance and Service = {blank} as a fall-back.</p>	False
<b>X_ConferenceBridge</b>	<p>The number of an external conference bridge to use with calls on this service.</p> <p>Note: If the number also specifies the service to use, such as SP1(bridge@xyz-domain.com), the phone calls the number as it is on the given service. Otherwise, the phone applies its digit map and outbound call route setting to determine which service to use for the call.</p>	<code>cbridge</code>

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>X_CallEndedActionURL</b>	<p>Specifies the <b>ActionURL</b> to invoke when a call ends. The value is a valid HTTP/HTTPS URL that returns an HTTP 2xx response with a valid Obihai Phone XML App in the body.</p> <p>Note: ActionURL automatically includes the called SIP CALL-ID and GMT-timedate as URL-query parameters. You can use the following macros:</p> <ul style="list-style-type: none"> <li>• <code>\$peername</code> = peer caller ID name</li> <li>• <code>\$peernumber</code> = peer caller ID number</li> <li>• <code>\$dur</code> = duration of call</li> <li>• <code>\$refid</code> = the Reference ID, used to control the call in an Phone XML App</li> <li>• <code>\$dir</code> = direction of the call (inbound or outbound)</li> <li>• <code>\$service</code> = the service or line the call is on (e.g., SP1 or SP6)</li> </ul> <p>For example:  <code>http://abc.com?name=\$peerName&amp;number=\$peernumber&amp;ref=\$refid</code></p>	None
<b>X_CallConnectedActionURL</b>	<p>Specifies the <b>ActionURL</b> to invoke when a call is answered. The value is a valid HTTP/HTTPS URL that returns an HTTP 2xx response with a valid Obihai Phone XML App in the body.</p> <p>Note: ActionURL automatically includes the called SIP CALL-ID and GMT-timedate as URL-query parameters. You can use the following macros:</p> <ul style="list-style-type: none"> <li>• <code>\$peername</code> = peer caller ID name</li> <li>• <code>\$peernumber</code> = peer caller ID number</li> <li>• <code>\$dur</code> = duration of call</li> <li>• <code>\$refid</code> = the Reference ID, used to control the call in an Phone XML App</li> <li>• <code>\$dir</code> = direction of the call (inbound or outbound)</li> <li>• <code>\$service</code> = the service or line the call is on (e.g., SP1 or SP6)</li> </ul> <p>For example:  <code>http://abc.com?name=\$peerName&amp;number=\$peernumber&amp;ref=\$refid</code></p>	None

SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>X_IncomingCallRingingActionURL</b>	<p>Specifies the <b>ActionURL</b> to invoke when an incoming call starts ringing.</p> <p>The value is a valid HTTP/HTTPS URL that returns an HTTP 2xx response with a valid Obihai Phone XML App in the body.</p> <p>Note: ActionURL automatically includes the called SIP CALL-ID and GMT-timedate as URL-query parameters. You can use the following macros:</p> <ul style="list-style-type: none"> <li>• \$peername = peer caller ID name</li> <li>• \$peernumber = peer caller ID number</li> <li>• \$dur = duration of call</li> <li>• \$refid = the Reference ID, used to control the call in an Phone XML App</li> <li>• \$dir = direction of the call (inbound or outbound)</li> <li>• \$service = the service or line the call is on (e.g., SP1 or SP6)</li> </ul> <p>For example:  <code>http://abc.com?name=\$peerName&amp;number=\$peernumber&amp;ref=\$refid</code></p>	None
<b>X_TranscriptionService</b>	Enables the line to stream PCM audio to a transcription service.	False
<b>X_IncomingCallerIDReplaceMap</b>	A digitmap of incoming caller id number patterns that may be replaced in call history when modified during the call. Default is empty, which matches nothing. Use (xx.) to match any number.	None
<b>Network Provided Services (VoiceService.1.VoiceProfile.1.Line.n.X_NetServices.), n = 1 – 6</b>		
<b>ACDAgent</b>	ACD Agent sign or off. Requires <b>BroadSoft/Feature-Sync</b> .	False
<b>ACDAgentUnavailableCodes</b>	ACD Agent Unavailable Codes. Requires <b>BroadSoft/Feature-Sync</b> .	False
<b>AnonymousCall</b>	Allows Anonymous Call service, also known as Block Caller ID. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False
<b>AnonymousCallRejection</b>	Allows Anonymous Call Rejection service. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False
<b>BroadWorksAnywhere</b>	Allows Broadworks Anywhere service. Requires <b>BroadSoft/XSI</b> .	False
<b>BuddyList</b>	Buddy List Service. Requires BroadSoft or Generic XMPP Service.	False
<b>CallCenter</b>	Call Center. Requires BroadSoft.	False
<b>CallForwardAlways</b>	Call Forward Always. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False



## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>CallForwardBusy</b>	Call Forward on Busy. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False
<b>CallForwardNoAnswer</b>	Call Forward on No Answer. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False
<b>CallLogs</b>	Call Logs Service. Requires BroadSoft.	False
<b>CallPark</b>	Park a call (softkey) against current extension with a Feature Access Code.	False
<b>CallParkStatus</b>	Get notified if a call is parked against current extension. Requires BroadSoft.	False
<b>CallPickup</b>	Retrieve a call (softkey) parked against current extension with a Feature Access Code.	False
<b>CallRecording</b>	Call Recording Control. Requires BroadSoft.	False
<b>CallTrace</b>	Customer Originated Call Trace. Requires BroadSoft.	False
<b>CallWaiting</b>	Call Waiting. Requires BroadSoft.	
<b>Directory</b>	Directory Service. Requires BroadSoft or similar directory.	False
<b>DispositionCode</b>	Disposition Code. Requires BroadSoft.	False
<b>DoNotDisturb</b>	Enable subscription of Do Not Disturb (XSI) Event. Requires <b>BroadSoft/Feature-Sync</b> or <b>BroadSoft/XSI</b> .	False
<b>Escalation</b>	Escalation (to a supervisor). Requires BroadSoft.	False
<b>Executive</b>	Executive Call Filtering On/Off. Requires BroadSoft.	False
<b>ExecutiveAssistant</b>	Executive Assistant. Requires BroadSoft.	False
<b>Hoteling</b>	Hoteling or Guest User Sign On or Off. Requires BroadSoft.	False
<b>RemoteOffice</b>	Remote Office. Requires <b>BroadSoft/XSI</b> .	False
<b>SecurityClass</b>	Security Classification. Requires <b>BroadSoft/Feature-Sync</b> .	False
<b>SimultaneousRing</b>	Simultaneous Ring. Requires <b>BroadSoft/XSI</b> .	False
<b>UserProfile</b>	User Profile. Requires <b>BroadSoft/XSI</b> .	False
<b>VisualVoiceMail</b>	Visual Voice Mail. Requires <b>BroadSoft/XSI</b> .	False
<b>ConferenceBridgeControl</b>	View who is on the bridge and apply some control on selected participants. Requires <b>BroadSoft/XSI</b> .	False
<b>Network Directory Setup (VoiceService.1.VoiceProfile.1.Line.n.X_NetServices.Dir.), n = 1 – 6</b>		
<b>URL</b>	URL to download a Group directory. This parameter must be {blank} when using BroadWorks Directories where the URL to download the directory is formed internally based on BroadSoft XSI API.	None

**SPn Services Settings (continued)**

Parameter	Description	Default Setting
<b>EnableSearch</b>	Enables the Search softkey option on the directory pages.	True
<b>EnableResultFilter</b>	Enables the option to let a user enter a value to filter results on the current page.	False
<b>ResultsPerPage</b>	Number of results per page to suggest to the server. Default is 0, which means unspecified. Note that this is just a suggestion to the server, which may return a different number of results per page. The phone always tries to display all the results returned by the server.	0
<b>EnableGroup</b>	Enables the Group subdirectory.	True
<b>EnableGroupCommon</b>	Enables the GroupCommon subdirectory.	True
<b>EnableEnterprise</b>	Enables the Enterprise subdirectory.	True
<b>EnableEnterpriseCommon</b>	Enables the EnterpriseCommon subdirectory.	True
<b>EnablePersonal</b>	Enables the Personal subdirectory.	True
<b><i>Buddy List Setup (VoiceService.1.VoiceProfile.1.Line.n.X_NetServices.BuddyList.), n = 1 – 6</i></b>		
<b>ServerType</b>	Value can be <code>BroadSoft</code> or <code>Generic</code>	<code>BroadSoft</code>
<b>Priority</b>	Priority of the contact. Must be in the range of –127 to 128, inclusive.	33
<b>VerifyServerCert</b>	Boolean. Turns the verification of server certificate in a SSL connection with the XMPP server on or off.	False
<b>QueryVcard</b>	Boolean. Turns the reading of VCARD on or off for each item in the roster list. Turn on this option if you wish to get a picture of each buddy.	False
<b>MyPresence</b>	The (XMPP) Presence value of this line. Choice of: <ul style="list-style-type: none"> <li>• <code>online</code></li> <li>• <code>away</code></li> <li>• <code>xa</code></li> <li>• <code>dnd</code></li> <li>• <code>offline</code></li> </ul>	<code>online</code>
<b>MyStatus</b>	The (XMPP) Status value of this line. Can be any human readable string.	None
<b><i>Action URL Setup (VoiceService.1.VoiceProfile.1.Line.n.X_NetServices.ActionURL.), n = 1 – 6</i></b>		
<b>SyncURL</b>	The <b>ActionURL</b> the phone should execute every time when it starts up.	None
<b>OBiStatusNotifyURL</b>	The URL to send <obi-status> XML on start up and for status update. Default is SIP/NOTIFY. To use HTTP/POST, specify <code>http://</code> or <code>https://</code> at the beginning of the URL.	None

## SPn Services Settings (continued)

Parameter	Description	Default Setting
<b>OBiStatusFormat</b>	Choose the format to report obi-status with. Value can be either <code>obi-status</code> or <code>dialog-info</code> . The former one is an OBiPhone proprietary XML, the latter a <code>&lt;dialog-info&gt;</code> XML based on RFC4235.	<code>obi-status</code>
<b>OBiStatusFilter</b>	<p>Specifies which status changes to report to the given URL. The value is a string that is a comma-separated list of filter groups. There are three filter groups:</p> <ul style="list-style-type: none"> <li><b>Calls:</b> For status related to current calls on the phone, such as Call State, Caller-ID, and so forth.</li> <li><b>Services:</b> For status related to a voice service, such as up or down</li> <li><b>Features:</b> For status related to the setting of a feature, such as DND on or off</li> </ul> <p>Each filter group takes a comma-separated list of arguments to specify which status within each group to report. For the Calls group, the arguments can be one or more of the following: <code>sp1, sp2, sp3, sp4, sp5, sp6, pp</code>. Each argument indicates the service whose call status are to be reported. For the Services group, the arguments can be one or more of the following: <code>sp1, sp2, sp3, sp4, sp5, sp6, pp</code>. Each argument indicates the service whose statuses (except call status) are reported. For the Features group, each argument specifies a feature whose setting changes be reported. The following is a list of valid feature arguments:</p> <ul style="list-style-type: none"> <li><code>aans</code> (Auto Answer Intercom setting)</li> <li><code>aud</code> (Audio Path setting)</li> <li><code>bac</code> (Block Anonymous Call setting)</li> <li><code>bci</code> (Block Caller ID setting)</li> <li><code>cwa</code> (Call Waiting setting)</li> <li><code>dnd</code> (Do Not Disturb setting)</li> <li><code>dnr</code> (Do Not Ring setting)</li> <li><code>miss</code> (Number of new missed calls)</li> <li><code>mwi</code> (New Messages Waiting status)</li> <li><code>ncalls</code> (Number of calls)</li> <li><code>pg1</code> through <code>pg10</code> (setting for Page Group 1 through 10)</li> <li><code>ring</code> (Ring status)</li> </ul> <p>Note: If <b>OBiStatusFormat</b> = <code>dialog-info</code>, only the Calls filter group is used. The other filter groups are ignored.</p> <p>For example:  <code>Calls(sp1), Services(sp1, sp2), Features(dnd)</code></p>	<code>Calls(sp1, sp2, sp3, sp4, sp5, sp6)</code>

## OBiTALK Service Web Page

The following configuration parameters are available on this page.

## OBiTALK Service Settings

Parameter	Description	Default Setting
<b>OBiTALK Service Settings (VoiceService.1.X_P2P.1.)</b>		
<b>Enable</b>	Enables the OBiTALK Service (the built-in free voice service that comes with every OBi Edition phone).	True
<b>DisplayLabel</b>	A label that represents this service on the phone screen. Special characters must include a closing character. For example, an open curly bracket must also include a closing curly bracket.	None
<b>DisplayNumber</b>	A number that represents this service on the phone screen. Special characters must include a closing character. For example, an open curly bracket must also include a closing curly bracket.	None
<b>LocalPort</b>	The UDP or TCP port used by the phone to send and listen for OBiTALK messages.	10000
<b>TryMultiplePorts</b>	Enables the phone to try a few random UDP ports until it can successfully join the OBiTALK network.	True
<b>Transport</b>	The transport protocol to use for OBiTALK traffic. Choice of: <ul style="list-style-type: none"> <li>• UDP</li> <li>• UDP/TCP</li> <li>• UDP/TLS</li> <li>• TLS</li> </ul>	UDP/TCP
<b>ServerAddress</b>	OBiTALK Server IP Address Should be empty in normal operation.	None
<b>LastRegistrarAddress</b>	IP address and port number of the last registrar address used.	None
<b>TLSserverPort</b>	TLS server port.	443
<b>DisplayName</b>	Display name to identify the subscriber, for the purpose of displaying a Caller ID Name on the recipient's phone.	None
<b>DigitMap</b>	Digit map to restrict numbers that can be dialed or called with this service. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify a digit map.	(<ob>xxxxxxxxxx   obxxxxxxxxxx)
<b>InboundCallRoute</b>	Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the phone. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify this parameter.	ph
<b>RingProfile</b>	Select a Ring Profile to ring the phone with when an incoming call is routed to the phone. Choices are A or B.	A

## OBiTALK Service Settings (continued)

Parameter	Description	Default Setting
<b>CodecProfile</b>	Select a Codec Profile to be used for all calls on this service. Choices are A or B.	A
<b>DefaultRing</b>	Default ring pattern number to ring the phone for incoming calls on this trunk that are routed to the phone according to the <b>InboundCallRoute</b> of this service. The ring pattern is taken from the selected Ring Profile. Valid choices are 1–10.	2
<b>RingtoneFile</b>	Default ring tone for OBiTALK Service.	None
<b>CallOnHoldRing</b>	Pattern to ring phone when holding a call on this trunk that has been connected to the phone. Typically this is a very short distinctive ring pattern that serves as a reminder to you that a call is being on hold. The ring pattern is taken from the selected Ring Profile. Choice of: <ul style="list-style-type: none"> <li>• NO Ring</li> <li>• 1-10.</li> </ul>	8
<b>RepeatDialRing</b>	The ring pattern number to use to ring the phone when a repeat dial operation on this trunk is successful as the called party is either ringing or answered.	4
<b>DTMFMethod</b>	Method to pass DTMF digits to peer phone. Choice of: <ul style="list-style-type: none"> <li>• Inband: DTMF tone are sent as inband audio signal</li> <li>• RFC283 : DTMF tone events are relayed per RFC2833</li> <li>• SIPInfo: DTMF tones are relayed with SIP INFO request</li> <li>• Auto: Method to use based on call setup negotiation (either Inband or RFC2833 can be negotiated)</li> </ul>	Auto
<b>InbandDTMFVolume</b>	DTMF tone volume to use when sending inband DTMF. Choice of: <ul style="list-style-type: none"> <li>• -24dB</li> <li>• -21dB</li> <li>• -18dB</li> <li>• -15dB</li> <li>• -12dB</li> <li>• -9dB</li> <li>• -6dB</li> <li>• -3dB</li> <li>• 0dB</li> </ul>	-15dB
<b>UseFixedDurationRFC2833DTMF</b>	When relaying DTMF digit events on this trunk using RFC2833, the RFC2833 RTP packets normally keep streaming for as long as the digit is pressed. With this option set to True, the phone sends only one RTP digit event packet with a fixed duration of 150 ms regardless how long the digit has been pressed.	False

## OBiTALK Service Settings (continued)

Parameter	Description	Default Setting
<b>SymmetricMedia</b>	If incoming payload type changes unannounced, after 10 packets with the new payload type, decoding switches to the new format. If symmetric media is enabled, outgoing packets are also in the new format.	True
<b>Calling Features (VoiceService.1.X_P2P.1.CallingFeatures.)</b>		
<b>CallForwardUnconditionalEnable</b>	Enables call forwarding of all calls unconditionally by the phone. If <b>CallForwardUnconditionalNumber</b> is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the phone using a Star Code.	False
<b>CallForwardUnconditionalNumber</b>	Directory number to forward all incoming calls on this service unconditionally. Maximum length is 127 characters. Note: You can set this parameter from the phone using a Star Code.	None
<b>CallForwardOnBusyEnable</b>	Enables call forwarding of all incoming calls when the phone is busy. If <b>CallForwardOnBusyNumber</b> is blank, this parameter is treated as if it has been set to False. The phone is considered busy if one of the following conditions holds: <ul style="list-style-type: none"> <li>This service already reaches the limit of simultaneous calls as specified in <b>MaxSessions</b>.</li> <li>DND (Do Not Disturb) Service is enabled on this service</li> <li>If the call is routed to the phone where the phone is in a busy state (such as ringing, dialing, playing reorder, or already having 2 calls in progress)</li> </ul> Note: You can set this parameter from the phone using a Star Code.	False
<b>CallForwardOnBusyNumber</b>	Directory number to forward all incoming calls on this service when the phone is busy. Maximum Length is 127 characters. Note: You can set this parameter from the phone using a Star Code.	None
<b>CallForwardOnNoAnswerEnable</b>	Enable call forwarding of all incoming calls when the call isn't answered after a period as specified in <b>CallForwardOnNoAnswerRingCount</b> . If <b>CallForwardOnNoAnswerNumber</b> is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the phone using a Star Code.	False
<b>CallForwardOnNoAnswerNumber</b>	Directory number to forward all incoming calls when the call isn't answered after a period specified in <b>CallForwardNoAnswerRingCount</b> Note: You can set this parameter from the phone using a Star Code.	None

**OBiTALK Service Settings (continued)**

Parameter	Description	Default Setting
<b>CallForwardOnNoAnswerRingCount</b>	Number of rings to be considered by the phone as no answer to an incoming call. Note: 1 ring is approximately 6 seconds.	2
<b>BlockedCallers</b>	A comma-separated list of as many as 10 caller numbers to block from calling this service.	None
<b>MaxSessions</b>	The maximum number of simultaneous calls that can be established on this service.	2
<b>AnonymousCallBlockEnable</b>	Enables blocking Anonymous Calls on this service. Anonymous calls are rejected with a SIP 486 (Busy) response and Call Forward On Busy service isn't applied. Note: You can set this parameter from the phone using a Star Code.	False
<b>AnonymousCallEnable</b>	Enables Anonymous Call.	False
<b>DoNotDisturbEnable</b>	Enables Do Not Disturb.	False
<b>CallStatusAggregator</b>	An OBi number that serves as call status aggregator for OBi phones.	None
<b>RTPRecordURL</b>	URL to upload RTP recording.	None
<b>Accept SipCall</b>	Accept SIP call. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	yes without authentication
<b>SipCallAuthPassword</b>	Password to authenticate inbound SIP calls.	None
<b>TranscriptionService</b>	Enables the line to stream PCM audio to a transcription service.	False

***Inbound Direct Dialing Authentication (VoiceService.1.X\_P2P.1.VoiceGateway.)***

**OBiTALK Service Settings (continued)**

Parameter	Description	Default Setting
<b>AuthMethod</b>	<p>The OBiTALK protocol allows incoming calls to indicate a target number that is different from this phone's OBi number. In this case, the phone attempts to establish and bridge the call to the target number according to the rules configured in the trunk's <b>InboundCallRoute</b> parameter. Hence, the phone acts as a gateway and the method is referred to direct dialing or 1-stage dialing (versus 2-stage dialing via the Auto Attendant). Since the caller isn't able to enter a PIN in such cases, an automated method based on signaling protocol must be used to authenticate the caller if authentication is required. The phone offers the following choices for this purpose:</p> <ul style="list-style-type: none"> <li>• <b>None</b> = Disable authentication</li> <li>• <b>HTTP Digest</b> = Use <b>HTTP Digest</b> with user ID and password pairs. Note that at least one of <b>AuthPasswordx</b> (x=1,2,3,4) must be specified, otherwise authentication is disabled.</li> </ul>	HTTP Digest
<b>AuthUserID1</b>	One of 4 user IDs for authenticating direct dialing callers.	None
<b>AuthPassword1</b>	One of 4 passwords for authenticating direct dialing callers.	None
<b>AuthUserID2</b>	One of 4 user IDs for authenticating direct dialing callers.	None
<b>AuthPassword2</b>	One of 4 passwords for authenticating direct dialing callers.	None
<b>AuthUserID3</b>	One of 4 user IDs for authenticating direct dialing callers.	None
<b>AuthPassword3</b>	One of 4 passwords for authenticating direct dialing callers.	None
<b>AuthUserID4</b>	One of 4 user IDs for authenticating direct dialing callers.	None
<b>AuthPassword4</b>	One of 4 passwords for authenticating direct dialing callers.	None
<b>Jitter Buffer (VoiceService.1.X_P2P.1.JIB.)</b>		
<b>Adaptive</b>	Enables jitter buffer adaptation.	True
<b>MaximumSize</b>	Maximum jitter buffer size in ms.	250
<b>SetPoint</b>	Initial play out delay in ms.	60
<b>Target</b>	Target play out delay in ms.	20
<b>AdaptationSlope</b>	Maximum adaptation slope in samples per 10 ms.	16

## Auto Attendant Web Page

The following configuration parameters are available on this web page.



**Auto Attendant Settings**

Parameter	Description	Default Setting
<i>Auto Attendant (VoiceService.1.X_AA.1.)</i>		
<b>Enable</b>	Enables AA. When enabled, the AA answers an incoming call that has been routed to it after the period specified in <b>AnswerDelay</b> . If disabled, the AA won't attempt to answer any incoming call.	True
<b>DigitMap</b>	Once the AA answers an incoming call, it presents the caller with an option to make a further call using one of the available voice services on the device. This Digit map serves to restrict the numbers that can be dialed or called via this AA option. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify a digit map.	<code>([1-9]x?* (Mpli)   [1-9]   [1-9] [0-9]   &lt;0:\$1&gt;   [0-8]   **1 (Msp1)   **2 (Msp2)   **3 (Msp3)   **4 (Msp4)   **9 (Mpp)   (Mpli) )</code>
<b>OutboundCallRoute</b>	After the caller dials a number that is acceptable by the AA (according to its <b>DigitMap</b> ) to make a further call, the phone uses this outbound call routing rule to determine which service to make this call with. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify this parameter.	<code>{ ([1-9]x?* (Mpli) ):pp}, {0:ph}, { (&lt;**1:&gt; (Msp1) ):sp1}, { (&lt;**2:&gt; (Msp2) ):sp2}, { (&lt;**3:&gt; (Msp3) ):sp3}, { (&lt;**4:&gt; (Msp4) ):sp4}, { (&lt;**9:&gt; (Mpp) ):pp}, { (Mpli) :pli}</code>
<b>PrimaryLine</b>	The primary line is the service that does not require any access code prefix (such as **1 or **9) when dialing. It is the default service to be used for making the call when no explicit access code prefix is entered. This parameter indicates to the phone which voice service is considered as the primary line when dialing out via the Auto Attendant. Choice of: <ul style="list-style-type: none"> <li>• SP1 Service (code = sp1)</li> <li>• SP2 Service (code = sp2)</li> <li>• SP3 Service (code = sp3)</li> <li>• SP4 Service (code = sp4)</li> <li>• SP5 Service (code = sp5)</li> <li>• SP6 Service (code = sp6)</li> <li>• OBiTALK Service (code = pp1)</li> <li>• Trunk Group 1 (code = tg1)</li> <li>• Trunk Group 2 (code = tg2)</li> </ul> The phone processes the parameter by substituting occurrences of pli and (Mpli) in <b>DigitMap</b> and <b>OutboundCallRoute</b> with the corresponding code and (Mcode).	SP1 Service

**Auto Attendant Settings (continued)**

Parameter	Description	Default Setting
<b>AnswerDelay</b>	Time in milliseconds that the AA waits before answering an incoming call that has been routed to it.	4000
<b>CallbackAnswerDelay</b>	Delay in ms the AA answers after ring when the AA should call the caller back if the caller hangs up before it answers.	10000
<b>NumberOnNoInput</b>	If the caller does not enter any option from the top level menu after the menu has been announced 3 times, the AA directs the caller to the number specified in this parameter. If this number isn't specified, the AA terminates the current call.	0 Note: According to the default <b>DigitMap</b> and <b>OutboundCallRoute</b> parameters, calling 0 means ringing the phone.
<b>UsePIN</b>	Enables using a PIN to authenticate callers when they select the option to make a further call. If <b>PIN1</b> , <b>PIN2</b> , <b>PIN3</b> , and <b>PIN4</b> are all empty, the phone treats it as if <b>UsePIN</b> is set to No. Otherwise, the caller must enter one of the non-empty PINs to proceed.	False
<b>PIN1</b>	PIN code to make a call (must be all digits). Maximum length = 15 digits.	None
<b>PIN2</b>	PIN code to make a call (must be all digits). Maximum length = 15 digits.	None
<b>PIN3</b>	PIN code to make a call (must be all digits). Maximum length = 15 digits.	None
<b>PIN4</b>	PIN code to make a call (must be all digits). Maximum length = 15 digits.	None

## Gateways and Trunk Groups Web Page

The following configuration parameters are available on this page.

**Gateways and Trunk Groups Settings**

Parameter	Description	Default Setting
<b>Voice Gateway <i>n</i> (VoiceService.1.X_VoiceGateway.n), <i>n</i> = 1 – 8</b>		
<b>Enable</b>	Enables this voice gateway.	True
<b>Name</b>	An arbitrary user-friendly name to identify this gateway (optional).	None

**Gateways and Trunk Groups Settings (continued)**

Parameter	Description	Default Setting
<b>AccessNumber</b>	The gateway's OBiTALK number, including trunk information, such as: PP (ob200112334) or PP (ob300331456). If the value is blank, the phone treats this voice gateway as disabled. This can also be set to a SIP URL, such as: SP1 (sip.mycompany.com:5060), or SP2 (192.168.15.113).	None
<b>DigitMap</b>	<b>DigitMap</b> for this voice gateway. It can be referenced as (Mvgn).	(xx.)
<b>AuthUserID</b>	A user ID to authenticate with the voice gateway.	None
<b>AuthPassword</b>	A password to authenticate with the voice gateway.	None
<b>Trunk Group <i>n</i> (VoiceService.1.X_TrunkGroup.n.), <i>n</i> = 1 – 4</b>		
<b>Enable</b>	Enables this trunk group.	True
<b>Name</b>	An arbitrary user-friendly name to identify this trunk group (optional).	None
<b>TrunkList</b>	A comma-separated list of names of trunks to include in this trunk group.	For TG1 and TG2, the default is: sp1, sp2, sp3, sp4 For other TGs, the default is (blank).
<b>DigitMap</b>	Digit map associated with this trunk group. It can be referenced as (Msp1) for TG1 and TG2.	For TG1 and TG2, the default is: (1xxxxxxxxxxx <1> [2-9]xxxxxxxx 011xx. xx.) For other TGs, the default is (xx.).

## IP Phone Settings

The IP Phone Settings parameters web pages show parameters for customizing your phone. They include these pages:

- [Phone Settings Web Page](#)
- [Left Line Keys Web Page](#)
- [Right Line Keys Web Page \(VVX 450 only\)](#)
- [Programmable Keys Web Page](#)
- [Side Car X Web Page \(X = 1, 2\)](#)

- [LED Settings Web Page](#)
- [Soft Keys Web Page](#)
- [LDAP Search Web Page](#)
- [Line Key Customization Web Page](#)
- [Screen Item Customization Web Page](#)

## Phone Settings Web Page

The following configuration parameters are available on this page.

### Phone Settings

Parameter	Description	Default Setting
<i>Phone Settings (VoiceService.1.Phone.)</i>		
<b>DigitMap</b>	This digit map restricts the numbers that can be dialed or called from the phone. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify a digit map.	<code>{ [1-9]x?* (Mpli)   [1-9]S9   [1-9] [0-9]S9   ***   **0   **1 (Msp1)   **2 (Msp2)   **3 (Msp3)   **4 (Msp4)   **9 (Mpp)   (Mpli) }</code>
<b>OutboundCallRoute</b>	After the caller dials a number that is acceptable according to the <b>DigitMap</b> , the phone uses this outbound call routing rule to determine which service to make this call with. See the <a href="#">Call Routing</a> and <a href="#">Digit Map Configuration</a> sections for a description of the syntaxes to specify this parameter.	<code>{ ([1-9]x?* (Mpli)) :pp}, { **0:aa }, { ***:aa2 }, { (&lt;**1:&gt; (Msp1)) :sp1 }, { (&lt;**2:&gt; (Msp2)) :sp2 }, { (&lt;**3:&gt; (Msp3)) :sp3 }, { (&lt;**4:&gt; (Msp4)) :sp4 },  { (&lt;**9:&gt; (Mpp)) :pp }, { (Mpli) :pli }</code>
<b>CallReturnDigitMaps</b>	Call Return enables you to call the last caller by dialing a star code (*69 by default). The phone implements this service by remembering the number of the last caller in memory. However, the stored information does not include any dialing prefix to tell the phone which voice service to use to call back the last caller. This list of digit maps serves the purpose of mapping a caller's number to one that includes the desired dialing prefix used exclusively for call return service.	<code>{ pli: (xx.) }, { sp1: (&lt;**1&gt;xx.) }, { sp2: (&lt;**2&gt;xx.) },  { pp: (&lt;**9&gt;xx.) }</code>

Phone Settings (continued)

Parameter	Description	Default Setting
<b>PrimaryLine</b>	<p>The primary line is the service that does not require any access code prefix (such as **1 or **9) when dialing. It is the default service to be used for making the call when no explicit access code prefix is entered. This parameter indicates to the phone which voice service is considered as the primary line when dialing out from the phone. Choice of:</p> <ul style="list-style-type: none"> <li>• SP1 Service (code = sp1)</li> <li>• SP2 Service (code = sp2)</li> <li>• SP3 Service (code = sp3)</li> <li>• SP4 Service (code = sp4)</li> <li>• SP5 Service (code = sp5)</li> <li>• SP6 Service (code = sp6)</li> <li>• OBiTALK Service (code = ppl)</li> <li>• Trunk Group 1 (code = tg1)</li> <li>• Trunk Group 2 (code = tg2)</li> </ul> <p>The phone processes the parameter by substituting of the occurrences of pli and (Mpli) in <b>DigitMap</b>, <b>OutboundCallRoute</b>, and <b>CallReturnDigitMaps</b> with the corresponding code and (Mcode).</p>	SP1 Service
<b>Phone XML App (VoiceService.1.Phone.Control.)</b>		
<b>AcceptPhoneXmlApp ViaHTTP</b>	<p>Controls whether to accept XML App pushed to the phone via HTTP/Post. Choice of:</p> <ul style="list-style-type: none"> <li>• no</li> <li>• yes with authentication</li> <li>• yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlApp ViaOBiTALK</b>	<p>Controls whether to accept XML App pushed to the phone from OBiTALK or PDMS-SP using a SIP/NOTIFY. Choice of:</p> <ul style="list-style-type: none"> <li>• no</li> <li>• yes with authentication</li> <li>• yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlApp ViaSP1</b>	<p>Controls whether to accept XML App pushed to the phone via SP1 using a SIP/NOTIFY. Choice of:</p> <ul style="list-style-type: none"> <li>• no</li> <li>• yes with authentication</li> <li>• yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlApp ViaSP2</b>	<p>Controls whether to accept XML App pushed to the phone via SP2 using a SIP/NOTIFY. Choice of:</p> <ul style="list-style-type: none"> <li>• no</li> <li>• yes with authentication</li> <li>• yes without authentication</li> </ul>	no

Phone Settings (continued)

Parameter	Description	Default Setting
<b>AcceptPhoneXmlAppViaSP3</b>	Controls whether to accept XML App pushed to the phone via SP3 using a SIP/NOTIFY. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlAppViaSP4</b>	Controls whether to accept XML App pushed to the phone via SP4 using a SIP/NOTIFY. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlAppViaSP5</b>	Controls whether to accept XML App pushed to the phone via SP5 using a SIP/NOTIFY. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	no
<b>AcceptPhoneXmlAppViaSP6</b>	Controls whether to accept XML App pushed to the phone via SP6 using a SIP/NOTIFY. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	no
<b>PhoneXmlAppAuthPassword</b>	If authentication is enabled for the corresponding channel, this parameter specifies the authentication password for the pushed Phone XML App requests (HTTP/Post or SIP/NOTIFY) using MD5 digest.	None
<b>AcceptResyncViaOBiTALK</b>	Controls whether to accept NOTIFY/Resync or NOTIFY/Reboot from OBiTALK or PDMS-SP. Choice of: <ul style="list-style-type: none"> <li>no</li> <li>yes with authentication</li> <li>yes without authentication</li> </ul>	no
<b>ResyncViaOBiTALKAuthPassword</b>	If authentication is enabled, this parameter specifies the authentication password for the resync/reboot from OBiTALK or PDMS-SP using MD5 digest.	None

**GUI Menus (VoiceService.1.Phone.GUI.Menus.)**

<b>MainMenu1</b>	A comma-separated list of items to display on the phone's Main menu.	directories, calls, call-histories, all-preferences, settings, prod-info, messages, buddy
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Phone Settings (continued)

Parameter	Description	Default Setting
<b>PreferencesMenu1</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu. Note: The values of <b>PreferencesMenu1</b> through <b>PreferencesMenu6</b> are concatenated together to form a complete list of items to be displayed.	language;Language,timeFormat;TimeFormat,dateFormat;DateFormat,edialer;EnhancedDialer,skin;Skin,bgpic;BackgroundPicture,dring;DefaultRingtone,hpTime;AutoHome Page
<b>PreferencesMenu2</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu.	sdim,sdimDelay,ssvr;ScreenSaver,ssvrDelay;ScreenSaverDelay,ssvrType;ScreenSaverType,ssvrType2,ssvrLock;Require Passcode OnWake Up,ssvrPass;WakeUpPasscode,ssvrCustom;ScreenSaver Show CustomContents
<b>PreferencesMenu3</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu.	brightness;ScreenBrightness,dfont;Font,appCols;Home AppColumns,packcalls;PackCalls OnDisplay,audioDevice;Preferred AudioDevice,headsetDevice;Preferred HeadsetDevice,ehs;Electronic Hook Switch,dnd;Do Not Disturb,dnr;Do Not Ring
<b>PreferencesMenu4</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu.	cfa,cfb,cfna,cwa;CallWaiting,bac;Block AnonymousCall,bci;AnonymousCall,aans;Auto Answer Page,pg1;Join Page Group 1,pg2;Join Page Group 2,ringerVol;Ringer Volume

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>PreferencesMenu5</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu.	speakerVol;Speakerphone Volume,micGain;Speakerphone Mic Gain,handsetVol;Handset Volume,handsetGain;Handset Mic Gain,headsetRJ9Vol,RJ9 Headset Volume,headsetRJ9Gain;RJ9 Headset Mic Gain,headsetRJ9Side;RJ9 Headset Sidetone
<b>PreferencesMenu6</b>	A comma-separated list of items to display on the phone's <b>Preferences</b> Menu.	headsetUSBVol;USB Headset Volume,headsetUSBGain;USB Headset Mic Gain,headsetAEC;AEC for RJ9 Headset,headsetENS;Acoustic Fence for RJ9 Headset,handsetENS;Acoustic Fence for Handset,swapspkhs,swapspkhsled
<b>SettingsMenu1</b>	A comma-separated list of items to display on the phone's <b>Settings</b> Menu. Note: The values of <b>SettingsMenu1</b> and <b>SettingsMenu2</b> are concatenated together to form a complete list of items to be displayed.	net;Network,wifi;WiFi,obile;OBiLine,flash;Storage,ringfile;Ringtones,pkeys;Programmable Keys
<b>SettingsMenu2</b>	A comma-separated list of items to display on the phone's <b>Settings</b> Menu.	lkeys;Line Keys;sidecar1;Side Car 1,sidecar2;Side Car 2,voice;Voice Services,sd99,clrcache,admin;Device Admin,login
<b>ProductInfoMenu1</b>	A comma-separated list of items to display on the phone's <b>Product Info</b> Menu. Note: The values of <b>ProductInfoMenu1</b> and <b>ProductInfoMenu2</b> are concatenated together to form a complete list of items to be displayed.	ipaddr,model;Model,obinumber;OBi Number,mac;MAC Address,wfmac;WiFi MAC Address,serial;Serial Number
<b>ProductInfoMenu2</b>	A comma-separated list of items to display on the phone's <b>Product Info</b> Menu.	swver;Software Version,hwver;Hardware Version,ztinfo;Customization Status,uptime;Up Time



Phone Settings (continued)

Parameter	Description	Default Setting
<b>NetServicesMenu1</b>	A comma-separated list of items to display on the phone's <b>Net Services</b> Menu, also known as <b>Per Line Preferences</b> Menu. Net Services is short for Network-Provided Services.	regst,ringtone,acd,bac,bci,bwanw,buddy,ccs,cfa,cfb,cfna,rec,dnd,dispcode,exec,xass,hotel,clog,dir,rmoff,secClass,simring
<b>NetDirectoryMenu1</b>	A comma-separated list of items to display on the phone's <b>Network Directory</b> (Net Dir for short) Menu.	name,firstName,lastName,ext,number,email,groupId,department,mobile,userId,impId,presence
<b>DeviceAdministrationMenu1</b>	A comma-separated list of items to display on the phone's <b>Device Administration</b> Menu.	admin.port,admin.adminPassword,admin.userPassword,admin.syslogServer,prov.itsp.Method,prov.itsp.Interval,prov.itsp.ConfigURL,fwup.Method,fwup.Interval,fwup.FirmwareURL
<b>FactoryResetRequireAdminPassword</b>	Requires you to enter the Web Administrator Password to factory reset the phone from the phone UI.	True
<b>CustomDictionary</b>	A dictionary XML to replace built-in GUI messages with custom messages.	None

**Calling Features (VoiceService.1.Phone.CallingFeatures.)**

<b>MWIEnable</b>	Enables MWI Signal (stutter dial tone) generation. If enabled, any SP voice service enabled on the phone that has MWI Service enabled triggers the generation of stutter dial tone if there are new voicemails for the subscriber on the service provider's voicemail system.	True
<b>VMWIEnable</b>	Enables VMWI Signal generation. If enabled, any SP voice service enabled on the phone that has VMWI Service enabled triggers the generation of VMWI signal if there are new voicemails for the subscriber on the service provider's voicemail system.	True

Phone Settings (continued)

Parameter	Description	Default Setting
<b>CallTransferEnable</b>	<p>Enables Call Transfer. You can start Call Transfer by hanging up the phone in one of the following scenarios:</p> <ul style="list-style-type: none"> <li>• One call on hold while a second outgoing call ringing</li> <li>• One call on hold while a second outgoing call connected</li> <li>• One call connected while a second outgoing call ringing</li> <li>• 3-way conference with both calls connected</li> </ul> <p>If Call Transfer is disabled, hanging up the phone in the above scenarios ends all the calls except for the one that is holding, which remains on hold (the first and second cases above).</p>	True
<b>ConferenceCallEnable</b>	<p>Enables 3-way Conference Call with local audio mixing. You can start a Conference Call by hook flashing the phone in one of the following scenarios:</p> <ul style="list-style-type: none"> <li>• One call on hold while a second outgoing call ringing</li> <li>• One call on hold while a second outgoing call connected</li> </ul> <p>The first case is an early conference. The second conferee is still ringing. The other two parties can converse while hearing ringback tone in the background until the third party answers. In either case, you can end the call with the second conferee by hook flashing another time and the call reverts to a 2-way call.</p> <p>If Conference Call service is disabled, then hook flashing the phone resumes the holding call but ends the second outgoing call in the first case, and swaps between the two calls in the second case, as in a call waiting situation.</p>	True
<b>UseExternalConferenceBridge</b>	<p>Uses the external conference bridge when starting a conference call.</p>	False
<b>CallWaitingEnable</b>	<p>Enables call waiting service. Call Waiting is the situation where a new incoming call is routed to the phone when there is already another call connected. If this service is enabled, the phone plays the call-waiting tone to alert you and generates the CWCID signal if CWCID is enabled. Then you can then swap between the two calls by hook flashing. If the service is disabled, the phone rejects the incoming call as busy.</p> <p>Note: You can set this parameter from the phone using a Star Code.</p>	True

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>DoNotDisturbEnable</b>	Enables the Do Not Disturb (DND) phone feature.	False
<b>DoNotRingEnable</b>	Enables the Do Not Ring phone feature.	False
<b>CallParkRingEnable</b>	Enables playing of a short ring burst at configurable regular intervals for a configurable total duration wherever there is parked call in any of the call park orbit being monitored with a feature key (using the Call Park Monitor function).	False
<b>CallHoldRingEnable</b>	Enables playing a short ring burst at configurable regular intervals for a configurable total duration whenever there is a call on hold.	False
<b>AnonymousCallBlockEnable</b>	Enables the Anonymous Call Block feature.	False
<b>AnonymousCallEnable</b>	Enables the Anonymous Call feature.	False
<b>CallForwardUnconditionalEnable</b>	Enables the Call Forward All Calls feature.	False
<b>CallForwardUnconditionalNumber</b>	Target number to forward all callers.	None
<b>CallForwardOnBusyEnable</b>	Enables the Call Forward on Busy feature.	False
<b>CallForwardOnBusyNumber</b>	Target number to forward all callers on busy.	None
<b>CallForwardOnNoAnswerEnable</b>	Enables the Call Forward on No Answer feature.	False
<b>CallForwardOnNoAnswerNumber</b>	Target number to forward all callers when not answered after a few rings.	None
<b>CallForwardOnNoAnswerRingCount</b>	Number of rings to wait before activating call forward on no answer.	2
<b>JoinPageGroup1</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 1.	False
<b>JoinPageGroup2</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 2.	False
<b>JoinPageGroup3</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 3.	False
<b>JoinPageGroup4</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 4.	False
<b>JoinPageGroup5</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 5.	False

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>JoinPageGroup6</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 6.	False
<b>JoinPageGroup7</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 7.	False
<b>JoinPageGroup8</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 8.	False
<b>JoinPageGroup9</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 9.	False
<b>JoinPageGroup10</b>	Join (if enabled) or Leave (if disabled) Multicast Page Group 10.	False
<b>HoldCallsOnPage</b>	Put calls on hold when there is an incoming group page.	False
<b>AutoAnswerEnable</b>	Enables auto-answering an incoming call for paging application.	True
<b>ToneProfile</b>	Selects a Tone Profile for call progress tone generation. Choices are A or B.	A
<b>StarCodeProfile</b>	Selects a Star Code Profile for interpreting the Star Codes you enter. Choice of: <ul style="list-style-type: none"> <li>• None</li> <li>• A</li> <li>• B</li> </ul> If the value is set to None, no star code is recognized by the phone.	A
<b>LastDialedNumber</b>	This read-only status parameter shows the last number dialed out on the phone.	
<b>LastCallerNumber</b>	This read-only status parameter shows the last number to ring the phone.	
<b>AcceptMediaLoopback</b>	Enables the phone to accept incoming media loopback calls.	True
<b>MediaLoopbackAnswerDelay</b>	Delay in milliseconds before the phone answers an incoming media loopback call.	0
<b>MediaLoopbackMaxDuration</b>	Maximum duration in seconds to allow for an inbound media loopback call. Set the value to blank or 0 to make it unlimited.	0
<b>RepeatDialInterval</b>	Interval in seconds between redial in a repeat dial operation.	30
<b>RepeatDialExpires</b>	Duration of time in seconds when a repeat dial operation remains active.	1800

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>MOHServiceNumber</b>	A number to call and bridge to get some audio played when the phone user holds the call	None
<b>PlaySITOnCallFailureCodes</b>	Specify which INVITE failure response codes trigger the phone to play SIT tone to alert the phone user. It must be specified with a valid digit map format. If nothing valid is specified, the phone plays the normal reorder tone when the outbound call fails.	(404 9xx)
<b>InvalidNumberAnnouncementOption</b>	Option to play an announcement of the error when user dials an invalid number. Choice of: <ul style="list-style-type: none"> <li>SIT_ONLY</li> <li>DEFAULT_ONLY</li> <li>SIT_AND_DEFAULT</li> <li>CUSTOM_ONLY</li> <li>SIT_AND_CUSTOM</li> </ul>	SIT_ONLY
<b>CustomInvalidNumberAnnouncement</b>	The custom announcement to play when user dials an invalid number.	None
<b>CallRemovalDelayOnPeerEndCall</b>	Number of seconds to delay removal of the call from the screen when the far end ends the call. During that delay, reorder tone is played if the call was connected (active) when it was ended	0
<b>CallHistoryUploadURL</b>	URL to upload call history.	None
<b>CallHistoryUploadFilter</b>	A comma-separated list of services to include in the call history upload.	sp1, sp2, sp3, sp4, sp5, sp6, pp
<b>MergeConfereeCalls</b>	Enables grouping all the active conferee calls into a single call item and not showing each individual conferee call items on screen. The call state of this virtual call item is always <b>Conferencing</b> . Pressing the <b>End</b> softkey on this special call item ends all the conferee calls. Pressing the <b>Split</b> or <b>NewCall</b> softkey splits it up into individual conferee call items on screen while placing each conferee call on hold. Pressing the <b>Join</b> softkey lets you drop out of the conference while transferring one conferee to the other so they can continue with the conversation.	False
<b>TransferOnHangUpConference</b>	Enables transferring one conferee call to the other (in other words, Join the two conferees) when hanging up on a merged conference call. Otherwise, all the conferees are ended just like pressing the <b>End</b> softkey. Hanging up means either on-hook the handset while using the handset, turn off the speakerphone while using the speakerphone, or turn off the headset while using a headset.	False

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>ConfirmHangUpConference</b>	Ask for confirmation before ending a conference call.	False
<b>XslRefresh</b>	Enables Refresh option for some XSL functions.	False
<b>EmergencyCallActionURL</b>	The Action URL to invoke when making an emergency call.	None
<b>Network Services (VoiceService.1.Phone.NetServices.)</b>		
<b>Enable</b>	Makes this option available on the phone Main menu.	True
<b>Name</b>	A user-friendly name to identify this directory on the Phone screen.	None
<b>Network Directory (VoiceService.1.Phone.NetDir.)</b>		
<b>Enable</b>	Makes this option available on the phone Main menu.	True
<b>Name</b>	A user-friendly name to identify this directory on the Phone screen.	None
<b>VoiceService</b>	The Service that provides the directory. Choice of: <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• LDAP</li> </ul>	SP1
<b>Buddy List (VoiceService.1.Phone.BuddyList.)</b>		
<b>Enable</b>	Makes this option available on the phone Main menu.	True
<b>Name</b>	A user-friendly name to identify this buddy list on the Phone screen.	None
<b>VoiceService</b>	The Service that provides the directory. Choice of: <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• LDAP</li> </ul>	SP1
<b>MyStatusHistory</b>	A comma-separated list of (XMPP) Status that you can select from the phone UI to be used as the current (XMPP) Status.	None

**User Preferences Settings (VoiceService.1.Phone.SoftKeys.)**

Phone Settings (continued)

Parameter	Description	Default Setting
<b>CallForwardUnconditionalFeatureProvider</b>	<p>Selects the provider of the Call Forward Unconditional feature under User Preferences. Choice of:</p> <ul style="list-style-type: none"> <li>• Phone</li> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	Phone
<b>CallWaitingFeatureProvider</b>	<p>Selects the provider of the Call Waiting feature under User Preferences. Choice of:</p> <ul style="list-style-type: none"> <li>• Phone</li> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	Phone
<b>DoNotDisturbFeatureProvider</b>	<p>Selects the provider of the Do Not Disturb feature under User Preferences. Choice of:</p> <ul style="list-style-type: none"> <li>• Phone</li> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	Phone
<b>AnonymousCallBlockFeatureProvider</b>	<p>Selects the provider of the Block Anonymous Callers feature under User Preferences. Choice of:</p> <ul style="list-style-type: none"> <li>• Phone</li> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	Phone

Phone Settings (continued)

Parameter	Description	Default Setting
<b>AnonymousCallFeatureProvider</b>	Selects the provider of the Anonymous Call feature under User Preferences. Choice of: <ul style="list-style-type: none"> <li>Phone</li> <li>SP1</li> <li>SP2</li> <li>SP3</li> <li>SP4</li> <li>SP5</li> <li>SP6</li> <li>OBiTALK</li> </ul>	Phone
<b>MessageStatusFeatureProvider</b>	Selects the provider of the primary Message Status displayed on the phone. Choice of: <ul style="list-style-type: none"> <li>Phone</li> <li>SP1</li> <li>SP2</li> <li>SP3</li> <li>SP4</li> <li>SP5</li> <li>SP6</li> </ul>	Phone
<b>ACDFeatureProvider</b>	Selects the provider of the ACD Agent State feature to be shown on the phone's status bar area. Choice of: <ul style="list-style-type: none"> <li>SP1</li> <li>SP2</li> <li>SP3</li> <li>SP4</li> <li>SP5</li> <li>SP6</li> </ul>	SP1

**Timers (VoiceService.1.Phone.Timer)**

<b>ReorderDelayTime</b>	Delay in ms to start reorder tone after peer hangs up.	5500
<b>DigitMapLongTimer</b>	Default number of seconds when the digit map processor should timeout waiting for more digits for matching patterns with unspecified length.	10
<b>DigitMapShortTimer</b>	Default number of seconds when the digit map processor should timeout waiting for more digits when at least one pattern with specific length has matched.	2

**Page Group n (VoiceService.1.Phone.PageGroup.n.), n = 1, 2, 3, 4, 5, 6, 7, 8, 9, 10**

<b>GroupName</b>	A user-friendly name to identify the multicast page group.	None
<b>MulticastAddress</b>	The multicast address to use for the page group.	224.1.1.100



Phone Settings (continued)

Parameter	Description	Default Setting
<b>MulticastPort</b>	The multicast port to use for the page group.	PG1: 65322 PG2: 65324
<b>TTL</b>	TTL of multicast packet (1-255).	2
<b>ParticipantName</b>	A human-readable name to identify this participant to the group.	None
<b>AudioCodec</b>	The audio codec to use for the page group. Choice of: <ul style="list-style-type: none"> <li>G711U</li> <li>G711A</li> <li>G729</li> <li>iLBC</li> <li>G722</li> <li>OPUS</li> </ul>	G711U
<b>TxPacketSize</b>	RTP transmission packet size in milliseconds.	20
<b>RTCPTxInterval</b>	RTCP transmission interval when talking in milliseconds. PG1 only.	0
<b>SilenceSuppression</b>	Enables silence suppression when talking. PG1 only.	False
<b>PushToTalk</b>	Enables push to talk mode. PG1 only.	False
<b>PlayToneOnIncomingPage</b>	Play a short Paging Tone on receiving a new incoming page. PG1 only.	True
<b>PreferredAudioDevice</b>	PG1 only. Choice of: <ul style="list-style-type: none"> <li>Speaker</li> <li>Headset</li> <li>System</li> </ul>	System
<b>PageTimeout</b>	Timeout in seconds to automatically end an outgoing page. Set to 0 to disable this automatic timeout mechanism. PG1 only.	0
<b>EnableJoinLeaveGroupUserControl</b>	Enables you option to join or leave the group by pressing the key quickly. PG1 only.	True
<b>SpeakerVolume</b>	Speaker volume when paging. 0 follows the current ringer volume.	0
<b>Stats</b>	Three statistics: <ul style="list-style-type: none"> <li>Packets received in the past hour</li> <li>Packets received in the past minute</li> <li>Packets received in the past 10 seconds</li> </ul> Updated once every 10 seconds.	0, 0, 0

**Transcription Service (VoiceService.1.Phone.SpeechToText)**

## Phone Settings (continued)

Parameter	Description	Default Setting
<b>URL</b>	<p>URL for the transcription service.</p> <p>For example:  <code>https://mynlp-service.abc.com/test,no-verify</code></p> <p><b>Note:</b> The keyword <i>no-verify</i> is an option to tell the option not to verify the server certificate on the HTTPS connection, in the case that the certificate may not be signed by a CA known to the phone. However, this option should not be used to ensure security of the connection.</p> <p>The keyword <b>vb</b> (Voicebase) or <b>dg</b> (Deepgram) is for the NLP platform through which you can stream the transcription audio.</p> <p>The keyword <b>save</b> is an option that allows you to save the transcription against the call logs.</p>	
<b>Key</b>	The authentication key is required to use the transcription service. The key is provided by your transcription service provider.	
<b>Secret</b>	The secret for authentication. The secret is provided by your transcription service provider.	
<b>Config</b>	<p>Configuration string for setting service parameters.</p> <p>The keyword <b>dir</b> specifies the direction to transcribe from the perspective of the local phone user, with one of the following values:</p> <ul style="list-style-type: none"> <li>• <b>t</b> – Transmit (i.e. the sending or outgoing) direction only,</li> <li>• <b>r</b> – Receive (i.e. incoming) direction only, and</li> <li>• <b>tr</b> – Transmit and receive</li> </ul> <p>The keyword <b>block</b> specifies the size in bytes of audio samples to send out in each block.</p> <p>The keyword <b>alert</b> is for the sound when a transcription call is saved. Alert = cw1, cw2, cw3, up to cw10, where cw is <b>Call Waiting Tone</b>. If the keyword <b>callerid = True</b> the phone sends metadata information to the server.</p> <p>The following keywords are only applicable while transcribing audio through Deepgram:</p> <ul style="list-style-type: none"> <li>• <b>Interim</b> - If interim = false, Deepgram sends the audio results only when the confidence level is highest.</li> <li>• <b>Model</b> - What type of conversation is happening on the phone. It can be general, phone call, or a meeting.</li> </ul>	<code>dir=tr,block=2048</code>

## Left Line Keys Web Page

The following configuration parameters are available on this page.

### Left Line Keys Settings

Parameter	Description	Default Setting
<i>Left Line Key n (VoiceService.1.Phone.LeftLineKey.n), n = 1 – 2 for VVX 150, 1 – 16 for VVX 250, 1 – 24 for VVX 350, 1 – 24 for VVX 450</i>		
<b>InvalidKeyHandling</b>	<p>Selects the method to handle an invalid key that has its <b>Service</b> field specified with a service that has not been configured. Applies unconditionally to all Line Keys, Left and Right. Choice of:</p> <ul style="list-style-type: none"> <li>• Unmodify</li> <li>• Pack</li> <li>• Disable</li> </ul>	Unmodify
<b>HiddenTabActivityIndicator</b>	Enables indication of Busy Lamp Field or Call Park activities under a hidden tab by blinking the Tab Number. Applies unconditionally to all Line Keys, Left and Right.	False

**Left Line Keys Settings (continued)**

Parameter	Description	Default Setting
<b>Function</b>	<p>Function assigned to this key. Can be set for each Left Line Key. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto Answer Intercom</li> <li>• ACD Sign On/Off</li> <li>• Blind Transfer</li> <li>• Block Anonymous Call</li> <li>• Block Caller ID</li> <li>• Busy Lamp Field</li> <li>• Call Appearance</li> <li>• Call Forward</li> <li>• Call Park Monitor</li> <li>• Call Waiting</li> <li>• Conference</li> <li>• Disposition Code</li> <li>• Do Not Disturb</li> <li>• Do Not Ring</li> <li>• Exec Filter On/Off</li> <li>• Exec Assistant</li> <li>• Hold</li> <li>• Hoteling</li> <li>• Line Monitor</li> <li>• Next Tab</li> <li>• Message Status</li> <li>• Page Group 1 through Page Group 10</li> <li>• Presence Monitor</li> <li>• Security Class</li> <li>• Speed Dial</li> <li>• Transfer</li> <li>• Action URL</li> <li>• Add To Conference</li> <li>• Blind Transfer 2</li> <li>• Left Next Tab</li> </ul>	Call Appearance
<b>Service</b>	<p>The Voice Service on which to apply the selected function. Choice of:</p> <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	None

**Left Line Keys Settings (continued)**

Parameter	Description	Default Setting
<b>Number</b>	The number, extension, or user ID of the entity associated with the assigned function.	None
<b>Name</b>	A display name for the entity associated with the assigned function.	None
<b>Group</b>	ID of the Custom Line Key Group to use to form the contents of the line key window.	None
<b>Style</b>	ID of the Custom Line Key Style to use to format the contents of the line key window.	None
<b>MaxCalls</b>	Maximum number of calls this key should handle if the function is Call Appearance.	1
<b>Status</b>	This read-only status parameter shows the last number to ring the phone.	

***Right Line Keys Web Page (V VX 450 only)***

The following configuration parameters are available on this page.

**Right Line Keys Settings**

Parameter	Description	Default Setting
<b>Right Line Key <i>n</i> (VoiceService.1.Phone.LineKey.n.), <i>n</i> = 1 – 24</b>		
<b>Function</b>	<p>Function assigned to this key. Can be set for each Line Key. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto Answer</li> <li>• Intercom</li> <li>• ACD Sign On/Off</li> <li>• Blind Transfer</li> <li>• Block Anonymous Call</li> <li>• Block Caller ID</li> <li>• Busy Lamp Field</li> <li>• Call Appearance</li> <li>• Call Forward</li> <li>• Call Park Monitor</li> <li>• Call Waiting</li> <li>• Conference</li> <li>• Disposition Code</li> <li>• Do Not Disturb</li> <li>• Do Not Ring</li> <li>• Exec Filter On/Off</li> <li>• Exec Assistant</li> <li>• Hold</li> <li>• Hoteling</li> <li>• Line Monitor</li> <li>• Next Tab</li> <li>• Message Status</li> <li>• Page Group 1 through Page Group 10</li> <li>• Presence Monitor</li> <li>• Security Class</li> <li>• Speed Dial</li> <li>• Transfer</li> <li>• Action URL</li> <li>• Add To Conference</li> <li>• Blind Transfer 2</li> <li>• Left Next Tab</li> </ul>	Call Appearance

**Right Line Keys Settings (continued)**

Parameter	Description	Default Setting
<b>Service</b>	The Voice Service on which to apply the selected function. Choice of: <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	None
<b>Number</b>	The number, extension, or user ID of the entity associated with the assigned function.	None
<b>Name</b>	A display name for the entity associated with the assigned function.	None
<b>Group</b>	ID of the Custom Line Key Group to use to form the contents of the line key window.	None
<b>Style</b>	ID of the Custom Line Key Style to use to format the contents of the line key window.	None
<b>MaxCalls</b>	Maximum number of calls this key should handle if the function is Call Appearance.	1
<b>Status</b>	This read-only status parameter shows the last number to ring the phone.	

***Programmable Keys Web Page***

The following configuration parameters are available on this page.

**Programmable Keys Settings**

Parameter	Description	Default Setting
<b>Key n (VoiceService.1.Phone.ProgKey.n.), n = 1 – 2 for VVX 150 and VVX 250, = 1 – 3 for VVX 350 and VVX 450</b>		
<b>Function</b>	Function assigned to this key. Can be set for each Programmable Key. Choice of: <ul style="list-style-type: none"> <li>• Auto Answer</li> <li>• Intercom</li> <li>• ACD Sign On/Off</li> <li>• Blind Transfer</li> <li>• Block Anonymous Call</li> <li>• Block Caller ID</li> <li>• Busy Lamp Field</li> <li>• Call Appearance</li> <li>• Call Forward</li> <li>• Call Park Monitor</li> <li>• Call Waiting</li> <li>• Conference</li> <li>• Disposition Code</li> <li>• Do Not Disturb</li> <li>• Do Not Ring</li> <li>• Exec Filter On/Off</li> <li>• Exec Assistant</li> <li>• Hold</li> <li>• Hoteling</li> <li>• Line Monitor</li> <li>• Next Tab</li> <li>• Message Status</li> <li>• Page Group 1 through Page Group 10</li> <li>• Presence Monitor</li> <li>• Security Class</li> <li>• Speed Dial</li> <li>• Transfer</li> <li>• Action URL</li> <li>• Add To Conference</li> <li>• Blind Transfer 2</li> <li>• Left Next Tab</li> </ul>	Transfer



**Programmable Keys Settings (continued)**

Parameter	Description	Default Setting
<b>Service</b>	The Voice Service on which to apply the selected function. Choice of: <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	None
<b>Number</b>	The number, extension, or user ID of the entity associated with the assigned function.	None
<b>Name</b>	A display name for the entity associated with the assigned function.	None
<b>Group</b>	ID of the Custom Line Key Group to use to form the contents of the line key window.	None
<b>Style</b>	ID of the Custom Line Key Style to use to format the contents of the line key window.	None
<b>Status</b>	This read-only status parameter shows any applicable status for the assigned function.	

**Side Car X Web Page (X = 1, 2)**

The following configuration parameters are available on each page.

**Side Car Settings**

Parameter	Description	Default Setting
<b>Status</b>		
<b>Connected</b>	This read-only status parameter shows if a VVX EM50 side car is connected.	

**Left Line Key n (VoiceService.1.Phone.LeftSideCar.1.Key.n.), n = 1 – 45**

Side Car Settings (continued)

Parameter	Description	Default Setting
<b>Function</b>	<p>Function assigned to this key. Can be set for each Left Line Key. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto Answer Intercom</li> <li>• ACD Sign On/Off</li> <li>• Blind Transfer</li> <li>• Block Anonymous Call</li> <li>• Block Caller ID</li> <li>• Busy Lamp Field</li> <li>• Call Appearance</li> <li>• Call Forward</li> <li>• Call Park Monitor</li> <li>• Call Waiting</li> <li>• Conference</li> <li>• Disposition Code</li> <li>• Do Not Disturb</li> <li>• Do Not Ring</li> <li>• Exec Filter On/Off</li> <li>• Exec Assistant</li> <li>• Hold</li> <li>• Hoteling</li> <li>• Line Monitor</li> <li>• Next Tab</li> <li>• Message Status</li> <li>• Page Group 1 through Page Group 10</li> <li>• Presence Monitor</li> <li>• Security Class</li> <li>• Speed Dial</li> <li>• Transfer</li> <li>• Action URL</li> <li>• Add To Conference</li> <li>• Blind Transfer 2</li> <li>• Left Next Tab</li> </ul>	Speed Dial
<b>Service</b>	<p>The Voice Service on which to apply the selected function. Choice of:</p> <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	None

**Side Car Settings (continued)**

<b>Parameter</b>	<b>Description</b>	<b>Default Setting</b>
<b>Number</b>	The number, extension, or user ID of the entity associated with the assigned function.	
<b>Name</b>	A display name for the entity associated with the assigned function.	
<b>Group</b>	ID of the Custom Line Key Group to use to form the contents of the line key window.	
<b>Style</b>	ID of the Custom Line Key Style to use to format the contents of the line key window.	
<b>MaxCalls</b>	Maximum number of calls this key should handle if the function is Call Appearance.	1
<b>Status</b>	This read-only status parameter shows the last number to ring the phone.	

***Right Line Key n (VoiceService.1.Phone.SideCar.1.Key.n), n = 1 – 45***

Side Car Settings (continued)

Parameter	Description	Default Setting
<b>Function</b>	<p>Function assigned to this key. Can be set for each Left Line Key. Choice of:</p> <ul style="list-style-type: none"> <li>• Auto Answer Intercom</li> <li>• ACD Sign On/Off</li> <li>• Blind Transfer</li> <li>• Block Anonymous Call</li> <li>• Block Caller ID</li> <li>• Busy Lamp Field</li> <li>• Call Appearance</li> <li>• Call Forward</li> <li>• Call Park Monitor</li> <li>• Call Waiting</li> <li>• Conference</li> <li>• Disposition Code</li> <li>• Do Not Disturb</li> <li>• Do Not Ring</li> <li>• Exec Filter On/Off</li> <li>• Exec Assistant</li> <li>• Hold</li> <li>• Hoteling</li> <li>• Line Monitor</li> <li>• Next Tab</li> <li>• Message Status</li> <li>• Page Group 1 through Page Group 10</li> <li>• Presence Monitor</li> <li>• Security Class</li> <li>• Speed Dial</li> <li>• Transfer</li> <li>• Action URL</li> <li>• Add To Conference</li> <li>• Blind Transfer 2</li> <li>• Left Next Tab</li> </ul>	Speed Dial
<b>Service</b>	<p>The Voice Service on which to apply the selected function. Choice of:</p> <ul style="list-style-type: none"> <li>• SP1</li> <li>• SP2</li> <li>• SP3</li> <li>• SP4</li> <li>• SP5</li> <li>• SP6</li> <li>• OBiTALK</li> </ul>	None

**Side Car Settings (continued)**

Parameter	Description	Default Setting
<b>Number</b>	The number, extension, or user ID of the entity associated with the assigned function.	None
<b>Name</b>	A display name for the entity associated with the assigned function.	None
<b>Group</b>	ID of the Custom Line Key Group to use to form the contents of the line key window.	None
<b>Style</b>	ID of the Custom Line Key Style to use to format the contents of the line key window.	None
<b>MaxCalls</b>	Maximum number of calls this key should handle if the function is Call Appearance.	1
<b>Status</b>	This read-only status parameter shows the last number to ring the phone.	None

## LED Settings Web Page

The following table lists the meanings for LED default settings. When more than one LED setting is listed, it means that the LED is cycling between those states.

For example, *R50, X50* means the LED is flashing red (cycling between red and off).

**LED Meanings**

LED Default Setting	Description
<b>R</b>	LED is red.
<b>X</b>	LED is turned off.
<b>n, nn, nnn, nnnn</b>	Where <i>n</i> represents a number from 0 to 9, any string of <i>n</i> represents that value in ms.

The following table lists the meanings for LED displays on line keys.

**LED Settings**

Parameter	Description	Default Setting
<i>Call State (VoiceService.1.Phone.LED.Call.)</i>		
<b>Idle</b>	No call.	X
<b>Dialing</b>	Dialing a number.	G
<b>Trying</b>	Trying an outgoing call before the called party rings.	G
<b>PeerRingng</b>	Called party is ringing.	G50, X50
<b>Connected</b>	Call connected.	G

**LED Settings (continued)**

<b>Parameter</b>	<b>Description</b>	<b>Default Setting</b>
<b>Ringing</b>	Incoming call ringing.	R250, X150
<b>Holding</b>	Holding the call.	R500, X500
<b>Error</b>	Call failed.	G500, X500
<b>CallParked</b>	A call is parked on the bound extension.	R100, X1000
<b>ServiceDown</b>	Service is down.	X

**SCA (VoiceService.1.Phone.LED.SCA.)**

<b>Idle</b>	SCA not used by anyone.	X
<b>Seized</b>	SCA has been seized by another user.	R150, X150
<b>Trying</b>	Another user is trying an outgoing call on the SCA.	R
<b>PeerRinging</b>	The call peer is ringing.	R
<b>Connected</b>	Call connected.	R
<b>Held</b>	Holding the call.	G500, X500
<b>PrivateHeld</b>	Another is user is holding the call as private.	R500, X500
<b>ServiceDown</b>	SCA Service is down.	X

**BLF State (VoiceService.1.Phone.LED.BLF.)**

<b>Idle</b>	Monitored extension is idle.	X
<b>CallParked</b>	A call is parked against the monitored extension.	R100, X1000
<b>Ringing</b>	Monitored extension is ringing.	R520, X150
<b>Busy</b>	Monitored extension is on a call.	R
<b>Holding</b>	Monitored extension is holding a call.	R500, X500
<b>ServiceDown</b>	BLF Service is down.	X

**Service State (VoiceService.1.Phone.LED.Service.)**

<b>Idle</b>	No calls.	G
<b>InUse</b>	One or more calls connected or trying.	G500, X500
<b>Ringing</b>	One or more incoming calls ringing.	R250, X150
<b>Holding</b>	One or more calls holding.	R500, X500
<b>ServiceDown</b>	Service is down.	X

**ACD Agent State (VoiceService.1.Phone.LED.ACD.)**

<b>LoggedOff</b>	Agent has logged off.	X
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## LED Settings (continued)

Parameter	Description	Default Setting
<b>Available</b>	Agent logged on and is available.	G
<b>Unavailable</b>	Agent logged on but isn't available.	R
<b>WrappingUp</b>	Agent logged on but is wrapping up last call.	R500, X500
<b>ServiceDown</b>	Service is down.	X
<b>LoggedOn</b>	The agent just logged on.	R
<b><i>Presence (VoiceService.1.Phone.LED.Presence.)</i></b>		
<b>Offline</b>	Same as Invisible.	X
<b>Online</b>	Online or Available.	G
<b>Busy</b>	Same as DND or Do Not Disturb.	R
<b>Away</b>	Away.	G500, X500
<b>ExtendedAway</b>	Same as XA.	R500, X500
<b>ServiceDown</b>	Service is down.	X
<b><i>Feature State (VoiceService.1.Phone.LED.FeatureKey.)</i></b>		
<b>AnonymousCallEnabled</b>	Anonymous Call feature is enabled.	R
<b>AnonymousCallDisabled</b>	Anonymous Call feature is disabled.	X
<b>AnonymousCallServiceDown</b>	Anonymous Call service is down.	X
<b>AnonymousCallBlockEnabled</b>	Block Anonymous Callers feature is enabled.	R
<b>AnonymousCallBlockDisabled</b>	Block Anonymous Callers feature is disabled.	X
<b>AnonymousCallBlockServiceDown</b>	Block Anonymous Callers service is down.	X
<b>AutoAnswerIntercomEnabled</b>	Auto Answer Intercom feature is enabled.	X
<b>AutoAnswerIntercomDisabled</b>	Auto Answer Intercom feature is disabled.	R
<b>CallForwardEnabled</b>	Call Forward feature is enabled.	R
<b>CallForwardDisabled</b>	Call Forward feature is disabled.	X
<b>CallForwardServiceDown</b>	Call Forward service is down.	X

## LED Settings (continued)

Parameter	Description	Default Setting
<b>CallParkedYes</b>	Call is parked on line.	R
<b>CallParkedNo</b>	No Call parked on line.	X
<b>CallParkMonitorServiceDown</b>	Call Park Monitor service is down.	X
<b>CallWaitingEnabled</b>	Call Waiting feature is enabled.	X
<b>CallWaitingDisabled</b>	Call Waiting feature is disabled.	R
<b>DoNotDisturbEnabled</b>	Do Not Disturb feature is enabled.	R
<b>DoNotDisturbDisabled</b>	Do Not Disturb feature is disabled.	X
<b>DoNotDisturbServiceDown</b>	Do Not Disturb service is down.	X
<b>DoNotRingEnabled</b>	Do Not Ring feature is enabled.	R
<b>DoNotRingDisabled</b>	Do Not Ring feature is disabled.	X
<b>ExecFilterEnabled</b>	Executive (Call) Filter feature is enabled.	R
<b>ExecFilterDisabled</b>	Executive (Call) Filter feature is disabled.	X
<b>ExecFilterServiceDown</b>	Executive (Call) Filter service is down.	X
<b>ExecAssistEnabled</b>	Executive Assistant feature is enabled.	R
<b>ExecAssistDisabled</b>	Executive Assistant feature is disabled.	X
<b>ExecAssistDivertOn</b>	Executive Assistant Call Divert is on.	R100, X500
<b>ExecAssistServiceDown</b>	Executive Assistant Call Divert is off.	X
<b>HotelingGuestLoggedOn</b>	Hoteling Guest has logged on.	R
<b>HotelingGuestLoggedOff</b>	Hoteling Guest has logged off (same as No Hoteling Guest has logged on).	X
<b>HotelingServiceDown</b>	Hoteling Guest logon/logoff service is down.	X
<b>NewMessagesWaitingYes</b>	New messages waiting.	R
<b>NewMessagesWaitingNo</b>	No new messages waiting.	X
<b>MWIServiceDown</b>	MWI (message waiting indication) service is down.	X
<b>PageGroupJoined</b>	Joined Page Group.	R
<b>PageGroupLeft</b>	Left Page Group.	X



**LED Settings (continued)**

Parameter	Description	Default Setting
<b>PageGroupMeTalking</b>	Currently I am talking to the page group.	R500,X500
<b>PageGroupThemTalking</b>	Currently someone else is talking to the page group.	R50,X50
<b>SecurityClassServiceDown</b>	Security Classification service is down.	X
<b>SpeedDialOn</b>	Speed dial is enabled.	X
<b>SpeedDialOff</b>	Speed dial is disabled.	X
<b>VMWI Lamp (VoiceService.1.Phone.LED.VMWI.)</b>		
<b>NewMessagesWaiting Yes</b>	New messages available.	R
<b>NewMessagesWaiting No</b>	No new messages available	X
<b>Disabled</b>	VMWI is disabled.	X
<b>Ringing</b>	Phone is ringing.	None

## Soft Keys Web Page

The following configuration parameters are available on this page.

**Soft Keys Settings**

Parameter	Description	Default Setting
<b>Soft Key Set (VoiceService.1.Phone.SoftKeySet.)</b>		
<b>Home</b>	Home screen.	redial,cfa,dnd,missed lines
<b>SCAInUse</b>	SCA being used by another phone.	sca.barge ,sca.monitor ,,newcall
<b>CallParked</b>	Call is parked.	pickup,,newcall
<b>CallConnected</b>	Call is connected.	end,hold,totrs,tospk,conf,transfer,privhold,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume,tomobile
<b>CallHolding</b>	Call is holding.	end,resume,add2conf,conf,transfer,park,dispcode,escalate,trace,rec.start,rec.stop,rec.pause,rec.resume,tomobile
<b>Dialtone</b>	Dial tone playing.	redial,phbk,mode,lines

**Soft Keys Settings (continued)**

Parameter	Description	Default Setting
Dialing	Dialing a number.	redial,dial,mode,backspace
OnDialing	Dialing a number while on-hook.	switch.line,dial,mode,backspace
Ringing	Incoming call is ringing.	answer,reject,ignore
CallConnecting	Trying outgoing or called party is ringing.	end,,,newcall
ConfTrying	Calling 2nd conferee.	end
ConfRinging	2nd Conferee ringing.	end,,conf.now
ConfConnected	Connected with 2nd conferee.	end,hold,resume,conf.now
ConfBridgeControl	Connected with an external conference bridge.	end,deaf,mute
Conferencing	Merged conference call item.	end,split,newcall
XferTrying	Calling transfer target.	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.acd
BLFCall	BLF Call options.	blf.call,blf.answer,blf.pickup,blf.barge,blf.coach,blf.monitor
Settings	Settings menu.	reboot,reset,fwupdate

## LDAP Search Web Page

The following configuration parameters are available on this page.

**LDAP Search Settings**

Parameter	Description	Default Setting
<i>IP Phone – LDAP – Server (X_DeviceManagement.LDAPServer.)</i>		
Host	IP address or domain name of host, with optional ldap:// or ldaps:// scheme prefix.	None.
Port	LDAP Server listening (TCP) port. The standard port is 389 for ldap:// and 636 for ldaps://. If the port value is 0 or blank, the phone uses the corresponding standard port.	0
Password	Password for authentication with the LDAP server.	None.

## LDAP Search Settings (continued)

Parameter	Description	Default Setting
<b>TLSSecurityProfile</b>	Security profile when using HTTPS. Choices are 1 or 2.	1
<b><i>IP Phone – LDAP – Search Parameters (X_DeviceManagement.LDAPSearch.)</i></b>		
<b>BindDN</b>	A Distinguished Name (DN) that is authorized to use the LDAP service.	None.
<b>SearchBase</b>	Specifies the starting point of the LDAP search.	None.
<b>ProtocolVersion</b>	Protocol version. Choice of: <ul style="list-style-type: none"> <li>• 2</li> <li>• 3</li> </ul>	3
<b>TLS_ReqCert</b>	Controls whether to verify the server's certificate on a TLS connection. Choice of: <ul style="list-style-type: none"> <li>• never</li> <li>• demand</li> </ul>	never
<b>ResultsPerPage</b>	Specifies how many results to display on screen per page. Choice of: <ul style="list-style-type: none"> <li>• 20</li> <li>• 40</li> <li>• 60</li> <li>• 80</li> <li>• 100</li> </ul>	20
<b>DefaultSearchFilter</b>	Default search filter to append to each search.	(objectclass=*)
<b>QueryFields</b>	A comma-separated list of user input LDAP attributes/Display-Name to form the query filter.	givenName/First Name,sn/LastName,telephoneNumber/Tel,mobile/Mobile,homePhone/Home
<b>ResultFields</b>	Comma-separated list of LDAP fields to display for each entry of the search result.	cn/Common Name,sn/LastName,givenName/FirstName,telephoneNumber/Tel/y,mobile/Mobile/y,homePhone/Home/y
<b>NameFieldPreference</b>	Comma-separated list of LDAP attributes to be used as the Caller ID Name to display on screen, ordered by preference.	cn,givenName sn
<b>NumberFieldPreference</b>	Comma-separated list of LDAP attributes to be used as the Number to display on screen and to call by default, ordered by preference.	telephoneNumber,mobile,homePhone
<b>PhotoFieldPreference</b>	Comma-separated list of LDAP attributes to be used as the photo to display on screen, ordered by preference.	thumbnailPhoto

**LDAP Search Settings (continued)**

Parameter	Description	Default Setting
<b>SortByAttribute</b>	The LDAP attribute to use for sorting the search results by the server.	cn
<b>Attributes</b>	Comma-separated list of attributes to request from server.	None

***IP Phone – LDAP – SASL Authentication Parameters (X\_DeviceManagement.LDAPSearchSASL.)***

<b>SASL_AuthMethod</b>	Method to use for SASL authentication. If the value is <code>Disabled</code> , the phone uses Simple authentication only. Choice of: <ul style="list-style-type: none"> <li>• <code>Disabled</code></li> <li>• <code>Plain</code></li> <li>• <code>MD5</code></li> </ul>	Disabled
<b>SASL_AuthCID</b>	The authentication ID for SASL authentication.	None.

## Line Key Customization Web Page

The following configuration parameters are available on this page.

**Line Key Customization Settings**

Parameter	Description	Default Setting
<b><i>Call Appearance (VoiceService.1.Phone.CstmLineKey.Call.)</i></b>		
<b>Enable</b>	Boolean. Enables using customized feature key display format for Call Appearance key function.	False
<b>TextLine1</b>	Use this text for line 1 when the key has no calls.	@text1
<b>TextLine2</b>	Use this text for line 2 when the key has no calls.	@text2
<b>TextLine1InCall</b>	Use this text for line 1 when the key has one or more calls.	@text1
<b>TextLine2InCall</b>	Use this text for line 2 when the key has one or more calls.	@text2
<b><i>Busy Lamp Field (VoiceService.1.Phone.CstmLineKey.BLF.)</i></b>		
<b>Enable</b>	Boolean. Enables using customized feature key display format for Busy Lamp Field key function.	False
<b>TextLine1</b>	Use this text for line 1 when the monitored extension has no calls.	@text1
<b>TextLine2</b>	Use this text for line 2 when the monitored extension has no calls.	@text2
<b>TextLine1InCall</b>	Use this text for line 1 when the monitored extension has one or more calls.	@text1

**Line Key Customization Settings (continued)**

Parameter	Description	Default Setting
<b>TextLine2InCall</b>	Use this text for line 2 when the monitored extension has one or more calls.	@text2

**Speed Dial (VoiceService.1.Phone.CstmLineKey.SpeedDial.)**

<b>Enable</b>	Boolean. Enables using customized feature key display format for Speed Dial key function.	False
<b>TextLine1</b>	Information to show on line 1 of a customized Speed Dial line key.	@text1
<b>TextLine2</b>	Information to show on line 2 of a customized Speed Dial line key.	@text2

**Action URL (VoiceService.1.Phone.CstmLineKey.ActionURL.)**

<b>Enable</b>	Boolean. Enables using a customized feature key display format for <b>ActionURL</b> key function.	False
<b>TextLine1</b>	Information to show on line 1 of a customized <b>ActionURL</b> line key.	@text1
<b>TextLine2</b>	Information to show on line 2 of a customized <b>ActionURL</b> line key.	@text2

**Group n (VoiceService.1.Phone.CstmLineKey.Group.n.), n = 1, 2, 3, 4**

<b>ID</b>	The group ID of a customized line key.	None
<b>TextLine1</b>	Information to show on line 1 of a line key that is in this group.	@text1
<b>TextLine2</b>	Information to show on line 2 of a line key that is in this group.	@text2
<b>Icon</b>	The icon to show for a line key that is in this group.	@icon

## Screen Item Customization Web Page

The Screen Item Customization web page is available only for the VVX 250, VVX 350, and VVX 450 phones.

**Screen Item Customization Settings**

Parameter	Description	Default Setting
<b>Screen Item (VoiceService.1.Phone.Screen.)</b>		
<b>RingItem</b>	The value is a <ScreenItem> XML that customizes the format of the Ringing Calls display on the phone screen.	None
<b>CallItem</b>	The value is a <ScreenItem> XML that customizes the format of the Current Calls display on the phone screen.	None

## Screen Item Customization Settings (continued)

Parameter	Description	Default Setting
<b>CallTransferItem</b>	The value is a <ScreenItem> XML that customizes the format of the Call Transfer display on the phone screen. Unlike the last two items which are items in a table, there is only 1 <b>CallTransferItem</b> .	None
<b>NetDirItem</b>	Customization XML that renders items displayed in a Network Directory.	None
<b>ScreenSaverCustomContents</b>	The value is a <ScreenItem> XML that customizes the contents to be shown on top of the current screen saver picture when the screen saver is being shown.	<pre>&lt;ScreenItem&gt; &lt;time align='center' ypos='60' font='@gfont-bold' size='60' format='@timeFmt' shadow='2,1,0x444444' /&gt; &lt;date align='center' ypos='134' font='@gfont' size='24' format='w &amp;#160;&amp;#160; mn/d/y' shadow='2,1,0x444444' /&gt; &lt;/ScreenItem&gt;</pre>
<b>HomeScreenCustomContents</b>	The value is a <ScreenItem> XML script that defines the size, position, and contents of the rectangular area on the Home Screen. If this value is not specified or invalid, the real time clock displays.	None
<b>TitleBarStyle</b>	Customized style to render the Title Bar (the top Status Bar) of the phone.	<pre>&lt;!-- TitleBarStyle size='12'&gt; &lt;time xpos='2' /&gt;&lt;date xpos='68' /&gt; &lt;text xpos='2' ypos='16'&gt;\$prod.obinumber (\$prod.mac)&lt;/text&gt; &lt;Notifications list='*' width='\$eval(\$sys.hidetabs?@dispW, \$sys.ntabs==4)?@lkpg1,@lkpg1v2)' align='right' xpos='0'&gt; &lt;/Notifications&gt; &lt;/TitleBarStyle --&gt;</pre>

## Screen Item Customization Settings (continued)

Parameter	Description	Default Setting
<b>SoftKeyStyles</b>	Contains a list of as many as 32<style> elements to be used to format a softkey. The enclosing root element must be <SoftKeyStyles>. To use a style, specify a style attribute in the custom softkey (in a softkey set parameter), with the value equal to id attribute of the corresponding <style> element. If no style attribute is specified or the given style isn't found, the <style> with id="default" is used for the softkey, if one is found. Otherwise the internally defined style is used for the softkey.	None
<b>LineKeyStyles</b>		None
<b>MainMenu</b>	The enclosing root element must be <MainMenu>.	See device web page.
<b>MainMenuItemStyles</b>	The enclosing root element must be <MainMenuItemStyles>.	<pre>&lt;MainMenuItemStyles hling=''&gt;   &lt;style id='default'&gt;     &lt;label ypos='50'/&gt;     &lt;label2 ypos='30' xpos='44' width='21' height='18' valign='center' align='center'/&gt;   &lt;/style&gt;   &lt;style id='mwi'&gt;     &lt;label ypos='50'/&gt;     &lt;icon2 ypos='30' xpos='44' width='24' height='24'/&gt;   &lt;/style&gt; &lt;/MainMenuItemStyles&gt;</pre>
<b>ProgressAlertStyles</b>	The enclosing root element must be <ProgressAlertStyles>.	<pre>&lt;!-- ProgressAlertStyles border='2' bordercolor='white' xpos='100' ypos='100' width='160' height='100' opacity='240' bgcolor='red,white,0' bgimg='- ' &gt; &lt;style id='default'&gt; &lt;icon src='http://myserver/icon/alert.png ' width='20' height='20' xpos='2' ypos='2'/&gt; &lt;msg size='11' font='@gfont-bold' width='150' height='70' xpos='2' ypos='24'/&gt; &lt;/style&gt; &lt;/ProgressAlertStyles --&gt;</pre>

# Codec Profiles

The Codec Profile parameters web pages include one page for each codec profile, as shown:

## Codec Profile X Web Page (X = A, B)

The following configuration parameters are available on each page.

### Codec Profile Settings

Parameter	Description	Default Setting
<b>G711U Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.1.), n = 1, 2 for X = A, B respectively</b>		
<b>Codec</b>	Codec name.	PCMU
<b>BitRate</b>	Read-only parameter for bit rate in bits/sec.	64000
<b>Enable</b>	Enables this codec.	True
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in ms.	20
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	3
<b>PayloadType</b>	Dynamic payload type for this codec.	0
<b>G711A Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.2.), n = 1, 2 for X = A, B respectively</b>		
<b>Codec</b>	Codec name.	PCMA
<b>BitRate</b>	Read-only parameter for bit rate in bits/sec.	64000
<b>Enable</b>	Enables this codec.	True
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in ms.	20
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	4
<b>PayloadType</b>	Dynamic payload type for G711-alaw codec.	8
<b>G729 Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.3.), n = 1, 2 for X = A, B respectively</b>		
Note: Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.		
<b>Codec</b>	Codec name.	G729
<b>BitRate</b>	Read-only parameter for bit rate in bits/sec.	8000
<b>Enable</b>	Enables this codec.	True
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in ms.	20
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	5



## Codec Profile Settings (continued)

Parameter	Description	Default Setting
<b>PayloadType</b>	Dynamic payload type for G.729 codec.	18

**iLBC Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.8.), n = 1, 2 for X = A, B respectively**

Note: Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.

<b>Codec</b>	Codec name.	iLBC
<b>BitRate</b>	Bit rate in bits/sec. Choice of: <ul style="list-style-type: none"> <li>13333 bps</li> <li>15200 bps</li> </ul>	13333
<b>Enable</b>	Enables this codec.	False
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in ms.	30
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	10
<b>PayloadType</b>	Dynamic payload type for this codec. Valid range is 96–127.	98

**G722 Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.9.), n = 1, 2 for X = A, B respectively**

<b>Codec</b>	Codec name for this RTP event, as used in SDP.	G722
<b>BitRate</b>	Read-only parameter for bit rate in bits/sec.	64000
<b>Enable</b>	Enables this codec.	True
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in ms.	20
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	2
<b>PayloadType</b>	Dynamic payload type for this codec.	9

**Opus Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.10.), n = 1, 2 for X = A, B respectively**

Note: Opus is not compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC are not published; if you set G.729 and iLBC to the highest priority, Opus is not published.

<b>Codec</b>	Codec name for this RTP event, as used in SDP. Note that some servers may expect this to be lower case <code>opus</code> .	OPUS
<b>BitRate</b>	Variable.	20000
<b>Enable</b>	Enables this codec.	True
<b>SilenceSuppression</b>	Enables silence suppression for this codec.	False
<b>PacketizationPeriod</b>	Packet size in milliseconds.	20
<b>Priority</b>	Priority assigned to this codec (1 is the highest).	2
<b>PayloadType</b>	Dynamic payload type for this codec.	109

**Codec Profile Settings (continued)**

Parameter	Description	Default Setting
<b>UseInbandFEC</b>	Enables in-band forward error correction.	False
<i>Telephone Event (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_TelephoneEvent.), n = 1, 2 for X = A, B respectively</i>		
<b>Codec</b>	Codec name for this RTP event, as used in SDP.	telephone-event
<b>Enable</b>	Enables this codec.	True
<b>PayloadType</b>	Dynamic payload type to be used for RFC2833 telephone (DTMF) events. Valid range is 96–127.	101
<i>Encap RTP (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_EncapRTP), n = 1, 2 for X = A, B respectively</i>		
<b>Codec</b>	Codec name. This codec encapsulates RTP packets during a packet loopback call.	encaprtp
<b>PayloadType</b>	Dynamic payload type for this codec. Valid range is 96–127.	107
<i>Loopback Primer (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_LoopbackPrimer.), n = 1, 2 for TX = A, B respectively</i>		
<b>Codec</b>	Codec name. The phone uses this codec when it acts as a media loopback mirror and before receiving any packets from the loopback source during a media loopback call.	loopbkprimer
<b>PayloadType</b>	Dynamic payload type for this codec. Valid range is 96–127.	108

## Tone Settings

The Tone Profile parameters web pages include one page for each tone profile, as shown:

### ***Tone Profile X Web Page (X = A, B)***

The following configuration parameters are available on each page.

**Tone Profile Settings**

Parameter	Description	Default Setting
<i>Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.1.), n = 1, 2 for X = A, B respectively</i>		
<b>ToneName</b>	Dial Tone.	
<b>TonePattern</b>	Obihai Tone Pattern Script.	350-18, 440-18;20
<i>Ringback Tone (VoiceService.1.VoiceProfile.n.Tone.Description.2.), n = 1, 2 for X = A, B respectively</i>		
<b>ToneName</b>	Ringback Tone.	
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18, 480-18;-1; (2+4)

## Tone Profile Settings (continued)

Parameter	Description	Default Setting
<b><i>Busy Tone (VoiceService.1.VoiceProfile.n.Tone.Description.3.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Busy Tone.	
TonePattern	Obihai Tone Pattern Script.	480-18, 620-18;10; (.5+.5)
<b><i>Reorder Tone (VoiceService.1.VoiceProfile.n.Tone.Description.4.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Reorder tone or Fast busy.	
TonePattern	Obihai Tone Pattern Script.	480-18, 620-18;10; (.25+.25)
<b><i>Confirmation Tone (VoiceService.1.VoiceProfile.n.Tone.Description.5.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Confirmation Tone.	
TonePattern	Obihai Tone Pattern Script.	600-18;1; (.2+.2)
<b><i>Holding Tone (VoiceService.1.VoiceProfile.n.Tone.Description.6.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Holding Tone played when peer holding the call.	
TonePattern	Obihai Tone Pattern Script.	800-18;30; (.1+10)
<b><i>Second Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.7.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Second Dial Tone played when dialing second call in a 3-way call.	
TonePattern	Obihai Tone Pattern Script.	385-18, 484-18;20
<b><i>Stutter Tone (VoiceService.1.VoiceProfile.n.Tone.Description.8.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Stutter Dial Tone.	
TonePattern	Obihai Tone Pattern Script.	350-18, 440-18;20;2 (.1+.1); ( )
<b><i>Howling Tone (VoiceService.1.VoiceProfile.n.Tone.Description.9.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Howling Tone for off-hook warning.	
TonePattern	Obihai Tone Pattern Script.	480+3, 620+3;10; (.125+.125)
<b><i>Prompt Tone (VoiceService.1.VoiceProfile.n.Tone.Description.10.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Prompt Tone to prompt user to enter a number for configuration, such as speed dial.	
TonePattern	Obihai Tone Pattern Script.	480-16;20
<b><i>Call Forwarded Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.11.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Call Forward Dial Tone (Special dial tone to indicate call-forward-all active).	
TonePattern	Obihai Tone Pattern Script.	350-18, 440-18;20; (.2+.2)

## Tone Profile Settings (continued)

Parameter	Description	Default Setting
<b><i>DND Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.20.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	DND Dial Tone (Special dial tone to indicate DND active).	
TonePattern	Obihai Tone Pattern Script.	350-18,440-18;20; (.2+.2)
<b><i>Conference Tone (VoiceService.1.VoiceProfile.n.Tone.Description.12.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Conference Tone (Indicates conference has started).	
TonePattern	Obihai Tone Pattern Script.	350-16;10; (.1+.1, .1+9.7)
<b><i>SIT Tone 1 (VoiceService.1.VoiceProfile.n.Tone.Description.13.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Special Information Tone 1.	
TonePattern	Obihai Tone Pattern Script.	985-16,1428-16,1777-16;20; (1/.380+0,2/.380+0,4/.380+ 0,0/0+4)
<b><i>SIT Tone 2 (VoiceService.1.VoiceProfile.n.Tone.Description.14.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Special Information Tone 2.	
TonePattern	Obihai Tone Pattern Script.	914-16,1371-16,1777-16;20; (1/.274+0,2/.274+0,4/.380+ 0,0/0+4)
<b><i>SIT Tone 3 (VoiceService.1.VoiceProfile.n.Tone.Description.15.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Special Information Tone 3.	
TonePattern	Obihai Tone Pattern Script.	914-16,1371-16,1777-16;20; (1/.380+0,2/.380+0,4/.380+ 0,0/0+4)
<b><i>SIT Tone 4 (VoiceService.1.VoiceProfile.n.Tone.Description.16.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Special Information Tone 4.	
TonePattern	Obihai Tone Pattern Script.	985-16,1371-16,1777-16;20; (1/.380+0,2/.380+0,4/.380+ 0,0/0+4)
<b><i>Outside Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.17.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	Outside Dial Tone.	
TonePattern	Obihai Tone Pattern Script.	385-16;10
<b><i>R-Command Tone (VoiceService.1.VoiceProfile.n.Tone.Description.18.), n = 1, 2 for X = A, B respectively</i></b>		
ToneName	R-Command Tone.	
TonePattern	Obihai Tone Pattern Script.	400-16;5

## Tone Profile Settings (continued)

Parameter	Description	Default Setting
<b><i>Paging Tone (VoiceService.1.VoiceProfile.n.Tone.Description.19.), n = 1, 2 for X = A, B respectively</i></b>		
<b>ToneName</b>	Paging Tone (The short tone played before playing the caller's voice).	
<b>TonePattern</b>	Obihai Tone Pattern Script.	480-16;1; (.2+.2)

## Ring Settings

The Ring Profile parameters web pages include one page for each ring profile, as shown:

### ***Ring Profile X Web Page (X = A, B)***

The following configuration parameters are available on each page.

#### Ring Settings

Parameter	Description	Default Setting
<b><i>Call Waiting Tone 1 (VoiceService.1.VoiceProfile.n.Tone.Description.1.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 1.	Bellcore-dr1
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.25+10)
<b><i>Call Waiting Tone 2 (VoiceService.1.VoiceProfile.n.Tone.Description.2.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 2.	Bellcore-dr2
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.1+.1, .3+.1, .1+10)
<b><i>Call Waiting Tone 3 (VoiceService.1.VoiceProfile.n.Tone.Description.3.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 3.	Bellcore-dr3
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.1+.1, .1+10)
<b><i>Call Waiting Tone 4 (VoiceService.1.VoiceProfile.n.Tone.Description.4.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 4.	Bellcore-dr4
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.1+.1, .1+.1, .1+10)
<b><i>Call Waiting Tone 5 (VoiceService.1.VoiceProfile.n.Tone.Description.5.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 5.	Bellcore-dr5
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.3+.1, .1+.1, .3+10)
<b><i>Call Waiting Tone 6 (VoiceService.1.VoiceProfile.n.Tone.Description.6.), n = 1,2 for X=A,B respectively</i></b>		

## Ring Settings (continued)

Parameter	Description	Default Setting
<b>ToneName</b>	Distinctive Call Waiting Tone 6.	User-dr1
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.1+.1, .3+.2, .3+10)
<b><i>Call Waiting Tone 7 (VoiceService.1.VoiceProfile.n.Tone.Description.7.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 7.	User-dr2
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.3+.1, .3+.1, .1+10)
<b><i>Call Waiting Tone 8 (VoiceService.1.VoiceProfile.n.Tone.Description.8.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 8.	User-dr3
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.3+2)
<b><i>Call Waiting Tone 9 (VoiceService.1.VoiceProfile.n.Tone.Description.9.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 9.	User-dr4
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.3+2)
<b><i>Call Waiting Tone 10 (VoiceService.1.VoiceProfile.n.Tone.Description.10.), n = 1,2 for X=A,B respectively</i></b>		
<b>ToneName</b>	Distinctive Call Waiting Tone 10.	User-dr5
<b>TonePattern</b>	Obihai Tone Pattern Script.	440-18;30; (.3+2)
<b><i>Ring Pattern 1 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.1.), n = 1,2 for X=A,B respectively</i></b>		
<b>RingName</b>	Ring Pattern 1	Bellcore-dr1
<b>RingPattern</b>	OBIHAI Ring Pattern Script.	60; (2+4)
<b><i>Ring Pattern 2 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.2.), n = 1,2 for X=A,B respectively</i></b>		
<b>RingName</b>	Ring Pattern 2	Bellcore-dr2
<b>RingPattern</b>	OBIHAI Ring Pattern Script.	60; (.3+.2, 1+.2, .3+4)
<b><i>Ring Pattern 3 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.3.), n = 1,2 for X=A,B respectively</i></b>		
<b>RingName</b>	Ring Pattern 3	Bellcore-dr3
<b>RingPattern</b>	OBIHAI Ring Pattern Script.	60; (.8+.4, .8+4)
<b><i>Ring Pattern 4 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.4.), n = 1,2 for X=A,B respectively</i></b>		
<b>RingName</b>	Ring Pattern 4	Bellcore-dr4
<b>RingPattern</b>	OBIHAI Ring Pattern Script.	60; (.4+.2, .3+.2, .8+4)
<b><i>Ring Pattern 5 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.5.), n = 1,2 for X=A,B respectively</i></b>		
<b>RingName</b>	Ring Pattern 5	Bellcore-dr5

## Ring Settings (continued)

Parameter	Description	Default Setting
RingPattern	OBIHAI Ring Pattern Script.	60; (.2+.2, .2+.2, .2+.2, 1+4)
<i>Ring Pattern 6 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.6.), n = 1,2 for X=A,B respectively</i>		
RingName	Ring Pattern 6	User-dr1
RingPattern	OBIHAI Ring Pattern Script.	60; (.2+.4, .2+.4, .2+4)
<i>Ring Pattern 7 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.7.), n = 1,2 for X=A,B respectively</i>		
RingName	Ring Pattern 7	User-dr2
RingPattern	OBIHAI Ring Pattern Script.	60; (.4+.2, .4+.2, .4+4)
<i>Ring Pattern 8 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.8.), n = 1,2 for X=A,B respectively</i>		
RingName	Ring Pattern 8	User-dr3
RingPattern	OBIHAI Ring Pattern Script.	60; (.25+9.75)
<i>Ring Pattern 9 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.9.), n = 1,2 for X=A,B respectively</i>		
RingName	Ring Pattern 9	User-dr4
RingPattern	OBIHAI Ring Pattern Script.	60; (.25+9.75)
<i>Ring Pattern 10 (VoiceService.1.VoiceProfile.1.Line.n.Ringer.Description.10.), n = 1,2 for X=A,B respectively</i>		
RingName	Ring Pattern 10	User-dr5
RingPattern	OBIHAI Ring Pattern Script.	60; (.25+9.75)

## Star Code Profiles

The Star Code Profile parameters web pages include one page for each Star Code profile, as shown:

### ***Star Code Profile X Web Page (X = A, B)***

The following configuration parameters are available on each page.

#### Star Code Settings

Parameter	Description	Default Setting
Code1	Redial Star Code.	*07, Redial, call(\$Ldn)
Code2	Call Return Star Code.	*69, Call Return, call(\$Lcn)
Code3	Block Caller ID (Persistent) Star Code.	*81, Block Caller ID, set(\$Bci,1)
Code4	Unblock Caller ID (Persistent) Star Code.	*82, Unblock Caller ID, set(\$Bci,0)

## Star Code Settings (continued)

Parameter	Description	Default Setting
<b>Code5</b>	Block Caller ID Once Star Code.	*67, Block Caller ID Once, set(\$Bci1,1)
<b>Code6</b>	Unblock Caller ID Once Star Code.	*68, Unblock Caller ID Once, set(\$Ubcil,1)
<b>Code7</b>	Call Forward Unconditional Star Code.	*72, C fwd All, coll(\$Cfan), set(\$Cfa,1)
<b>Code8</b>	Disable Call Forward Unconditional Star Code.	*73, Disable C fwd All, set(\$Cfa, 0)
<b>Code9</b>	Call Forward on Busy Star Code.	*60, C fwd Busy, coll(\$Cfbn), set(\$Cfb,1)
<b>Code10</b>	Disable Call Forward on Busy Star Code.	*61, Disable C fwd Busy, set(\$Cfb, 0)
<b>Code11</b>	Call Forward on No Answer Star Code.	*62, C fwd No Ans, coll(\$Cfnn), set(\$Cfn,1)
<b>Code12</b>	Disable Call Forward on No Answer Star Code.	*63, Disable C fwd No Ans, set(\$Cfn,0)
<b>Code13</b>	Block Anonymous Calls Star Code.	*77, Block Anonymous Call, set(\$Bac,1)
<b>Code14</b>	Unblock Anonymous Calls Star Code.	*87, Unblock Anonymous Call, set(\$Bac,0)
<b>Code15</b>	Enable Call Waiting Star Code.	*56, Enable Call Waiting, set(\$Cwa,1)
<b>Code16</b>	Disable Call Waiting Star Code.	*57, Disable Call Waiting, set(\$Cwa,0)
<b>Code17</b>	Do Not Disturb Star Code.	*78, Do Not Disturb, set(\$Dnd,1)
<b>Code18</b>	Disable Do Not Disturb Star Code.	*79, Disable DND, set(\$Dnd,0)
<b>Code19</b>	Repeat Dial Star Code.	*66, Repeat Dial, rpdi(\$Ldn)
<b>Code20</b>	Disable Repeat Dial Star Code.	*86, Disable Repeat Dial, rpdi
<b>Code21</b>	Set Speed Dial Star Code.	*74(x xx), Set Speed Dial, coll(\$Spd[\$Code])
<b>Code22</b>	Check Speed Dial Star Code.	*75(x xx), Check Speed Dial, say(\$Spd[\$Code])
<b>Code23</b>	Loopback Media Star Code.	*03, Loopback Media, set(\$Lbm1,1)
<b>Code24</b>	Loopback RTP Star Code.	*04, Loopback RTP Packet, set(\$Lbp1,1)
<b>Code25</b>	Force G711u Codec Star Code.	*4711, Use G711 Only, set(\$Cdm1,3)



**Star Code Settings (continued)**

Parameter	Description	Default Setting
<b>Code26</b>	Force G729 Codec Star Code.	*4729, Use G729 Only, set(\$Cdm1,4)
<b>Code27</b>	Clear Speed Dial Star Code.	*76([1-9]  [1-9]x), Clear Speed Dial, set(\$Spd[\$Code],)
<b>Code28</b>	unused	None
<b>Code29</b>	Barge In Star Code.	*96, Barge In, set(\$Bar1,1)
<b>Code30</b>	unused	None
<b>Code31</b>	Star Code entry 31.	*4722, Use G722 Only, set(\$Cdm1,512)
<b>Code32</b>	Star Code entry 32.	*4678, Use OPUS Only, set(\$Cdm1,1024)

## User Settings

The User Settings parameters web pages include the following pages:

- [User Preferences Web Page](#)
- [Speed Dials Web Page](#)
- [User Defined Digit Maps Web Page](#)

### ***User Preferences Web Page***

The following configuration parameters are available on this page.

## User Preferences Settings

Parameter	Description	Default
<b>User Preferences Settings</b>		
<b>Language</b>	<p>Sets the language for the on-screen messages. Choice of:</p> <ul style="list-style-type: none"> <li>• English-US</li> <li>• Deutsche (German)</li> <li>• Francais (French)</li> <li>• Español (Spanish)</li> <li>• English-UK</li> <li>• Dansk (Danish)</li> <li>• Suomalainen (Finnish)</li> <li>• Nederlands (Dutch)</li> <li>• Norsk (German)</li> <li>• Svenska (Swedish)</li> <li>• 繁體中文 (not available on VVX 150)</li> <li>• 한국어 (not available on VVX 150)</li> <li>• 日本語 (not available on VVX 150)</li> <li>• 简体中文 (not available on VVX 150)</li> <li>• Custom</li> </ul> <p>Except for <code>Custom</code>, each built-in language corresponds to a dictionary file in that language. When the value is set to <code>Custom</code>, the corresponding customer dictionary file (cloud or local) is used instead. The file will download once per reboot because it is not cached. If that custom dictionary file is not found or has an error, the phone reverts to use <code>English-US</code> as the language.</p>	English-US
<b>CustomLanguage</b>	Specifies the language corresponding to a custom dictionary file installed on the phone.	None
<b>TimeFormat</b>	<p>Chooses the time display format on the title bar of the phone screen. Choice of:</p> <ul style="list-style-type: none"> <li>• default</li> <li>• 1:35 pm</li> <li>• 13:35</li> <li>• 1:35</li> </ul>	default
<b>DateFormat</b>	<p>Chooses the date display format on the title bar of the phone screen. Choice of:</p> <ul style="list-style-type: none"> <li>• default</li> <li>• 1/31/2014</li> <li>• 1/31/14</li> <li>• 1/31</li> <li>• 31/1/2014</li> <li>• 31/1/14</li> <li>• 31/1</li> <li>• Jan 31, 2014</li> </ul>	default

User Preferences Settings (continued)

Parameter	Description	Default
<b>AutoHomePage</b>	Chooses the delay to automatically return the screen to the Home (idle) screen. Choice of: <ul style="list-style-type: none"> <li>disabled</li> <li>30 seconds</li> <li>1 minute</li> <li>10 minutes</li> <li>1 hour</li> <li>4 hours</li> </ul>	1 minute
<b>Skin</b>	Choose the skin (or style and color scheme) of the UI. Choice of: <ul style="list-style-type: none"> <li>Tomáš</li> <li>Ulrik</li> <li>Tomáš-Wide</li> <li>Ulrik-Wide</li> </ul>	Tomáš-Wide
<b>BackgroundPicture</b>	A URL (HTTP or HTTPS) to a picture file or the path name of an internally stored picture.	Default
<b>SideCarBackground Picture</b>	Background picture on the side car(s). If two side cars are installed, then the picture is the same. The optimum image resolution is 480H x 800V. If no picture is specified, then the side car background picture(s) are the same as the phone background picture.	None
<b>DefaultRingtone</b>	A URL (HTTP or HTTPS) to a ring tone file (.wav or MP3 format) or the path name of an internally stored ring tone file.	/data/ringtones/Office A.wav
<b>CallParkRingtone</b>	A URL (HTTP or HTTPS) to a ring tone file (.wav or MP3 format) or the name of an internally short ring: ding or blop. It can be followed by two optional comma-separated parameter={value} pairs: <ul style="list-style-type: none"> <li>duration={value}. {value} is total duration to play in seconds. Must be &gt; 0. 0 means play forever. Default is 60.</li> <li>interval={value}. {value} is interval between ring bursts in seconds. Must be &gt; 0. Default is 10.</li> </ul>	ding,interval=10,duration=60
<b>DefaultFont</b>	The default font used to render text on the phone screen.	None
<b>PopUpMoreSoftKey Options</b>	If enabled, shows more softkey options in a popup window when user presses the <b>More</b> softkey when there are more than four options. Otherwise, show softkey options in a circular list when user presses the <b>More</b> softkey.	True

## User Preferences Settings (continued)

Parameter	Description	Default
<b>PackCallsOnDisplay</b>	This option is applicable when one or more Call Keys (feature key that is assigned the Call Appearance function) have the <b>MaxCalls</b> parameter set to greater than 1. If the option is disabled, calls controlled by different Call Keys are shown on separate screens as you scroll through the calls. Otherwise, calls controlled by different Call Keys can be shown on the same screen (packed together).	True
<b>LineKeyTabs</b>	This option controls whether to show multiple tabs of (virtual) Line Keys on the screen.	True
<b>LineKeyMaxTabs</b>	Must be either 2 or 4. Applies only if <b>LineKeyTabs</b> = true.	4
<b>LeftLineKeyTabs</b>	This option controls whether to show multiple tabs of (virtual) Left Line Keys on the screen.	True
<b>LeftLineKeyMaxTabs</b>	Must be either 2 or 4. Applies only if <b>LineKeyTabs</b> = true.	4
<b>DimScreenDelay</b>	If greater than 0, this is the delay in seconds before dimming the screen when there are no activities. Dimming is done by setting the LCD brightness to the minimum value of 0. Any key press or new call restores the normal LCD brightness. This operation is independent of Screen Saver settings.	0
<b>ScreenSaver</b>	Turn the screen saver features on or off.	False
<b>ScreenSaverDelay</b>	Number of seconds of inactivity before starting screen saver.	10
<b>ScreenSaverType</b>	Choose the type of screen saver to use: <ul style="list-style-type: none"> <li>Slide Show</li> <li>Slide Show (User Pictures Only)</li> <li>Slide Show (Specified URL Only)</li> <li>Turn Off Display</li> </ul>	Slide Show
<b>ScreenSaverPicture URL</b>	When <b>ScreenSaverType</b> is Slide Show (Specified URL Only), this parameter specifies the URL to get the pictures for the slide show. It can be an http:// or https:// URL to get the screen save picture(s), or the absolute path of an internal folder to get the screen saver pictures. If an http URL is used, the server may indicate 'no-cache' in the Cache-Control header of the http response to the phone to force the phone to download a fresh picture from the server on every screen picture refresh.  You can use \$count macro in the http/https URL which is expanded into the number of refreshes when the URL is executed. It starts with 1, 2, 3, ..., max-count and the recycles. The max-count value can be specified as a comma-separated attribute in this URL also. The default is 0x7fffffff. For example:  http://xyz.com/pic-\$count.png,max-count=4 http://xyz.com/pic-{\$count}.png,max-count=15	0x7fffffff

## User Preferences Settings (continued)

Parameter	Description	Default
<b>SlideShowInterval</b>	The interval to refresh each picture in a slide show when screen saver is running.	5
<b>ScreenSaverLock</b>	Control whether dismissing the screen saver requires a passcode.	False
<b>ScreenSaverShowCustomContents</b>	Show the custom contents defined in the parameter <b>ScreenSaverCustomContents</b> (under the Screen Item Customization group), super-imposed on the current screen saver picture.	True
<b>UnlockPasscode</b>	Passcode to unlock the screen saver.	123456
<b>LockIncomingCall</b>	When the screen saver is activated and protected by a passcode, this option controls whether incoming calls to the phone can be answered without unlocking the screen.	False
<b>Brightness</b>	Screen brightness level.	6
<b>PreferredAudioDevice</b>	The preferred audio device (also known as an audio terminal) to use when answering a call by pressing a key (for example, a softkey or line key) that is not the speakerphone key or the headset key. Choice of: <ul style="list-style-type: none"> <li>• Speaker</li> <li>• Headset</li> </ul>	Speaker
<b>PreferredHeadsetDevice</b>	The preferred headset to use (when more than one are available) when answering a call by pressing the headset key. Choice of: <ul style="list-style-type: none"> <li>• RJ9 Headset</li> <li>• USB Headset</li> </ul>	RJ9 Headset
<b>USBHeadsetSync</b>	Enables signaling to the USB headset device when the headset or mute key on the phone is pressed.	True
<b>EhsEnable</b>	Enables using an EHS Headset. Choice of: <ul style="list-style-type: none"> <li>• Disabled</li> <li>• Plantronics</li> <li>• Jabra</li> <li>• Sennheiser</li> </ul>	Disabled
<b>RingerVolume</b>	The ringer audio volume when ringing.	9
<b>SpeakerVolume</b>	The speakerphone audio playback volume.	10
<b>HandsetVolume</b>	The handset audio playback volume.	11
<b>HandsetGain</b>	The gain to apply to the handset audio input.	5
<b>HeadsetVolume</b>	The audio playback level on the RJ9 Headset.	10
<b>HeadsetGain</b>	The audio gain to apply to the RJ9 Headset input.	5

## User Preferences Settings (continued)

Parameter	Description	Default
<b>HeadsetUSBVolume</b>	The audio playback level on the USB headset.	9
<b>HeadsetUSBGain</b>	The audio gain to apply to the USB Headset input.	5
<b>DuplexBalance</b>	Set to 1 for improved early-start full duplex.	0
<b>HandsetENS</b>	Enables Enhanced Noise Suppression (Acoustic Fence) for the handset.	False
<b>HeadsetENS</b>	Enables Enhanced Noise Suppression (Acoustic Fence) for the headset.	False
<b>HeadsetAEC</b>	Enables Acoustic Echo Cancellation for the headset.	False
<b>HandsetLineAGC</b>	Enables line Automatic Gain Control for the handset.	False
<b>HeadsetLineAGC</b>	Enables line Automatic Gain Control for the headset.	False
<b>TxPacketFilter</b>	Defines high-pass filters that can be applied to transmitted narrow-band signals to improve intelligibility. Choice of: <ul style="list-style-type: none"> <li>Disabled</li> <li>300Hz high-pass</li> <li>300Hz high-pass with pre-emphasis (best for G.729)</li> </ul>	Disabled
<b>RxPacketFilter</b>	Defines high-pass filters that can be applied to received narrow-band signals to improve intelligibility. Choice of: <ul style="list-style-type: none"> <li>Disabled</li> <li>300Hz high-pass</li> <li>300Hz high-pass with pre-emphasis (best for G.729)</li> </ul>	Disabled
<b>IncomingPageAlertTone</b>	Choose whether to play a beep when the phone answers an incoming Intercom/Page call. Choice of: <ul style="list-style-type: none"> <li>none</li> <li>beep</li> </ul>	beep
<b>UseEnhancedDialer</b>	Enables enhanced dialer where a list of recently dialed number that partially match the currently entered digits is displayed below the dialer input box. You can highlight an entry in the list to make the call to.	True
<b>DefaultLineKeyTab</b>	The line key page (on the right side of the phone) to fall back on after some idle time.	1
<b>DefaultLeftLineKeyTab</b>	The line key page on the left side of the phone to fall back on after some idle time.	1
<b>ToggleCallsApp</b>	Hide or show the <b>Calls</b> app when pressing the <b>Home</b> Key when there is one or more calls.	False
<b>CustomMainMenu</b>	Use customized Main menu.	False

**User Preferences Settings (continued)**

<b>Parameter</b>	<b>Description</b>	<b>Default</b>
<b>AutoAnswerAutoMute</b>	Enables auto-mute when the phone auto-answers an incoming Intercom/page call.	False
<b>AutoAnswerOffWhenBusy</b>	Automatically turns off auto-answer intercom calls when the phone is busy (if you are making a call or talking on a call).	False
<b>ShowSharedCalls</b>	Enables showing calls on a shared line that are active on other phones. Choice of: <ul style="list-style-type: none"> <li>Always</li> <li>Never</li> <li>User Select</li> </ul>	Always
<b>Phone Book Fields</b>		
<b>Address</b>	Enables the Address field.	False
<b>Company</b>	Enables the Company field.	False
<b>Group</b>	Enables the Group field.	False
<b>Email</b>	Enables the Email field.	True
<b>FaxNumber</b>	Enables the Fax Number field.	True
<b>FirstName</b>	Enables the First Name field.	False
<b>HomeNumber</b>	Enables the Home Number field.	False
<b>LastName</b>	Enables the Last Name field.	False
<b>MobileNumber</b>	Enables the Mobile Number field.	True
<b>Name</b>	Enables the Name field.	True
<b>PrimaryNumber</b>	Enables the Primary Number field.	True
<b>OfficeNumber</b>	Enables the Work Number field.	True
<b>Picture</b>	Enables the Picture field.	True
<b>Ringtone</b>	Enables the Ringtone field.	True
<b>Service</b>	Enables the Service field.	True
<b>Phone Book Settings</b>		
<b>Name</b>	The label to display for this item on the phone's main menu screen.	Contacts
<b>ActionURL</b>	The (HTTP or HTTPS) URL to execute when you select the Phone Book item from the phone's menu, if one is specified. Otherwise, the phone launches the built-in Phone Book application.	None

**User Preferences Settings (continued)**

Parameter	Description	Default
<b>Groups</b>	A comma-separated list of group names to group contacts within the built-in phone book.	Co-Workers, Customers, Family, Friends
<b>Filter</b>	A comma-separated list of groups to filter. A blank entry implies no filtering.	None
<b>FilterEnable</b>	Enables filtering by group.	False
<b>SaveFromCallHistory</b>	Enables softkey option in call history to save an item into the phone book.	True
<b>ExportURL</b>	A target URL for exporting a phone book as a PolyPhoneBook XML.	None
<b>ImportURL</b>	A target URL for importing a phone book as a PolyPhoneBook XML.	None

## Speed Dials Web Page

The following configuration parameters are available on this page.

**Speed DialsSettings**

Parameter	Description	Default Setting
<b>N</b> (Repeated for N = 1 – 99)	Speed Dial N	None

## User Defined Digit Maps Web Page

For an explanation of this feature, please refer to the [User-Defined Digit Maps](#) section in the [Call Controls](#) chapter.

**User Defined Digit Map Settings**

Parameter	Description	Default Setting
<b>User Defined Digit Map1</b>		
<b>Label</b>	A 2- to 16-character long label to reference this digit map in other digit maps and call routing rules. It must be alphanumeric, not containing any spaces, and different from other user-defined or built-in digit map labels.	ipd



**User Defined Digit Map Settings (continued) (continued)**

Parameter	Description	Default Setting
<b>DigitMap</b>	A valid digit map.	(xx.<*:@>xx?x?<*:.>xx?x? <*:.>xx?x?<*:.>xx?x? xx. <*:@>xx?x?<*:.>xx?x?<*:. >xx?x?<*:.>xx?x?<*::>xx? x?x?x?)  Note: This default value supports IPv4 dialing.
<b>User Defined Digit MapN (N = 2 – 10)</b>		
<b>Label</b>	A 2- to 16-character long label to reference this digit map in other digit maps and call routing rules. It must be alphanumeric, not contain any spaces, and be different from other user-defined or built-in digit map labels.	None
<b>DigitMap</b>	A valid digit map.	None

# Troubleshooting

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Refer to the following topics to help you diagnose and fix issues with your system.

## System Logs

You can capture system logs to help troubleshoot your OBi Edition VVX business IP phones.

### *Syslog*

Your phone can be set up to send out syslog messages for troubleshooting. To capture syslog messages from your phone, you need to run a syslog server application at an IPv4 address that is reachable from the phone. On the phone side, the IP address and listening port of the syslog server are configured in **Device Admin – Syslog::Server** and **Device Admin – Syslog::Port**. The default port value is 514.

To include detailed SIP messages on an SP service in the syslog, use the **SPn Service::X\_SipDebugOption** parameter with one of the options: **Disable**, **Log All Messages**, or **Log All Except REGISTER Messages**. **SPn Service::X\_SipDebugExclusion** is a list of SIP methods (request and responses) to exclude from the log. For troubleshooting a call flow, methods such as **OPTIONS** that are used for keep-alive purpose may be excluded from the log in most cases.

## Factory Resetting Your Phone

You can reset all configuration parameters to factory default values or to the customized non-generic default values.

### *Factory Reset Your Phone in the Local Interface*

You can reset all phone settings to factory default values from the phone's local interface.

**To factory reset your phone in the local interface:**

- 1 Go to **Settings**.
- 2 Press the **Factory Reset** softkey.

### *Factory Reset Your Phone in the System Web Interface*

You can reset the phone user data and voice configuration settings to factory default values in the system web interface.

**To factory reset your phone in the system web interface:**

- 1 In the system web interface, go to **System Management > Device Update > Reset Configuration**.
- 2 Select **User Data, Voice Configuration**, or both.
- 3 Select **Reset**.

## Capture Your Device's Current Screen

From the system web interface, you can take a snapshot of the current phone screen and save it as a .bmp file on your host computer.

**To capture your device's screen:**

- 1 Enable the *Device Admin – Web Server::LCDScreenShot* option.
- 2 Open the **System Management – Device Update** web page of the phone.
- 3 Press **Snap**.