ADMINISTRATOR GUIDE

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VVX D230 DECT IP Phone

GETTING HELP

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

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Contents

Before You Begin

This guide describes how to administer, configure, and provision VVX D230 devices.

Audience, Purpose, and Required Skills

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

- Current telecommunications practices, protocols, and principles
- Telecommunication basics, audio teleconferencing, and voice or data equipment
- Open SIP 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 provides support documentation for active products, services, and solutions. The documentation displays in responsive HTML5 format so that you can easily access and view installation, configuration, or administration content from any online device
- The Polycom Community provides access to the latest developer and support information. Create an account to access Poly support personnel and participate in developer and support forums. You can find the latest information on hardware, software, and partner solutions topics, share ideas, and solve problems with your colleagues
- The Polycom Partner Network are industry leaders who natively integrate the Poly standards-based RealPresence Platform with their 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 get the most out of your investment through the benefits of collaboration.

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 native web portal or on OBiTALK.com.

Canonical Fashion

This example shows the format of the canonical fashion.

- ***Parameter Group Name***::**ParameterName** = `Parameter Value {replace-with-actual-value}`

The ***Parameter Group Name*** is the heading of the parameter group on the left side panel of the device local configuration or OBiTALK Configuration web page. This string may contain spaces. When a group heading has more than one level, each level is separated 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. ***Parameter Group Name*** and **ParameterName** are separated by two colons (::), as shown in the first example above.

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.myserviceprovider.com`
- **ProxyServerPort** = `5082`

Literal Fashion

These examples show the format of the literal fashion. The literal fashion is used when provisioning.

- ***ParameterGroupName***.**ParameterName**.`Parameter Value {replace-with-actual-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 is terminated with a period, as shown. More than one ***ParameterGroupName.*** may be used. The ***ParameterGroupName.*** is case-sensitive.

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

The `Parameter Value` is the literal value to assign to the named parameter and may contain spaces. The `Parameter Value` is not 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 need to exactly match the text string for ***ParameterGroupName***.**ParameterName**.`Parameter Value`, but text formatting such as bold face is not required and is removed when your script or app is processed.

Boolean Values

You can identify parameters that take a Boolean value on your phone's configuration web pages by a check box next to the parameter name. Throughout the document, we refer to a Boolean value as "enable or disable" or "yes or no", but the only valid Boolean parameter values to use in a phone configuration file is either `true/false` or `True/False` (case-sensitive). This is equivalent to selecting or clearing the check box on the configuration web pages.

Multiple Choice Values

You must provision parameters that take one of several valid options from a drop-down list on the device message 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 web page 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 space 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_STUNServerPort**, use just a single space and no extra space before and after the value.

For example: **X_STUNServerPort** `= UDP listen port of the STUN Server`

Getting Started

Built with a high-performance system-on-a-chip platform to ensure high-quality voice conversations, the VVX D230 is a dedicated system targeted at applications for VoIP services. VVX D230 devices have high availability and reliability because they're always on to make or receive calls.

Product Overview

VVX D230 devices support Polycom HD Voice technology. You can manage the handset's local interface and network interaction on VVXD230 devices directly from OBiTALK.com or through the system web interface.

VVX D230 devices implement the following features and functionalities:

- Aggregation and bridging of eight SIP accounts
- Recursive digit maps and associated call routing (outbound and inbound)
- Support for all standard SIP-based IP PBX and ITSPs/VSPs
- Cloud management enabled via OBiTALK.com with both a user portal and an ITSP partner portal
- OBiTALK managed VoIP network for endpoint devices and applications
- High-quality voice encoding using G.711, G.722, G.726, G.729, iLBC, and Opus codecs

LED Status Indicators

VVX D230 devices contain one LED on the base station and one on the handset.

The following table describes the device's base station LED behavior and status information.

VVX D230 Base Station LED Status Indicators

Indicator	Status
Solid red	Powering On On Idle
Blinking red	Waiting for network availability Locating a handset Registering a handset

Powering the Device On and Off

The VVX D230 device turns on when you plug it into a power source. Connect the power adapter to the base station if Power over Ethernet (POE) isn't available.

If you use the power adapter, use only the 5V adapter supplied with the original packaging to power the device. Using any adapter other than the one supplied voids the warranty and may cause the unit to malfunction.

Ethernet and PC Connections

By default, when you connect the device to an internet router or Ethernet switch, the device requests an IP address, a DNS, and an internet (LAN) gateway IP address via DHCP.

You can also connect your PC to the base station using an Ethernet cable. If you have one Ethernet port that normally connects to the PC, you lose it when used for the D230 device. To get your internet connectivity back to the PC, connect the PC to the PC port of the D230 device.

Configure the Primary Line

The primary line is the default service used to make calls when no explicit access code prefix is entered. You can select a service as the primary line.

The following list summarizes the choices available for the primary line:

- SP1–8 Service: Can be a SIP-based service
- OBiTALK Service: Peer-to-peer service provided free with all device models

Configuration and Management

VVX D230 provides the following options to configure and manage your device: Interactive voice response (IVR) system

- Device local interface
- OBITALK.com system web interface

Configure Your Device Using the IVR System

The VVX D230 device uses the IVR system for both its configuration and normal functionality. Access the IVR system to receive verbal prompts and information from the device (such as the device IP address).

Note the following information regarding the IVR system:

- If a setting change requires a reboot, the system reboots automatically when you quit the IVR system.
- You can access the next menu of the IVR system or invoke a command without waiting for the previous announcement to end.

Configure Basic Device Settings

Use the IVR system's main menu to configure your device's basic settings or to access additional configuration menus.

To configure basic device settings:

1. Dial ******* from the handset.
2. Enter the number for the configuration menu you want to access.

Menu Selection	Setting	Description
1	**Basic Network Status**	Device IP address and DHCP status.
2	**Advanced Network Status**	Information on the primary and back-up DNS server and primary and back-up NTP server.
3	**Set DHCP**	Current DHCP value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value.

Menu Selection	Setting	Description
4	**Set IP Address**	Current IP address. **Note:** If you enter a new value (static IP address), DHCP is disabled. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value.
5	**Set Password**	Current IVR password. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value.
6	**Software Update** The device plays one of the following messages: • **Software update available. Press 1 to update software.** • **Software update not available.**	If an update is available, press **1** to update the software. 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. Make sure the power and network cable stay connected to the unit until the process is complete.
8	**Restore Factory Default**	Restores the device to factory default settings. • Press **1** to confirm the factory restore. • Press **#** to return to the main configuration menu. • Press **# #** to exit the IVR system.
9	**Reboot**	Reboots the device. • Press **1** to confirm device reboot. • Press **#** to return to the main configuration menu. • Press **# #** or hang up to exit the IVR system.
0	**Additional Options**	Access other configuration settings for your handset.

Configure System Settings

You can configure system options through the system settings submenu. However, the device doesn't announce the available settings in the submenu.

To configure system settings:

1. Dial *****0** from the handset.
2. Enter the number for the configuration menu you want to access, followed by the **#** key.

Menu Selection	Setting	Description
1	Firmware Version	Current firmware version. • Press **0** to repeat the information. • Press **#** to enter another configuration menu selection.
2	IVR Password	Current IVR password. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
3	Debug Level	Current debug level. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
4	Syslog Server IP Address	Current syslog server IP address. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
5	Syslog Server Port	Current syslog server port value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value of **514**. • Press **#** to enter another configuration menu selection.

Configure Network Settings

You can configure network options through the network settings submenu. However, the device doesn’t announce the available settings in the submenu.

To configure network settings:

1 Dial *****0** from the handset.

2 Enter the number for the configuration menu you want to access, followed by the **#** key.

Menu Selection	Setting	Description
20	DHCP Configuration	Current DHCP configuration value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
21	IP Address	Current IP address. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
22	Default Gateway	Current default internet gateway. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
23	Subnet Mask	Current subnet mask. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
24	DNS Server (Primary)	Current primary DNS server. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

Menu Selection	Setting	Description
25	LLDP Discovery (Enable/Disable)	Current LLDP Discovery configuration value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. Press **#** to enter another configuration menu selection.
26	NTP Server (Primary)	Current primary NTP server. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

Configure SIP Service Provider Settings

You can configure SIP service provider options through the SIP service provider settings submenu. However, the device doesn't announce the available settings in the submenu.

To configure SIP service provider settings:

1. Dial ***0 from the handset.
2. Enter the number for the configuration menu you want to access, followed by the **#** key.

SP1 Configuration Settings

Menu Selection	Setting	Description
100	Enable Service Provider One (SP1)	Current SP1 value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
101	Registration State of SP1	SP1 registration state. • Press **0** to repeat the information. • Press **#** to enter another configuration menu selection.
102	SP1 User ID	SP1 user ID value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

SP1 Configuration Settings

Menu Selection	Setting	Description
167	SP1 Block Caller ID Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
168	SP1 Block Anonymous Call Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
172	SP1 Call Forward ALL – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
173	SP1 Call Forward ALL Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
174	SP1 Call Forward on Busy – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
175	SP1 Call Forward on Busy Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
176	SP1 Call Forward on No Answer – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
177	SP1 Call Forward on No Answer Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

SP2 Configuration Settings

Menu Selection	Setting	Description
200	Enable Service Provider Two SP2.	Current SP2 value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
201	Registration State of SP2	SP2 registration state. • Press **0** to repeat the information. • Press **#** to enter another configuration menu selection.
202	SP2 User ID	SP2 user ID value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
267	SP2 Block Caller ID Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
268	SP2 Block Anonymous Call Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
272	SP2 Call Forward ALL – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
273	SP2 Call Forward ALL Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

SP2 Configuration Settings

Menu Selection	Setting	Description
274	SP2 Call Forward on Busy – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
275	SP2 Call Forward on Busy Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
276	SP2 Call Forward on No Answer – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
277	SP2 Call Forward on No Answer Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

Configure OBiTalk Settings

You can configure OBitalk options through the OBitalk settings submenu. However, the device doesn't announce the available settings in the submenu.

To configure OBitalk settings:

1. Dial ***0** from the handset.
2. Enter the number for the configuration menu you want to access, followed by the **#** key.

Menu Selection	Setting	Description
900	Enable OBiTALK Service	Current OBiTALK service value. • Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
901	Registration State of OBiTALK	OBiTALK registration state. • Press **0** to repeat the information. • Press **#** to enter another configuration menu selection.
967	OBiTALK Block Caller ID Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
968	OBiTALK Block Anonymous Call Enable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
972	OBiTALK Call Forward ALL – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
973	OBiTALK Call Forward ALL Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
974	OBiTALK Call Forward on Busy – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
975	OBiTALK Call Forward on Busy Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

Menu Selection	Setting	Description
976	OBiTALK Call Forward on No Answer – Enable / Disable	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.
977	OBiTALK Call Forward on No Answer Number	• Press **0** to repeat the information. • Press **1** to enter a new value. • Press **2** to set the default value. • Press **#** to enter another configuration menu selection.

Remote Provisioning

Remote Provisioning is the process by which VVX D230 devices download a configuration file from a server, located in the cloud or in the same enterprise.

The configuration file contains all the necessary parameter values for the device to function normally. It also can 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 Poly products.

It supports two configuration file formats: A full XML format with the XML tags in full text and a short XML format with the XML tags substituted with a single letter abbreviation. The XML structure and parameter naming convention closely follows TR-069/TR-104.

Similar to the way parameters are grouped under different device configuration web pages, parameters are grouped into a number of configuration objects for remote provisioning.

The corresponding configuration object in a handset configuration XML file is:

VoiceService.1.VoiceProfile.1.Line.1.SIP.

as shown:

```
<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 you must not omit the dot (.) at the end of the object name is part of the name in the XML file. You must use the correct object name to create a valid configuration file for the handset. You can find the object name corresponding to each configuration web page/section listed in the Parameter Reference section at the end of this guide.

Data model is a collective list of configuration parameters, syntaxes, and valid values for a specific device. You can find the most up-to-date data model for the device online at the following URL:

https://www1.obitalk.com/Downloads/dev/datamodel/VVXD230.xml

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, customize the ***ITSP Provisioning*****::ConfigURL** parameter, which tells the handset where to download a configuration file. With this parameter configured, the first time a new handset 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.

Configure Your Device Using the System Web Interface

The device has an integrated system web interface that you can access from a PC or similar device using a browser. Although all popular browsers are tested for compatibility with the device management web server, there may be inconsistencies that arise from time to time. Contact obi.spsupport@polycom.com if you have any questions about the system web interface and how it appears in your browser window.

You must individually submit every configuration page after you make changes on the page. Otherwise those changes are discarded once you go to another page. Most changes require a reboot of the unit (by clicking **Reboot)** to take effect. However, you may reboot the unit just once after you have made and submitted all the necessary changes on all the pages.

Access the System Web Interface

Use the system web interface to configure and make changes to your device.

To access the system web interface:

1. From the handset, dial *** * *** to access the device Config Attendant.
2. Choose **1** to hear the IP address of the device read back to you. Write this down.
3. Enter the device IP address in a local PC web browser.
4. When prompted, enter `admin` for user name and `admin` for password.

   If there is a change in default user name and passwords, log in with the updated credentials.

Star Codes

Star codes are short sequences of digits where each sequence serves as a command to the device to perform certain operation. Each sequence usually starts with the * key followed by a 2-digit code (such as *69).

You can use star codes to set the value of one or more configuration parameters. The device allows you to issue star code from the handset only. Every star code and its operation are defined with a short star code script parameter. The set of star codes that can be dialed from the handset is collectively referred to as a *star code profile*.

Controlling Calls Using Star Codes

The device has two star code profiles available in its configuration, known as Star Code Profile A and Star Code Profile B. Each profile has 30 star code script parameters, known as Code1 to Code30. You can select which star code profile to use by setting **Handset::StarCodeProfile** to `A` or `B`, or `None` if star codes aren't used.

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

Star Codes

Your device has the following star codes preprogrammed.

Preprogrammed Star Codes

Code	Description
***03**	Request peer device to loop back media in the next outbound call
***04**	Request peer device to loop back RTP packets in the next outbound call
***05**	Tell device to periodically redial the last called number until the called party rings or answers
***06**	Cancel the last repeat dial request
***07**	Redial
***56**	Enable Call Waiting
***57**	Disable Call Waiting
***60**	Call Forward on Busy (Enter Number + #)

Preprogrammed Star Codes (continued)

Code	Description
***61**	Disable Call Forward in Busy
***62**	Call Forward on No Answer (Enter Number + #)
***63**	Disable Call Forward No Answer
***66**	Repeat Dial
***67**	Block Caller ID (One Time)
***68**	Unblock Caller ID (One Time)
***69**	Call Return
***72**	Call Forward All (Enter Number + #)
***73**	Disable Call Forward All
***77**	Block Anonymous Calls
***78**	Do Not Disturb – Turn On
***79**	Do Not Disturb – Disable
***81**	Block Caller ID (Persistent Mode)
***82**	Unblock Caller ID (Persistent Mode)
***86**	Disable Repeat Dial
***87**	Unblock Anonymous Calls
***4711**	Use G711 Only on the next outbound call
***4729**	Use G729 Only on the next outbound call

Status Pages

The parameters status pages show read-only values for certain parameters on your device.

System Status

The System Status page is divided into several sections:

- WAN Status
- Product Information
- SP*n* Service Status (*n* = 1–8)
- OBiTALK Service Status

WAN Status

The status of the WAN (Ethernet) interface includes information like the assigned IP address, default gateway, and subnet mask.

Product Information

The product information status shows some basic product information, as well as the system up-time with the last reboot reason code in parentheses. The reboot reason codes are defined as follows.

Reboot Reason Codes

Reason Code	Description
0	Reboot on power cycle.
1	Operating system reboot.
2	Forward upgrade.
3	Reboot after new profile invoked.
4	Reboot after parameter value change or firmware has changed and invoked via device web page.
5	Reboot after factory reset using the device hardware PIN.
6	New profile invoked AND profile URL changed.
7	Reboot from SIP Notify (Reserved).
8	Reboot from telephone port (IVR).

Reboot Reason Codes

Reason Code	Description
9	Reboot from webpage — no change in parameter values or firmware.
10	Reboot during OBiTALK signup.
12	Reboot after DHCP server offers IP, GW-IP, and/or netmask different from what the device is currently using.
13	Reboot on data networking link re-establishment.
18	Reboot on WAN IP address change.
19	Reboot on LAN IP address change.

SPn Service Status (n = 1–8)

The SP*n* service status values indicate the current state of the service with regard to its configuration (or not) and if configured its registration status. If there are problems with the registration or authentication of the device with a prescribed service, the **SIP 4*xx*** error message displays here. Youcan use this information for troubleshooting issues with SIP-based services.

OBiTALK Service Status

The status of the OBiTALK Service includes the following values:

- **Status**

  Possible values are:

  - Normal (User Mode): The service is functioning normally.
  - Backing Off: The service is currently down, and the device is taking a short pause before retrying the connection.
- **CallState**

  Possible values are:

  - *N* Active Calls, where *N* = 0, 1, .., as many as the maximum number of calls allowed in the configuration.

Call Status

The Call Status page shows a number of running call statistics and state parameters for each active call currently in progress.

Call History

The Call History page shows the last 200 calls made with the device. Detailed call information is available, including what terminals were involved, the name (if available) of the Peer endpoints making the call, and the direction / path the call took. The Call History page also captures what time various events took place.

The Call History can be saved at any time by clicking on the “Save All” button. The Call History can be saved as an XML formatted file called `callhistory.xml`.

SP Service Stats

Statistics relevant to SP*n* can be found on the SP*n* Service Stats page (where *n* = 1–8).

See the Parameter Reference for information on the parameters displayed on these pages.

Device Settings

You can control how handsets dial calls, speed dial numbers, and user-defined digit maps. You can also control device codec features, handset tones, and ring tones.

Codec Profile Features

Two codec profiles are available on the devices, selectable per trunk (SP*n* and OBiTALK).

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 this profile. Each of the SP and OBi services can be assigned 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 more information on codec profile parameters, see the Codec Profile X Web Page (X = A, B) table in the Parameter Reference section.

Tone Patterns

Your device enables you to create customized tone patterns.

Tone Profile A default settings are set for North American telephone standards. Tone Profile B default settings are set for Australian telephone standards. Tone profiles for other countries are available for download from the OBiTALK forum.

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 A & B Parameter Guide table in the Parameter Reference section.

For more information on call waiting parameters, see the Tone Profile A & B Parameter Guide table in the Parameter Reference section.

User Settings

Use the systemweb interface to configure user speed dial numbers and user-defined digit maps.

Speed Dial Numbers

D230 handset 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 dial values can be set using the system web interface or remote provisioning. The value can 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=1-9)`, `BT1, BT2`, or `PP`. For example, `PP(ob200112233)`, `SP2(14089991123)`, `BT2(4280913)`.

If trunk information isn't specified in the speed dial entry, the device applies **DigitMap** and **OutboundCallRoute** when making the call. Otherwise, neither **DigitMap** nor **OutboundCallRoute** is applied.

User-Defined Digit Maps

See the User-Defined Digit Maps section in the Digit Map Configuration section for more information on this feature.

For more information on user-defined digit map parameters, see the User-Defined Digit Maps Parameter Guide table in the Parameter Reference section.

Conference Calls

A conference call is a conversation involving two or more remote parties. To start a conference, there must be at least two calls and with at least one of them in the **Connected** state and the other in the **Holding** state.

Your phone supports two methods to conference multiple parties:

- Local mixing/bridging
- External conference bridge

Local Mixing or Bridging

After starting three-way conference calls, you can see the two remote parties in the **Connected** state.

The OPUS codec doesn't support three-way calling with both legs using OPUS.

External Conference Bridge

You can connect only SIP or SP calls with the external conference bridge. When using an external conference bridge, the conference size is not limited by the phone but by the conference bridge itself. Check with your conference service provider on the conference size limit.

To start a three-way conference, the phone first sends a new INVITE to the SIP URL of the conference bridge to request the conference resources. Once successful, the conference bridge replies back with a 2xx response contact header that includes the context information for other participants to access the bridge for this conference call.

Enable an External Conference Bridge

You can configure the conference bridge and enable it with the Phone Settings option.

To enable an external conference bridge:

1. Configure the **X_ConferenceBridge** parameter on the SPn Service web page with the userid or (SIP) URL of the external conference bridge.
2. Enable the option **DECT Wireless::HandsetX::Calling Features::UseExternalConferenceBridge**.

   X refers to the numbers(1-8) of the handset.

Note that the phone assumes that only participants that are on the same SP service or using the same ITSP profile as the conference bridge can be referred to the bridge. For participants that are referable, the phone keeps them in the conference using local mixing. They are then subject to the local mixing limit.

Add a Participant to the Conference Bridge

You can add participants to the conference bridge with conference parameter.

To add a participant to the conference bridge:

1. Make a new call to the target number (or answer a new incoming call from another party, if applicable).

   The phone automatically holds the call to the conference bridge.
2. Select **Option::Conference** when the called target answers.

   The call to the conference bridge is resumed while the new remote party is referred to the same conference bridge contact to be added to the conference bridge. You can add more participants until the conference reaches the specified limit.

Call Routing

Call routing is the process by which the device sets up a call bridge or an endpoint call based on information like the trunk on which the call originates, the caller's number, and the called number.

From the device's perspective, calls originated from the trunk side are considered inbound calls, while calls originated from an endpoint are outbound calls. The call routing rule syntaxes for inbound calls and outbound calls are slightly different.

Call routing rules are parameters used to instruct the device how to route calls. A call can transform into a call bridge or an endpoint call after being routed by the device according to the given routing rules. Call routing rule configuration relies heavily on digit maps. If you are not familiar with how digit maps work, please read the Digit Map Configuration section in this guide.

Inbound Call Route Configuration

Every trunk has a corresponding **InboundCallRoute** parameter in the device configuration. It is a comma-separated list of rules where each rule is also surrounded by a pair of curly braces `{}`. No extra white spaces are allowed. These rules tell the device how to handle an inbound call, such as sending it to the handset (and ringing the attached phone(s)).

The general format is:

InboundCallRoute:`= rule` OR `{rule},{rule},….`

Note that the curly braces can be omitted if there is only one rule in the route. The OR operator isn't part of the parameter syntax. It is used here to separate alternative values only.

A rule has the following format:

```
rule := peering-list : terminal-list
```

The following table shows the rule formats.

Rule Formats

Rule	Format	Notes
`peering-list :`	`peering,peering,…`	Comma-separated list of 0 or more peering objects.
`terminal-list :`	`terminal,terminal,….`	Comma-separated list of 0 or more terminal objects.
`peering :`	`caller-list > callee-list`	
`caller-list :`	`caller|caller|caller|…`	Vertical bar-separated list of 0 or more caller objects.

Rule Formats

Rule	Format	Notes
`callee-list :`	`callee|callee|callee| …`	Vertical bar-separated list of 0 or more callee objects.
`caller :`	`number` OR `embedded-digit-map` OR `?` OR `@`	`?` = anonymous, `@` = any number but anonymous.
`callee :`	`number` OR `embedded-digit-map` OR `@`	
`terminal :`	`SPx(arg)` OR `PPx(arg)`	`arg` object is optional.
`arg :`	`cid > target`	
`x :`	`1` OR `2` OR `3...`	Where applicable. Can be omitted if x = 1.
`cid :`	`spoofed-caller-number` OR `$1`	
`target :`	`number-to-call` OR `$2`	
`embedded-digit-map :`	`(Mlabel)` OR `digit-map`	

General notes:

- `Terminal-list` can be empty, which means to block this call. The preceding ':' can't be omitted. As many as four terminals can be specified in the list. The listed terminals are called/rung by the device simultaneously. This operation is known as forking the call. A terminal can be a trunk or an endpoint.
- Abbreviated terminal names are case-insensitive.
- `Number` and `number-to-call` are literal strings, such as `14089991234`.
- `Digit-map` is just any proper digit map, such as `(1xxx|xx.)`. Make sure to include the enclosing parentheses.
- `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`, `SP2`, `SP3`, `SP4`, `SP5`, `SP6`, `SP7 and SP8`.
- `$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.

Notes on peering-list and peering objects:

- `Peering-list` is optional in **InboundCallRoute**. If the 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 **InboundCallRoutes** are all equivalent:
  - `dt1`
  - `{dt}`
  - `{:dt}`
  - `{?|@>@:dt}`

- `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.
- `Caller-list` in a peering object can be empty. It implies the caller-list `@|?`, meaning any caller number including anonymous. The succeeding '>' can't be omitted if `caller-list` is empty but not the `callee-list`.

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 is `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 '>' can't be omitted if `target` is omitted but `cid` isn't.
- `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 and 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

`dt` OR `{dt}` OR `{:dt}` OR `{@|?>@:dt}` (all equivalent)

It says: Ring the handset for all incoming calls. This is the default **InboundCallRoute** for all trunks.

`{(14081223330|15103313456):aa},{(1800xx.|1888xx.):},{dt}`

It says: Ring both handset and AA for calls coming from `1 408 122 3330` or `1 510 331 3456`, block all `800`, `888`, and anonymous calls, and ring the handset for all other calls.

`{(x.4081113333|x.4152224444):aa},{dt}`

It says: Ring the AA for calls coming from any number that ends with `408 111 3333` or `415 222 4444`, and ring the handset 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.

Outbound Call Route Configuration

Every endpoint has an **OutboundCallRoute** parameter in the device configuration. It tells the device where to send the call when the endpoint attempts to make a call. Endpoints can call each other or an outside number using one of the trunks. The **OutboundCallRoute** syntaxes are almost identical to those of the **InboundCallRoute**. The differences are mainly in the implied value when an optional field is omitted, no caller objects, and one and only one terminal object per terminal-list in an **OutboundCallRoute**. Forking isn't supported when routing outbound calls.

The general format is:

OutboundCallRoute:= `rule` OR `{rule},{rule},`….

Note that the curly braces can be omitted if there is only one rule in the route. The OR operator isn't part of the parameter syntax. It is used here to separate alternative values only.

A rule has the following format:

```
rule := callee-list : terminal
```

where

- `callee-list:= callee|callee|callee|` …(vertical bar separated list of 0 or more callee object)
- `callee:= number` OR `embedded-digit-map` OR `@` (`@` = any number)
- `terminal:= DTx OR SPx(arg)` OR `PPx(arg)` (`arg` object is optional)
- `arg:= cid > target`
- `x:= 1` OR `2` OR `3...`(where applicable. Can be omitted `x` = 1.)
- `cid = spoofed-caller-number`
- `target = number-to-call` OR `$2`
- `embedded-digit-map = (Mlabel)` OR `digit-map`

General notes:

- A terminal can be a trunk or another endpoint.
- Abbreviated terminal names are case-insensitive.
- `Number` and `number-to-call` are literal strings, such as `14089991234`.
- `Digit-map` is just any proper digit map, such as `(1xxx|xx.)`. Make sure to include the enclosing parentheses.
- `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, SP2, LI, PP or DT.
- `$2` is an internal variable containing the called number of this outbound call, after any digit map transformation in the matched callee object.
- `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.

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 '>' can't be omitted if `target` is omitted but not the `cid`.
- `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 to place the more specific rules first in the **OutboundCallRoute** 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** for a 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**.

OutboundCallRoute Examples

`sp1` OR `{SP1}` OR `{:SP1}` OR `{@:Sp1}` (all equivalent)

This rule says: Make all calls using the SP1 Service, without any caller-id spoofing or digit transformation.

```
{(Mpli):pli},{(<**1:>(Msp1)):sp1},{(<**2:>(Msp2)):sp2},{(<**8:>(Mli)):li},{(<**9:>(Mpp)):pp}
```

This is the default **OutboundCallRoute** for the handset. It says:

- Dial *** to invoke the local device configuration IVR.
- `(Mpli)` and `pli` are substituted with the PrimaryLine's abbreviated name.
- Use SP1 Service to call all numbers that start with **1 and subsequent digits matching SP1 Service's **DigitMap**. 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**. 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 Service's **DigitMap**. Remove the **9 prefix from the resulting number before making the call.

Digit Map Configuration

A digit map serves to transform and restrict the number that can be dialed or called, and determine if you dialed sufficient digits to form a complete number. Each map is composed of one or more rules surrounded by parentheses (which MUST NOT be omitted). Here is the general format of a digit map:

```
(rule|rule|….|rule)
```

A digit map rule is a rule for matching a given sequence of digits. It can contain extra white spaces for readability. All spaces are removed by the device during parsing. A rule can contain one or more of the following elements:

- `literals` – Any combination of 0-9,*,#,+,-,A-Z,a-z, except m, M, s, S, x, X, which have special meaning in the digit map syntax. It matches digit sequences with exactly the same literals.
- `'literals'` – Everything inside a pair of single quotes is treated as a literal except for the single quote (') character.
- `x` – a wild card digit that matches any digit from 0-9. `x` is case-sensitive.
- `x.` – matches 0 or more `x`.
- `[123-7]` or `[135]` – A set of 1 or more digits surrounded by pair of `[ ]`. It matches any digit in the set. The `-` syntax represents an inclusive digit range, such as `0-9`, `3-7`. So `[123-7]` is equivalent to `[1-7]` or `[1234567]`.
- `S`, `S0`, `S1`, `S2`, `…S9` – Digit timer of 0, 1, 2, …,9 seconds. `S` is equivalent to `S1`. `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. `S` is case-sensitive. It should only be used either as the first element of a rule for hot/warm line implementation, or as the last element of a rule as a means of overriding the default interdigit timer.

- `<elements:literals>` – Substitute the digit sequence matching elements with the given literals. Single quote syntax isn't needed or allowed for the literals in this context. Special characters can be used here as they don't apply in this context either. Elements can be empty, in which case the '`:`' can be omitted. This case is useful for inserting some extra digits in certain part of the dialed digits. The literals part can be empty also but the '`:`' MUST NOT be omitted. This case is useful for removing part of dialed digits. Elements and literals MUST NOT be both empty.
- `(map)` – An embedded digit map for matching subsequent digits.
- `(Mlabel)` – A named embedded digit map for matching subsequent digits, where label is one of abbreviated terminal names. Possible choices are:
  - `(Msp1)` for ***SP1 Service*::DigitMap**
  - `(Msp2)` for ***SP2 Service*::DigitMap**
  - `(Msp3)` for ***SP3 Service*::DigitMap**
  - `(Msp4)` for ***SP4 Service*::DigitMap**
  - `(Msp5)` for ***SP5 Service*::DigitMap**
  - `(Msp6)` for ***SP6 Service*::DigitMap**
  - `(Msp7)` for ***SP7 Service*::DigitMap**
  - `(Msp8)` for ***SP8 Service*::DigitMap**
  - `(Mpp)` for ***OBiTALK Service*::DigitMap**

Starting with release 1.2, the following elements are added:

- `X` – A wildcard digit that matches 0–9 or *. This is equivalent to `[x*]` or `[0-9*x]`
- `@` – A wildcard character that matches any alphanumeric character except `#`
- `x?` – matches 0 or 1 `x`
- `@?` – matches 0 or 1 `@`
- `[^…]` – matches any single alphanumeric character that isn't in the set
- Allow alphanumeric and wildcard inside a set `[ ]`, such as `[x]`, `[X#]`, `[@#]`, `[a-zA-Zx]`

The last two elements imply that the device digit maps are recursive. Recursive digit maps allow digit maps to be re-used and make their specification more compact and readable. It is important that you don't specify digit maps that lead to infinite recursion. For example, a digit map must not include a named embedded digit map that references itself.

To bar users from calling numbers that match a rule, add a '`!`' in front of that rule in the digit map. The rule is then referred to as a barring rule.

Examples:

- `1408xxxxxxx` – Matches any 11-digit number that starts with `1408`.
- `011xx.` – Matches any number that starts with `011` followed by one or more digits.
- `<1408>xxxxxxx` – Matches any 7-digit number. The device prepends `1408` to the number when making the call.
- `<:1408>xxxxxxx` – Equivalent to the last example.
- `<+>1xxxxxxxxxx` – Prepends '+' to any 11-digit number that starts with `1`.
- `<**1:>1408xxxxxxx` – Matches any number that starts with `**11408` followed by 7 digits. The device removes the `**1` prefix when making the call.
- `*74(x|xx)` – Matches any number that starts with `*74`, followed by 1 or 2 digits.

- **1(Msp1) – Matches any number that starts with **1 and with the rest of digits matching the **DigitMap** in the SP1 Service.
- <:1234> – Matches an empty phone number and replaces with 1234. This is the syntax for a hotline to 1234.
- <S0:1234> – Equivalent to the last example.
- <:#> – Hotline to the number #.
- <S0:#> – Equivalent to the last example.
- <S4:1234> – Call 1234 if no digits entered for 4 seconds. This is the syntax of a warm line.
- xx.853 7683 – Matches any number with at least 8 digits and ends with 8537683, such as 15108537683, 98537683.
- (x.408 223 1122) – Matches any number with at least 10 digits and ends with 408 223 1122, such as 4082231122 or 1408 223 1122.
- xx.<#> – Adds a # to the end of any number with 1 or more digits.
- !1900xxx xxxx – Barring all 11-digit numbers that start with 1900.
- [^*]@@. – Arbitrarily long alphanumeric sequence (except #) that doesn't start with *
- xx? – Any 1- or 2-digit number.
- (1xxxxxxxxxxS0|xx.) – Arbitrarily long digit sequence not starting with 1. Otherwise it is limited to 11 digits.

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 xxxxxxx, 1xxxx, 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 xxxx, 1234, 1xxx, <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 `1xxx`. 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:

- Label: ipd
- Digit Map:
  (xx.<*:@>xx?x?<*:.>xx?x?<*:.>xx?x?<*:.>xx?x?|xx.<*:@>xx?x?<*:.>xx?x?<*:.>xx?x?<*:.>xx?x?<*::>xx?x?x?x?)

The map (Mipd) is referenced in the default setting of the **DigitMap** in ITSP Profiles A through H. It supports the following two forms of IPv4 dialing:

- <user-id>*<a>*<b>*<c>*<d>
- <user-id>*<a>*<b>*<c>*<d>*<port>

where <user-id> is an arbitrary length numeric user-id, such as 100345, <port> is a port number in the range 0–65535, and each of <a>,<b>,<c>,<d> is a 1- to 3-digit pattern in the range 1–255 that identifies one byte of an IP address. The dialed number is translated into <user-id>@<a>.<b>.<c>.<d> and <user-id>@<a>.<b>.<c>.<d>:<port>. Here are some examples:

> 1234*192*168*15*113 maps to 1234@192.168.15.113
>
> 123456*192*168*15*180*5061 maps to 123456@192.168.15.180:5061

Third-Party Servers

This section provides information on configuring phones and features with third-party servers.

Broadsoft

You can configure VVX D230 DECT IP Phones with BroadSoft server options.

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 view and change the network-provided features from the phone, you must enable the **SPn Service – Calling Features::X_ASFeatureEventSubscribe** option and enable the individual network-provided feature you want users to access from the handset.

> 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. You can set the expires value of the subscription dialog (initiated by the phone per SP service with the feature enabled) using the parameter **ITSP Profile X–Feature Configuration::X_ASFeatureEventSubscribeExpires**. When a setting is changed, the server also updates the phone with a NOTIFY that specifies the latest settings of just the affected features.

You can access **Call Forward** and the **Do Not Disturb** network provided features from the phone menu or softkey. You can also change any of the above network provided service settings from the web page of the handset.

Call Forward All

The functionality provided by **Call Forward All** is similar to that of the CallForwardUnconditional function provided natively by the handset (per line). Poly recommends that you disable the native version when using the network-provided version to avoid ambiguity.

To configure the Call Forward All settings:

» Enable the option **SPn Service – Network Provided Services::CallForwardAlways**.

Note that you can specify the number to forward all incoming calls. These settings are submitted to and stored on the server.

Call Forward Busy

To configure the Call Forward Busy settings:

» 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.

Call Forward No Answer

To configure the Call Forward No Answer settings:

» Enable the option **SPn Service – Network Provided Services::CallForwardNoAnswer.**

This feature is similar to the CallForwardOnNoAnswer that is available natively on the phone.

Do Not Disturb

You can enable the SPn Service option to view and change the settings from the phone.

To view and change the Do Not Disturb settings:

» Enable the option **SPn Service – Network Provided Services::DoNotDisturb**.

The functionality provided by this feature is similar to that of the **DoNotDisturb** that is available natively on the phone.

BroadSoft XSI Features

XSI features is a collection of features provided with a BroadSoft XSI application server. The phone makes XSI features available per SP/SIP service, so you can configure as many as eight independent sets of XSI services per phone, one per SP service.

You can access some of the XSI features from the phone by launching dedicated apps (such as Network Directories) via the handset menu.

BroadSoft XSI Feature Parameters

Parameter Group	Parameter	Description
ITSP Profile X – SIP	***X_XsiServer***	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.
ITSP Profile X – SIP	***X_XsiServerPort***	The server port. If not specified (or 0), the default port is used (80 for HTTP or 443 for HTTPS).
ITSP Profile X – SIP	***X_XsiServerScheme***	Must be HTTP or HTTPS.
SPn Service – SIP Credentials	***X_XsiUserName***	The username to authenticate to the XSI server with. If not specified (blank), the phone forms the user name as: {sip-userid}@{sip-domain} where {sip-userid} is the SIP Account User ID that is used for SIP Registration on the same SP service, and {sip-domain} is the domain name that is used for SIP Registration on the same SP service.
SPn Service – SIP Credentials	***X_XsiPassword***	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.

Network Directories

Network directories are directories hosted by a server somewhere in the network. With the BroadSoft BroadWorks platform, the phone supports the Enterprise Directory of a network. For more information on setting up and managing the directories on server, refer to the BroadSoft documentation.

To access this service from the handset, you must enable the **SPn Service – Network Provided Services::Directory** option. You can invoke the network directory service of a specific SP service from the handset by **Directories** from the Main menu of the handset UI. The SP service you use is controlled by the parameter **Phone Settings – Network Directory::VoiceService**. For the **Directory** to show on the phone's Main menu, you must enable **SPn Service – Network Provided Services::Directory** of the corresponding SP service.

Service Providers

Use the following information for SIP-based configurations. Each ITSP configuration is grouped together as an ITSP profile. The VVX D230 refers to them as ITSP Profile A through ITSP Profile H.

Voice Services

- SP1–8
- OBiTALK

SIP Service Provider Features

You can configure up to four SIP accounts or SIP Trunks on the device. For the purposes of this guide and elsewhere on the system web interface, documentation, and the OBiTALK portal, the term ITSP describes the logical entity providing the SIP Trunk service to the device. When the device is used with an IP PBX, the IP PBX takes the place of the ITSP if it is the entity providing the SIP Trunk account credential and connectivity to the device.

Each ITSP configuration is grouped together as an ITSP Profile, referred to as ITSP Profiles A, B, C, and D. On the other hand, the SP service account specifics are grouped under the heading ***SPn Service***, where *n* = 1–8. An ITSP Profile includes such parameters as **ProxyServer**, **OutboundProxy**, and **DigitMap**, but doesn't include account-specific parameters. An SP Service includes account-specific parameters such as **AuthUserName** (usually the phone number of the account), **AuthPassword**, **CallerIDName**, and **X_ServProfile** (which ITSP Profile to assume). If the SP Services use the same ITSP, then only one ITSP Profile needs to be configured with all SP Services referred to the same profile.

From the device point of view, the SP*n* Service using ITSP Profile *X* is enabled with the following minimal settings:

- ***ITSP Profile X – SIP*::ProxyServer** = *Not Blank*
- ***SPn Service*::Enabled** = Yes
- ***SPn Service*::AuthUsername** = *Not Blank*

where *X* = A or B, *n* = 1–8. Otherwise, the service is considered disabled.

SIP Registration

Devices can be set periodically register with a SIP Proxy Server or SIP Registration Server. SIP Proxy Server and SIP Registration Server can be different, although they are usually the same in practice. SIP Proxy Server is a required parameter that must be configured on the device. The Registration Server is optional and assumed to be the same as the SIP Proxy Server if it isn't configured on the device.

The main purpose of registration is to create and maintain a dynamic binding of the SIP account to the device's local contact address. The service provider can also rely on this periodic message to infer if the device is online and functional. Each device takes only one local IP address that is either statically assigned in the device's configuration, or dynamically obtained from a local DHCP server. The SP*n* services (for *n* = 1 through 8) each use a different local contact port for sending and receiving SIP messages (defaults are 5060, 5061, 5062, and 5063).

Note that dynamic address binding through periodic registration isn't strictly necessary if the local IP address of the device doesn't change. The device's contact address can be statically configured on the Registration Server.

SIP Outbound Proxy Server

An outbound proxy server can be configured on the device such that all outbound requests are sent via the outbound proxy server instead of directly to the SIP Proxy Server or Registration Server.

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

Even though your device 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 handset to the server are formed based on these values, not based on the **OutboundProxy** value. The **OutboundProxy** value should never appear in the SIP requests generated by your device, unless the **OutboundProxy** parameter has the same value as **ProxyServer**.

Some server implementations include the outbound proxy server in a Record-Route header such that your device 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 ***ITSP Profile X – SIP*****::X_BypassOutboundProxyInCall** option. However, this option has no effect when the **OutboundProxyTransport** is TCP or TLS, as your device always uses the same connection to send messages to the server.

DNS Lookup of SIP Servers

When sending out SIP requests to the server, the device looks up the IP address of the server using standard DNS query if the server is specified as a domain name instead of an IP address. If an Outbound Proxy Server is configured, it's 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:

- Try looking up the name as DNS A Record. If not found,
- Try looking up the name as DNS SRV Record. If not found,
- Try looking up the name as DNS SRV Record with "_sip._udp." prepended to the host name. If not found, fail the request.

If the result from the DNS query is an SRV record, the server port is taken from that record also. The server port value configured on the device is ignored. Otherwise, the server port is taken from the configured value or uses port 5060 if none is specified.

NAT Traversal Considerations

If the device sits behind a NAT router (typically the case), it can discover the mapped external address corresponding to its local SIP contact address as seen by the server 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 the device its mapped IP address and port number. This method is used if periodic registration is enabled on the device.
- From the response to a STUN binding request the device sent to a STUN server. This method is used by enabling **X_KeepAliveEnable** and setting **X_KeepAliveMsgType** to “stun”. The keep-alive messages are sent to the same server where a REGISTER request would be sent.

The device always uses the mapped external contact address in all outbound SIP requests instead of its local contact address if one is discovered by either method discovered above.

SIP Proxy Server Redundancy and Dual REGISTRATION

Server Redundancy specifically refers to the device’s capability to:

- Look for a working server to REGISTER with from among a list of candidates.
- Switch to another server once the server that it currently registers with becomes unresponsive. In other words, 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 the device currently registers with.

If 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 the device as a domain name instead of an IP address. This allows the device to collect a list of candidate servers based on DNS query.

The domain name can 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 are obtained, the device first looks for a working server according to the stated priority. A *working server* means one that the device can successfully register with. This is known as the *Primary Server*. Subsequently, the device maintains registration with the primary server the usual way. However, if no working server is found after traversing the entire list, device takes a short break and repeats the search in the same order.

While maintaining registration with the Primary Server, the device 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 the device is trying to fall back on is known as the *primary fallback list*, which may be empty.

In addition, the device 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 parameter **X_SecondaryRegistration** to YES. If **X_ProxyServerRedundancy** is NO, however, **X_SecondaryRegistration** doesn’t take any effect. If this feature is enabled, as soon as a primary server is found, the device 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, the device forms a *secondary fallback list* to continually attempt to fall back on if the list isn’t empty.

The intervals for checking the primary fallback list and the secondary fallback list are configured in the **X_CheckPrimaryFallbackInterval** and **X_CheckSecondaryFallbackInterval** parameters. These parameters are specified in seconds and the default value is 60 for both.

Notes:

- Existence of a secondary server implies a primary server exists.
- If the secondary server exists, it immediately becomes the primary server when the current primary server fails. The device then starts searching for a new secondary server if the candidate set isn't empty.
- The candidate list can change (be lengthened, shortened, have its priority changed, and so forth) on every DNS renewal (based on the entry's TTL). The device rearranges the primary and secondary servers and fallback lists accordingly, whichever applies.

If the server redundancy feature is disabled, the device resolves only one IP address from the server's domain name, and won't try other IP addresses if the server isn't responding.

SIP Privacy

The device observes inbound caller privacy and decodes caller's name and number from SIP INVITE requests by checking the FROM, P-Asserted-Identity (PAID for short), and Remote-Party-ID (RPID for short) message headers. All these headers can carry caller's name and number information.

If PAID is present, device takes the name and number from it. Otherwise, it takes the 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. This privacy can indicate one of the following options:

- *off* = no privacy requested. The device shows name and number.
- *full* = full privacy requested. The device hides both name and number.
- *name* = name privacy requested. The device shows the number but hides the name.
- *uri* = uri privacy requested. The device shows the name but hides the number.

Regardless, if PAID exists or not, the device always takes the privacy setting from the RPID if it's present in the INVITE request. Note that if the resulting caller name is "Anonymous" (case-insensitive), device treats it as if the caller is requesting full privacy.

For outbound calls, caller's preferred privacy setting can be stated by the device in a RPID header of the outbound INVITE request. To enable this behavior, the ***ITSP Profile X – SIP*****::X_InsertRemotePartyID** parameter must be set to YES or TRUE, which is the default value of this parameter. The device supports only two outbound caller privacy settings: privacy=off or privacy=full. The RPID header generated by the device carries the same name and number as the FROM header. If outbound caller-ID is blocked, the device sets privacy=full in RPID, and also sets the display name in the FROM and RPID headers to "Anonymous" for backward compatibility. The device won't insert PAID in outbound INVITE requests.

STUN and ICE

The device supports standard STUN based on RFC3489 and RFC5389 for passing inbound RTP packets to the device sitting behind NATs. The parameters that control the STUN feature are found in the **ITSP Profile X – General::** section:

- **STUNEnable** – Enables this feature (default is NO or 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 is `3478` (standard STUN port).

The STUN feature used in this context is only for RTP packets, not SIP signaling packets, which typically don't require STUN. The device sends a STUN binding request right before making or answering a call on SP1/2. If the request is successful, the device 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 external address in the SDP.

Standard RTP requires the use of an even-numbered port in the m= line. If the external port isn't an even number, the device changes the local RTP port and redoes STUN, and continues to do this as many as four times or until an even external port number is found. If the fourth trial still results in an odd external port number, the call goes on without using an external address in the SDP.

The device supports standard ICE based on RFC5245. ICE is done on a per-call basis for automatically discovering which peer address is the best route for sending RTP packets. To enable ICE on the device, set the ***ITSP Profile X – General*****::X_ICEEnable** parameter to YES (or TRUE). The default is NO (or FALSE).

ICE is effective if STUN is also enabled. However, STUN not a requirement for using ICE on the device. If STUN is enabled and an external RTP address different from its local address is discovered, the device offers two ICE candidates in its SDP:

- The local (host) address (highest priority)
- The external (srflx or server reflexive) address

Otherwise, only the local host candidate is shown in the device's SDP. Note that the device 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, the device sends STUN requests to each peer candidate from its local RTP port. As soon as it receives a response from the highest priority candidate, the device concludes ICE and uses this candidate to communicate with the peer subsequently. Otherwise, the device allows as long as 5 seconds to wait for the response from all the candidates, and selects the highest priority candidate that has a response. Once ICE completes successfully, the device further applies symmetric RTP concept to determine the peer's RTP address (that is, sends them to the address from which the peer's RTP packets are coming).

RTP Statistics – the X-RTP-Stat Header

When ending an established call, the device can include a summary of the RTP statistics collected during the call in the SIP BYE request or the 200 response to the SIP BYE request sent by the peer device. The summary is carried in an X-RTP-Stat header in the form of a comma-separated list of fields. The 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, set the ***ITSP Profile X – SIP*::X_InsertRTPStats** parameter to YES (or TRUE).

Media Loopback Service

The device supports the media loopback draft as described in *draft-mmusic-media-loopback-13.txt*. You can enable or disable this feature from **System Management > Device Admin > Media Loopback**.

The device 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 the device acts as a loopback mirror, it always sends primer packets so that incoming packets can get through NAT/Firewall. The media loopback feature is controlled by the following parameters (in the ***Device Admin – Media Loopback*** section):

- **AcceptMediaLoopback** – Enable device to accept incoming call that requests media loopback. Default is YES.
- **MediaLoopbackAnswerDelay** – The delay in ms before the device 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.

The device rejects an incoming media loopback call if:

- Handset port is off-hook.
- Handset port is ringing.

The device terminates an inbound media loopback call already in progress when:

- Handset port is off-hook.
- Handset port 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 an RTP packet loopback call.

Note that outbound Media Loopback Call isn't subject to call duration limit. It lasts until you hang up or until the called device ends the call.

For more information on general ITSP parameters, see the ITSP Profile X – General Web Page (X = A, B, C, D, E, F, G, H) Parameter Guide table in the Parameter Reference section.

For more information on ITSP SIP settings parameters, see the ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide table in the Parameter Reference section.

For more information on ITSP RTP settings parameters, see the ITSP Profile X – RTP Web Page (X = A, B, C, D, E, F, G, H) Parameter Guide table in the Parameter Reference section.

Using SPn as a Proxy for a SIP IP Phone

An SP service can be set up as a proxy for a legacy IP phone to let the phone access OBiTALK or OBiBlueTooth (on SP*n*) installed on the device. This proxy mode of operation must be explicitly enabled in the SP 's configuration on the device. It is disabled by default. The IP phone using this proxy service is known as the *local_client* of the SP service. It must be installed on the LAN side of the device.

In this mode, SP*n* accepts SIP Registration from the client device from the LAN side, which must be using the same user-id and password as this SP*n*'s **AuthUserName** and **AuthPassword** parameters for authentication. This client device can also send SIP INVITE to the device at this SP to make calls. This SP's **InboundCallRoute** must be set up with the proper routing rule to handle calls from the *local_client*.

The SIP Proxy Server parameter on the client device must be sent to:

```
<obi-number>.pnn.obihai.com:<spn-user-agent-port>
```

where <obi-number> is the 9-digit OBi number of this device, and <spn-user-agent-port> is SP*n*'s **X_UserAgentPort** parameter.

For example, SP1 has a *local_client* with the user-id 4086578118. The client wishes to make and receive calls on SP3. The SP1 **InboundCallRoute** shall include the following rule:

```
{4086578118>:sp3}
```

The SP3 **InboundCallRoute** shall be: {sp1(**408657118@local_client**)}

For more information on SP*n* services parameters, see the SPn Services (n = 1, 2, 3, 4, 5, 6, 7, 8) Settings table in the Parameter Reference section.

OBiTALK Service Settings

For more information on OBiTALK calling features parameters, see the OBiTALK Calling Features Parameter Guide table in the Parameter Reference section.

Parameter Reference

Use the following VVX D230 parameters to configure your device.

Depending on your device or your settings, the system web interface may not display all of these parameters.

Status Parameters

The Status Parameters web pages show read-only values for certain parameters on your device. They include these pages:

- System Status Settings
- Call Status Settings
- Call History
- SP Services Stats Settings

System Status Settings

The System Status page displays the status of your device and the configured services. It also displays the device product information.

System Status Settings

Parameter	Description
WAN Status (DeviceInfo.Network.Status.WAN.)	
AddressingType	Method currently used by the handset to get an IP address assignment. Example value: `DHCP`
IPAddress	IP address currently assigned to the handset when using static IP addressing. Example value: `192.168.15.165`
SubnetMask	Subnet mask to use when using static IP addressing. Example value: `255.255.255.0`
DefaultGateway	Gateway to use when using static IP addressing. Example value: `192.168.15.1`
DNSServer1	URL for domain name server 1 when using static IP addressing. Example value: `8.4.4.4`

System Status Settings

Parameter	Description
DNSServer2	URL for domain name server 2 when using static IP addressing. Example value: `4.2.2.2`
MACAddress	MAC address installed on the handset. Example value: `9CADEF90004E`
LLDP-MEDStatus	Enables LLDP media endpoint discovery for improved network connections. Example value: `Enabled`
Product Information (DeviceInfo.)	
ModelName	Your device's model name. Example value: `VVXD230`
MACAddress	Your device's MAC address. Example value: `9CADEF90004E`
SerialNumber	Your device's serial number. Example value: `88H01NA00ZXV`
OBiNumber	Your device's OBi number, a value that uniquely identifies your handset to Polycom and to other OBi devices. Example value: `552 860 300`
HardwareVersion	Your device's hardware version. Example value: `1.1`
SoftwareVersion	Your device's installed software version. This value changes with a firmware update or downgrade. Example value: `6.3.0.15058`
SystemTime	Shows the current time on the system. Example value: `15:32:35 01/29/2019, Wednesday`
UpTime	With last Reboot Reason in parentheses. Example value: `20 Days 5:04:13 (2)`
CertificateStatus	Indicates if a device certificate is installed on the device. Example value: `Installed`
CustomizationStatus	Indicates if this device is a customized unit. Example value: `Generic`
SPn Service Status (VoiceService.1.VoiceProfile.1.Line.n.), n = 1 – 8	
Status	Registration status of this service. If there are problems with the registration or authentication, the SIP 4xx – 6xx error code and error message display here. This is useful information for troubleshooting issues with SIP-based services. Example value: `Registered (server=192.168.15.118; expire in 39s)`

System Status Settings

Parameter	Description
PrimaryProxyServer	IP address of the current primary proxy server if proxy server redundancy is enabled on this service. Example value: `10.100.123.234`
SecondaryProxyServer	IP address of the current secondary proxy server if proxy server redundancy and secondary registration are both enabled on this service. Example value: `10.100.234.123`
CallState	Describes the state of an active call on this service. Example value: `0 Active Calls`
OBiTALK Service Status (VoiceService.1.X_P2P.1.Stats.)	
Status	Connection status with the OBiTALK network. Example value: `Normal (User Mode)`
CallState	Describes the state of an active call on OBiTALK. Example value: `0 Active Calls`

Call Status Settings

The Call Status page shows a number of running call statistics and state parameters for each active call. The call status is only available during the lifetime of the call.

Call Status Descriptions

Status	Description
Peer Name	Call peer’s same.
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.
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.
RTP Byte Count	Total number of RTP bytes transmitted and received.
Peer Clock Differential Rate	Clock difference between this handset and the peer in ppm (parts per million).

Call Status Descriptions

Status	Description
Packets Out-of-Order	Number of received RTP packets that are out of order.
Packets Lost	Number of incoming RTP packets assumed lost.
Packet Loss Rate	Number of incoming RTP packets assumed lost rate in percent.
Packet Drop Rate	Number 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.

Call History

The Call History page shows the last 200 calls. Detailed call information is available, including what terminals were involved, the name (if available) of the peer endpoints making the call and the direction / path the call took, and the time events took place.

The following buttons are available:

- **Remove All**: Clicking this button erases the entire call history.
- **Save All**: Clicking this button saves the call history to the `callhistory.xml` file.

SP Services Stats Settings

You can find statistics relevant to SP*n* on the SP*n* Stats page, where *n* = 1–8.

SP Services Statistics

Parameter	Description
ResetStatistics	
ResetStatistics	This is a Boolean option to reset the statistics for this SP Service. After submitting this change, the value reverts automatically to the default value. Default setting: `false`
RTP Statistics	
PacketsSent	Total RTP packets sent on this line.
PacketsReceived	Total RTP packets received on this line.
BytesSent	RTP payload bytes sent for this line.
BytesReceived	RTP payload bytes received for this line.
PacketsLost	Number of RTP packets lost on this line.
Overruns	Number of jitter buffer overruns on this line.
Underruns	Number of jitter buffer underruns on this line.

System Management Parameters

The System Management parameter web pages show network parameters on your device. They include these pages:

- WAN Settings
- Auto Provisioning Settings
- Device Admin Settings
- Device Update

WAN Settings

This page lists the Ethernet settings for your device.

WAN Settings

Parameter	Description	Default Setting
Internet Settings (DeviceInfo.WAN)		
AddressingType	Method currently used by the handset to get an IP address assignment. Example value: `DHCP`	`DHCP`
IPAddress	IP address currently assigned to the handset when using static IP addressing. Example value: `192.168.15.165`	
SubnetMask	Subnet mask to use when using static IP addressing. Example value: `255.255.255.0`	`255.255.255.0`
DefaultGateway	Gateway to use when using static IP addressing. Example value: `192.168.15.1`	
DNSServer1	URL for domain name server 1 when using static IP addressing. Example value: `8.4.4.4`	
DNServer2	URL for domain name server 2 when using static IP addressing. Example value: `4.2.2.2`	
VLANEnable	VLAN Operation Bool.	
VLANID	Valid range is 0 - 4094 (4095 is reserved). 0 menas VLAN is disabled and egress packets are not tagged by the device. This setting applies to all packets sent by the device.	`0`
VLANPriority	Valid choices are 0 - 7. This setting applies to all packets sent by the device.	`0`

WAN Settings

Parameter	Description	Default Setting
VLANDiscovery	Enables taking VLAN setting from DHCP options. Choice of: • `Disabled` • `Fixed` • `Custom`	`Disabled`
VLANDiscoveryOption	Specifies which DHCP option to use for VLAN discovery.	`129`
802_1XMode	Port-based network access control provides an authentication mechanism to attach to a LAN. Choice of: • `Disable` • `MDS` • `TLS` • `TTLS-MSCHAPv2` • `PEAP-MSCHAPv2`	`Disable`
802_1XIdentity	User name for 802.1x authentication.	
802_1XPassword	Password for 802.1x authentication.	
802_1XTLSSecurityProfile	The TLS platform profile to use for device certification.	`1`
LLDP-MED	Enable LLDP-MED discovery.	
LLDP-MEDExclusivePeriod	Delay in seconds before getting or setting up an IP address based on **AddressingType** to exclusively perform LLDP-MED.	`5`
LLDP-MEDAssetID	The LLDP-MED Asset ID to broadcast during LLDP-MED discovery.	`$OBN`
OpenSSLCiphers	OpenSSL ciphers to support for all SSL/TLS connections. An empty value tells the device to use the default ciphers. A valid value must start with `DEFUALT:` or `HIGH:`.	Yes
Local Time (DeviceInfo.Time.)		
CurrentLocalTime	Current local date and time of the device (read-only parameter).	
CurrentNTPServer1	Host name or IP address of the current first NTP server.	None
CurrentNTPServer2	Host name or IP address of the current second NTP server.	None
Time Service Settings (DeviceInfo.Time.)		
NTPServer1	Host name or IP address of the first NTP server.	`pool.ntp.org`
NTPServer2	Host name or IP address of the second NTP server.	
LocalTimeZone	Local time zone.	`GMT-8:00(Pacific Time)`

WAN Settings

Parameter	Description	Default Setting
DaylightSavingTimeEnable	Enables daylight saving time on the unit.	Yes
DaylightSavingTimeStart	Daylight Saving Time Start Date Format: month/day/week/hh:mm::ss, where month = 1-12, day = 1-21, weedkay=0, 1-7 (0=special), hh = 0-23, mm = 0-59, ss = 0-59. 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 > 0, or on the weekday on or before the last-day-of-the-given-month+day+1 (note that day = -1 equivalent to last day of the month). Seconds can be omitted if the value is 0. Minutes and seconds can be omitted if both values are 0.	`3/8/7/2`
DaylightSavingTimeEnd	Daylight Saving Time End Date. Same format as **DaylightSavingTimeStart**.	`11/1/7/2`
DaylightSavingTimeDiff	Amount of time to add to current time during Daylighyt Saving Time. Format: [-]hh:mm:ss. Seconds can be omitted if the value is 0. Minutes and seconds can be omitted if both values are 0.	`1`
DHCP Client Settings (X_DHCPC.)		
ExtraOptions	Comma-separated list of extra DHCP options to be requested. Choice of: • `42` • `66` • `150` • `159` • `160` • `161`	`66, 42`
DNS Control (X_DNScontrol.)		
DNSQueryOrder	When more than one DNS servers are available, the unit attempts to resolve a domain name by querying each server sequentially until a successful result is received. The parameter controls the order in querying the servers. Choose from: • `DNS Server1, DNS Server2, DHCP Offered DNS Servers` • `DHCP Offered DNS Servers, DNS Server1, DNS Server2`	`DNS Server1, DNS Server2, DHCP Offered DNS Servers`

WAN Settings

Parameter	Description	Default Setting
DNSQueryDelay	When multiple DNS servers are available, the unit attempts to resolve a domain name by querying each server sequentially until a successful result is received. This parameter controls the number of seconds between successive DNS queries made by the unit for a given domain name. Choos form 0 to 5 seconds.	`2`
Local DNS Records (X_LocalDNSRec.)		

N, where N = 1-32	One of 32 Local DNS Records (numbered 1 – 32). Each record is a mini script of either of the following formats:
	• `Name=A,A,A,...`
	• `Name=R,R,R,...`
	where `Name` represents the domain name to be resolved locally, and has the format `prefix+domain` (such as `machine.sip+obihai.com`). 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`.
	`A` represents an `A` record, which is just an IP address, such as 192.168.12.17.
	`R` represents an SRV record and has the format: `{host:port,pri,wt}` where
	• `host` is a host name with or without domain part (such as `xyz, xyz.abc.com.`). A dot (.) at the end of host indicates it is a complete host name that does not require the domain to be appended.
	• `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).
	• `pri` is the priority. Valid value is 0 (highest) – 65535 (lowest).
	• `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.
	The enclosing curly brackets `{ }` are also optional if there is only one `R`, or if no comma appears inside the `R`.
	Examples:
	`_sip._udp+obihai.com=abc,xyz,pqr:5080,{mmm,2},{super.abc.com.}`
	`abc.obihai.com=192.168.15.118,192.168.15.108`
	Note: If the `A` record of a given host name cannot be found in any of the local DNS records, the handset attempts to resolve it using external DNS queries. Any change applied to a local DNS record needs a reboot to take effect.
		None

Auto Provisioning Settings

The Auto Provisioning web page shows all the parameters related to remote provisioning of the device.

Auto Provisioning Parameter Guide

Parameter	Description	Default Setting
Auto Firmware Update (X_DeviceManagement.FirmwareUpdate.)		
Method	Current operational method of auto firmware updating. Choose from: • `Disabled`: don't check for f/w upgrade from **FirmwareURL**. • `System Start`: Check for f/w upgrade from **FirmwareURL** just once on system start. • `Periodically`: Check for f/w upgrade from **FirmwareURL** on system start, and then periodically at the interval specified in the Interval parameter. • `Time of Day`: Check once at the given **TimeofDay** value. Note: The first firmware upgrade check on system start is performed after a random delay of 0 to 30 seconds.	`Disabled`
Interval	When **Method** is set to `Periodically`, this is the number of seconds between each checking of f/w upgrade check from **FirmwareURL**. If value is `0`, the device checks once only on system start (equivalent to setting **Method** to `System Start`).	`0`
TimeOfDay	Time of the day in `hh:mm[+rr]` format, valid when method is set to `Time of Day`.	`00:00+30`
RandomDelayRange	The range of delay in seconds inserted before the first attempt only. The minimum value is `0`.	`30`
FirmwareURL	URL of firmware package. URL must include scheme. Supported schemes are `http://` and `tftp://`	
TLSSecurityProfile	Security profile when using HTTPS. Choices are `1` or `2`.	`1`
DnsLookupType	Controls what type of DNS record to lookup. Choose from: • `A Record Only` • `SRV Record Only` • `Try Both`	`A Record Only`
DnsSrvPrefix	Controls whether to add a standard prefix to the domain name when looking up a SRV Record. For HTTP and HTTPS, the prefix to add is `_http._tcp.` . For TFTP, the prefix to add is `_tfto._udp`. Choose from: • `No Prefix` • `With Prefix` • `Try Both`	`No Prefix`
Username	Username for authentication, if needed, if scheme is `http://`	
Password	Password for authentication, if needed, if scheme is `http://`	

Auto Provisioning Parameter Guide

Parameter	Description	Default Setting
Suspend	Suspend firmware update until canceled.	`false`
ITSP Provisioning (X_DeviceManagement.ITSPProvisioning.)		
Method	Current operational method of Provisioning. Choose from: • `Disabled`: don't download from **ConfigURL**. • `System Start`: Download from **ConfigURL** just once on system start. • `Periodically`: Download from **ConfigURL** on system start, and then periodically at the interval specified in the Interval parameter. • `Time of Day`: Check once at the given **TimeofDay** value. 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.	`System Start`
Interval	When **Method** is set to `Periodically`, this is the number of seconds between download from **ConfigURL**. If value is `0`, device downloads once only on system start (equivalent to setting **Method** to `System Start`).	`0`
TimeOfDay	Time of the day in `hh:mm[+rr]` format, valid when method is set to `Time of Day`.	`00:00+30`
ConfigURL	URL of config file.	`tftp://$DHCPOPT66/$MAC.xml`
DnsLookupType	Controls what type of DNS record to lookup. Choose from: • `A Record Only` • `SRV Record Only` • `Try Both`	`A Record Only`
DnsSrvPrefix	Controls whether to add a standard prefix to the domain name when looking up a SRV Record. For HTTP and HTTPS, the prefix to add is `_http._tcp.`. For TFTP, the prefix to add is `_tfto._udp.`. Choose from: • `No Prefix` • `With Prefix` • `Try Both`	`No Prefix`
Override	Defines which local settings can be overridden by this provisioning. Choose from: • `All` • `All except user settings`	`All`
GPRM0 to GPRM7	Non-volatile generic parameters that can be referenced in other parameters, such as **ConfigURL**.	
TPRM0 to TPRM3	Temporary variables used in scripts for **ConfigURL**.	

Auto Provisioning Parameter Guide

<table>
<tr><th>Parameter</th><th>Description</th><th>Default Setting</th></tr>
<tr><td colspan="3">OBiTALK Provisioning (X_DeviceManagement.Provisioning.)</td></tr>
<tr><td>Method</td><td>Current operational method of Provisioning. Choose from:
• <code>Disabled</code>: Don’t download from ConfigURL.
• <code>System Start</code>: Download from ConfigURL just once on system start.
• <code>Periodically</code>: Download from ConfigURL on system start, and then periodically at the interval specified in the Interval parameter.
• <code>Time of Day</code>: Check once at the given TimeofDay value.
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.</td><td><code>System Start</code></td></tr>
<tr><td>Interval</td><td>When Method is set to <code>Periodically</code>, this is the number of seconds between download from ConfigURL. If value is <code>0</code>, device downloads once only on system start (equivalent to setting Method to <code>System Start</code>).</td><td><code>0</code></td></tr>
<tr><td>TimeOfDay</td><td>Time of the day in <code>hh:mm[+rr]</code> format, valid when method is set to <code>Time of Day</code>.</td><td><code>00:00+30</code></td></tr>
<tr><td>ConfigURL</td><td>URL of config file.</td><td></td></tr>
<tr><td>DnsLookupType</td><td>Controls what type of DNS record to lookup. Choose from:
• <code>A Record Only</code>
• <code>SRV Record Only</code>
• <code>Try Both</code></td><td><code>A Record Only</code></td></tr>
<tr><td>DnsSrvPrefix</td><td>Controls whether to add a standard prefix to the domain name when looking up an SRV Record. For HTTP and HTTPS, the prefix to add is <code>_http._tcp.</code>. For TFTP, the prefix to add is <code>_tfto._udp.</code>. Choose from:
• <code>No Prefix</code>
• <code>With Prefix</code>
• <code>Try Both</code></td><td><code>No Prefix</code></td></tr>
<tr><td>GPRM0 to GPRM7</td><td>Non-volatile generic parameters that can be referenced in other parameters, such as ConfigURL.</td><td></td></tr>
<tr><td>TPRM0 to TPRM3</td><td>Temporary variables used in scripts for ConfigURL.</td><td></td></tr>
<tr><td colspan="3">User-Defined Macro 0–3 ($UDM0 – $UDM3)</td></tr>
<tr><td>Value</td><td>The value can be any plain text or a valid canonical parameter name preceded by a $ sign. For example:
$X_DeviceManagement.<code>WebServer.Port</code>
Note: Here you MUST NOT enclose the parameter name following the <b>$ sign with braces or parentheses.</td><td></td></tr>
</table>

Auto Provisioning Parameter Guide

Parameter	Description	Default Setting
ExpandIn	This is a comma-separated list of canonical parameter names, where the macro expansion can be used. As many as three parameter names can be specified. Specify `ANY` to allow the macro to expand in any parameter. Example: `X_DeviceManagement.HTTPClient.UserAgent` Note: There is no **$** sign in front of the parameter name. The macro can't be used in any parameter value if this value is set to blank (the default).	
SyntaxCheckResult	This is read only status value regarding the syntax of the UDM. `Pass` means that this UDM is valid. Otherwise, it shows the syntax error detected by the device either in the **Value** or **ExpandIn** parameters of the UDM.	

$MACRO Expansion Supported by the Device

$MACRO Expansion Supported by the Device

Macro Name	Description	Where It Can Be Used
MAC	Device MAC address, such as `9CADEF000000`	ANY
MACC	Device MAC address with colons, such as `9C:AD:EF:00:00:00`	ANY
mac	Device MAC address in lower case with colons, such as `9c:ad:ef:00:00:00`	ANY
FWV	Firmware version, such as `1.0.3.1626`	ANY
HWV	Hardware version, such as `2.8`	ANY
IPA	Current device IP address, such as `192.168.15.100`	ANY
DM	Device Model Name, such as `VVXD230`	ANY
DMN	Device model number, such as `508`	ANY
OBN	Device OBi number, such as `200123456`	ANY
DSN	Device S/N, such as `88B01NA00000`	ANY
GPRM*n* *n*=0–7	Value of ***Auto Provisioning*****::GPRM*n***	***Auto Provisioning*****::ConfigURL**, ***Auto Firmware Update*****::FirmwareURL**

$MACRO Expansion Supported by the Device

Macro Name	Description	Where It Can Be Used
TPRM*n* *n*=0-3	Value of ***Auto Provisioning*::TPRM*n***	***Auto Provisioning*::ConfigURL**, **Auto Firmware Update::FirmwareURL**
UDM*n*, *n*=0-3	Value of ***User-Defined Macro n*::Value**	The value of ***User-Defined Macro n*::ExpandIn**

Device Admin Settings

The Device Admin web page includes the following configuration parameters.

Device Administration Parameter Guide

Parameter	Description	Default Setting
Web Server (X_DeviceManagement.WebServer.)		
Port	Web server port number.	`80`
AdminPassword	Administrator password, case-sensitive.	`admin`
UserPassword	User password, case-sensitive.	`user`
IVR (X_DeviceManagement.IVR.)		
Enable	Enables IVR for local configuration.	Yes
Password	IVR access password (must be all digits).	
Syslog (X_DeviceManagement.Syslog.)		
Server	IP address of the Syslog server where the device sends syslog debug messages to. If the value is blank, syslog is disabled.	
Port	Syslog server port number.	`514`
Level	Syslog message level.	`7`
TAG	A string of text no longer than 32 characters to prepend every syslog message sent out by this unit.	
HTTP Client (X_DeviceManagement.HTTPClient.)		
UserAgent	Value of the User-Agent header in all HTTP Requests that are used in firmware upgrade and auto provisioning.	`$DM`
TimeOut	A time limit specified in number of seconds such that any file download (firmware or configuration file) by the device via HTTP must be completed within this limit or the device aborts and concludes that the operation has failed for the reason of “taking too long to complete”.	`600`
ProxyServer	Host name or IP address of the HTTP proxy server.	

Device Administration Parameter Guide

Parameter	Description	Default Setting
ProxyServerPort	Destination port to connect to the HTTP proxy server. Range = [0:65535]. Don't choose a port at random.	`80`
ProxyAuthUsername	User name for proxy authentication.	
ProxyAuthPassword	Password for proxy authentication.	
BypassProxyServerForLocalAddress	Enables **BypassProxyForSubnets** parameter.	No
BypassProxyForSubnets	List of intranet subnets, which bypass the proxy server.	
External Port Security (X_DeviceManagement.X_PortSecurity.)		
PCPort	Locks the PC port so that the handset does not allow any network traffic in and out of that port.	No
Remote PCAP Server (X_DeviceManagement.X_RPCAPD.)		
Enable	Enables PCAP (Packet Capture) server function on the handset.	No
Ports	PCAP server port number.	`2002`
Client	List of clients allowed to connect to this server. An empty list means everyone is allowed.	None
Packet Capture (X_DeviceManagement.X_RemotePCAP.)		
Enable	Enable Remote PCAP server and start the daemon according to the following parameters.	Disabled
Status	Running status of utility capture.	
Interface		`Primary`
Storage	Where the captured packets are stored. Internal Storage: Volatile memory space is no more than 10MB.	`Internal Storage`
RestartCaptureOnReboot	Automatically start the capture according to the current configuration as soon as the specified network interface is created.	`Disable`
PromiscuousMode	The Promiscuous Mode.	`Enable`
WebAccessExcluded	Exclude traffic packets to or from the local web server.	`Enable`
PostponeFirmwareUpdate	Postpone auto firmware update if capturing is ongoing.	`Enable`

Device Administration Parameter Guide

Parameter	Description	Default Setting
Platform CA n (X_DeviceManagement.PlatformCACert.n.), n = 1 or 2		
DownloadURL	URL to download certificate	None
MD5CheckSum	MD5 checksum of the certificate file to be downloaded. Failure to provide this causes the device to try to download the same file on every reboot or restart.	None
CommonName	The common name set in the installed certificate. Read-only status field.	
FingerPrint	SHA1 fingerprint of the installed certificates.	
Obsolete	When set to true, the certificate is deleted from the device. Also, the certificate downloading process is ignored.	False
Custom Device Certificate n (X_DeviceManagement.CustomDeviceCert.n.), n = 1 or 2		
DownloadURL	URL to download certificate.	None
MD5CheckSum	MD5 checksum of the downloaded certificate.	None
CommonName	The common name set in the installed certificate. Read-only status field.	None
FingerPrint	SHA1 fingerprint of the installed certificates.	None
Obsolete	When set to true, the certificate is deleted from the device. Also, the certificate downloading process is ignored.	False
TLSPlatform Profile n (X_DeviceManagement.TLSPlatform.n.), n = 1 or 2		
CipherSuite	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 handset).	None
CACertList	The CA Certificate List to use in a TLS profile. Choice of: • `Default` • `Default+P1` • `Default+P2` • `All` • `Platform1` • `Platform2` • `Platform1+2`	`Default`
DeviceCert	The Device Certificate List to use in a TLS profile. Choice of: • `Polycom` • `Custom1` • `Custom2`	`Polycom`

Device Update

The Device Update web page provides the following functions:

- Firmware Update
- Backup Configuration
- Restore Configuration
- Reset Configuration

Firmware Update

You can update the firmware for your handset from the native web page. The firmware file must be stored locally on a computer that you can access with a web browser.

To update the firmware

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 stakes 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 redownload the firmware package and try again.
System Is Busy	The device is busy because one of the services in an active call or device provisioning is in progress.	Try to update again later.
Firmware Is Not Modified	The device is already running the same firmware as the one selected for update.	No need to upgrade.

Backup Configuration

The current configuration of the handset 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 handset.

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
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

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 handset 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 handset 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.

Service Providers Parameters

The Service Providers web pages show parameters for the provisioned service providers on your device. They include these sets of pages for each of the eight ITSP Profiles A through H.

- General Settings
- SIP Settings
- RTP Settings

ITSP Profile X (X = A, B, C, D, E, F, G, H)

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 device to the service providers.

General Settings

The following configuration parameters are available on this page.

ITSP Profile *X* – General Web Page (*X* = A, B, C, D, E, F, G, H) Parameter Guide

<table>
<tr><th>Parameter</th><th>Description</th><th>Default Setting</th></tr>
<tr><td colspan="3">ITSP Profile X – General (VoiceService.1.VoiceProfile.n.) for X = A, B, …, H corresponding to n = 1, 2, …, 8, respectively</td></tr>
<tr><td>Name</td><td>Human-readable string to identify the profile instance. Maximum length is 127 characters.</td><td></td></tr>
<tr><td>SignalingProtocol</td><td>Signaling protocols for this ITSP.</td><td>SIP</td></tr>
<tr><td>DTMFMethod</td><td>Method to pass DTMF digits to peer device. Choose from:
• Inband: DTMF tones are sent as inband audio signal
• RFC2833: DTMF tone events are relayed per RFC2833
• SIPInfo: DTMF tones are relayed with SIP INFO request
• Auto: Method to use based on call setup negotiation (either Inband or RFC2833 can be negotiated).</td><td>Auto</td></tr>
<tr><td>InbandDTMFVolume</td><td>DTMF tone volume when sending inband DTMF</td><td>15</td></tr>
<tr><td>X_UseFixedDurationRFC2833DTMF</td><td>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 device sends only one RTP digit event packet with a fixed duration of 150 ms regardless how long the digit has been pressed.</td><td>False</td></tr>
<tr><td>X_FixedDurationRC2833DTMP</td><td>The fixed duration (in ms) to use when X_UseFixedDurationRFC2833DTMF is set to True.</td><td>16</td></tr>
<tr><td>DigitMap</td><td>A digit map to restrict the numbers that can be dialed or called with this service. Maximum length is 511 characters.</td><td>(1xxxxxxxxxx|<1>[2-9]xxxxxxxxx|011xx.|xx.)</td></tr>
<tr><td>STUNEnable</td><td>Enables device to send a STUN binding request for its RTP port prior to every call.</td><td>No</td></tr>
<tr><td>STUNServer</td><td>IP address of domain name of the STUN Server to use.</td><td></td></tr>
</table>

ITSP Profile *X* – General Web Page (*X* = A, B, C, D, E, F, G, H) Parameter Guide

Parameter	Description	Default Setting
X_STUNServerPort	UDP listen port of the STUN Server.	`3478`
X_ICEEnable	Enables device to use ICE algorithm to find the best peer RTP address to forward RTP traffic for every call.	No
X_SymmetricRTPEnable	Enables device to apply symmetric RTP behavior on every call: That is, send RTP to peer at the address where incoming RTP packets are received from.	No
Service Provider Info (VoiceService.1.VoiceProfile.n.ServiceProviderInfo.) for X = A, B, …, H corresponding to n = 1, 2, …, 8, respectively)		
Name	Human-readable string identifying this service provider. Maximum length is 127 characters.	
URL	Website of this service provider. Maximum length is 127 characters.	
ContactPhoneNumber	Phone number to contact this service provider. Maximum length is 31 characters.	
EmailAddress	Email address to contact this service provider. Maximum length is 127 characters.	

SIP Settings

The following configuration parameters are available on this page.

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
ITSP Profile X – SIP (VoiceService.1.VoiceProfile.n.SIP.) for X = A, B, …, H corresponding to n = 1, 2, …, 8, respectively)		
ProxyServer	Host name or IP address of the SIP proxy server.	
ProxyServerPort	Destination port to connect to the SIP server.	`5060`
ProxyServerTransport	Transport protocol to connect to SIP server. Choose from: • `UDP` • `TCP` • `TLS`	`UDP`
RegistrarServer	Host name or IP address of the SIP registrar. If a value is specified, device sends REGISTER to the given server. Otherwise, REGISTER is sent to **ProxyServer**.	
RegistrarServerPort	Destination port to connect to SIP registrar.	5060

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_XsiServer	Host name or IP address of the Broadsoft XSI application server.	
X_XsiServerPort	Listening port of the Broadsoft XSI application server. If not specified or `0`, the default ports are 80 for HTTP and 443 for HTTPS.	`0`
X_XsiServerScheme	Scheme to access the Broadsoft XSI application server.	`HTTP`
UserAgentDomain	CPE domain string. If empty, device uses **ProxyServer** 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 ***SPn Service*::URI** is specified, additional rules applied in forming the AOR. See the description of the **URI** parameter for more details and examples.	
OutboundProxy	Host name or IP address of the outbound proxy. Outbound proxying is disabled if this parameter is blank.	
OutboundProxyPort	Destination port to be used in connecting to the outbound proxy.	`5060`
X_OutboundProxyTransport	Controls the SIP transport for the outbound proxy server, which can be different from that of the proxy server. Choose from: • `UDP` • `TCP` • `TLS` • `Follow ProxyServerTransport`	`Follow ProxyServerTransport`
X_UserAgentContactFollowProxyServerTransport	If enabled, the user agent uses a Contact and Via transport that agrees with **ProxyServerTransport**.	No
X_BypassOutboundProxyInCall	Enables bypassing the **OutboundProxy** inside the SIP dialog.	No
RegistrationPeriod	Nominal interval between device register in seconds.	`60`
X_RegistrationMargin	Number of seconds before current registration expires that the device should re-Register (for example, 5 seconds). If value is less than one, it is interpreted as a fraction of the current expires value (for example, 0.1 of 60 seconds is 6 seconds). If value is 0 or blank, the device determines a proper margin on its own.	
TimerT1	Value of SIP timer T1 in ms.	`500`

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
TimerT2	Value of SIP timer T2 in ms.	4000
TimerT4	Value of SIP timer T4 in ms.	5000
TimerA	Value of SIP timer A in ms.	500
TimerB	Value of SIP timer B in ms.	32000
TimerD	Value of SIP timer D in ms.	32000
TimerE	Value of SIP timer E in ms.	500
TimerF	Value of SIP timer F in ms.	32000
TimerG	Value of SIP timer G in ms.	500
TimerH	Value of SIP timer H in ms.	32000
TimerI	Value of SIP timer I in ms.	5000
TimerJ	Value of SIP timer J in ms.	32000
TimerK	Value of SIP timer K in ms.	5000
InviteExpires	Invite request Expires header value in seconds.	60
ReInviteExpires	Reinvite Expires header value in seconds.	10
RegisterExpires	Register Expires header value in seconds (not used at the moment).	3600
RegistersMinExpires	Register Min-Expires header value in seconds (not used at the moment).	15
RegisterRetryInterval	Register retry interval in seconds.	30
X_RegisterRetryResponseCode	A set of SIP register error response codes and the corresponding retry delay (in seconds) specified in a digit map format. See the default value on the right as an example, where the value to the left of the colon of each rule represents a set of 3-digit response codes and the value to the right of the colon is the waiting time in seconds. If the waiting time is given as a range (with a '-'), a randomized waiting time within the specified range is used.	(<40[17]:w120>|<40[34]:w120>|<99[01]:w120-200>|[4-9]xx)
DSCPMark	Diffserv code outgoing SIP packets.	0

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_SpoofCallerID	Allows outbound Caller ID spoofing. If set to Yes, device 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 another SP Service). Otherwise, device always its own account information to form the FROM header. Note that most service providers won't allow originating a call if the FROM header field doesn't match the account credentials. Enable this option only if you're sure that the service provider allows it.	No
X_UseRefer	Enables using SIP REFER for call transfer. If disabled, device bridges the call instead when performing a call transfer (which consumes some resources on the device).	Yes
X_ReferAOR	Enables using the target's AOR (Address of Record or public address) in Refer-To header of SIP REFER. If disabled, the target's Contact is used instead.	Yes
X_HoldReferee	Holds the Referee before a blind transfer if the call isn't placed on hold. This may allow reconnecting with the Referee if the blind call transfer fails.	No
X_Use302ToCallForward	Enables using the 302 response to INVITE for call forward. If disabled, device bridges the call legs instead when forwarding a call (and consumes some resources on the device).	Yes
X_UserAgentName	If a value is specified, device 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}`
X_ProcessDateHeader	Enables the device 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 device can use to adjust its local time and date without relying on NTP.	Yes
X_InsertRemotePartyID	Enables the device 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).	Yes
X_EnforcePAssertedIdentity	Take caller-ID from P-Asserted identity header only.	No
X_InsertPPreferredIdentity	Insert P-Preferred-Identity header in all outbound INVITE.	No
X_InsertPrivacyHdr	Inserts a 'Privacy:id' header in outbound INVITE for anonymous calls.	No

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_UseAnonymousFROM	Enables using "sip:anonymous@localhost" in FROM header of SIP INVITE when attempting to make an anonymous call.	No
X_SessionRefresh	Enables session refresh signaling (with SIP Re-INVITE) during a connected call. This allows the device to detect if the connection with the peer is broken abnormally so it can release the call. Disable this option if the ITSP doesn't support Re-INVITE sent from the client device.	Yes
X_SessionTimer	Enable standard session timer behavior based on RFC4028	No
X_SessionExpires	Session Expires before value. If session refresh is enabled, the device refreshes half-time before the session expires.	`20`
X_AccessList	A comma-separated list of IP addresses such that the device only accepts SIP requests coming from one of the given addresses. If the list is empty, the device accepts SIP requests from any IP address.	
X_InsertRTPStats	Enables the device 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.	Yes
X_MWISubscribe	Enables the device to SUBSCRIBE to the message-summary event package to support MWI and VMWI service. The device handles NOTIFY of this event package regardless of whether **MWISubscribe** is enabled.	No
X_MWISubscribeURI	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 doesn't contain '@', it's 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's used in the TO and FROM header as well as the Request-URI as is. The device forms the Request-URI of SUBSCRIBE the same way as the TO header, with an additional port number.	
X_MWISubscribeExpires	Periodic interval to renew SUBSCRIBE.	`3600`

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_ProxyServerRedundancy	Enables proxy redundancy feature on the device. To use this feature, device registration must be enabled and the SIP Registration Server or Outbound Proxy Server must be configured as a domain name.	No
X_SecondaryRegistration	Enables device to register with a secondary server in addition to the primary server. **X_ProxyServerRedundancy** must be enabled for this parameter to take effect.	No
X_CheckPrimaryFallbackInterval	Interval in seconds at which the device checks the primary fallback list of candidate servers.	`60`
X_CheckSecondaryFallbackInterval	Interval in seconds at which the device checks the secondary fallback list of candidate servers.	`60`
X_ProxyFailoverResponseCodes	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)`
X_InviteFailoverWaitRegTimer	Maximum time (in milliseconds) to wait for successful register failover to retry INVITE after failure.	`32000`
X_ProxyRequire	If this parameter isn't blank, the device includes a Proxy-Require header stating the value of this parameter in all SIP requests sent to the ITSP.	
X_MaxForward	Value for the Max-Forward header in all SIP requests sent by the device.	`70`
X_AcceptLanguage	If this parameter isn't blank, the device includes an Accept-Language header stating the value of this parameter in all SIP requests sent to the ITSP.	
X_DnsSrv	Enable DNS SRV lookup for the proxy server or the outbound proxy server.	Yes
X_DnsSrvAutoPrefix	Enables letting the device 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 `_sip._udp.` for SIP over UDP, `_sip._tcp.` for SIP over TCP, and `_sip._tls.` for SIP over TLS.	No
X_Support100rel	Enables support for RFC3262 (reliable provisional SIP responses). If enabled, the device announces this support in a SIP Supported header, and requires a caller to use this option if the caller also supports this feature.	No

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_UserEqPhone	Includes the parameter 'user=phone' in Request-URI and To-URI of outbound INVITE.	
X_UseTelURI	Enables using tel: in outbound SIP Request-URI and TO-URL	
X_CallWaitingIndication	Enables including an indication in an 18x response to the calling peer if this is a call- waiting situation. Choose from: • `No` • `Alert-Info`	`No`
X_DiscoverPublicAddress	Enables letting the device use the public IP address and port it has discovered as its SIP Contact address.	Yes
X_UsePublicAddressInVia	Enables using the discovered external IP address (instead of the unit's assigned local IP address) in outbound Via header.	No
X_PublicIPAddress	A static public IPv4 address, if specified, is used by the device to form its SIP Contact address.	
X_UseRport	Enables letting the device insert a blank rport parameter in the VIA header our outbound SIP messages. This option should be turned off if you're using port forwarding on the external router to route inbound SIP messages to the device.	Yes
X_DetectALG	Enables detecting upstream SIP ALG.	No
X_UseCompactHeader	Enables using compact form SIP message header names.	No
X_OmitContentLength	Omit Content-Length header if **ProxyServerTransport** and **X_OutboundProxyTransport** parameters are both `UDP`.	No
X_FaxPassThroughSignal	Selects the signaling method to indicate to the peer to switch to FAX passthrough. Choose from: • `ReINVITE` • `RFC2833` • `Auto` • `None`	`ReINVITE`
X_IncludeMessageHash	Includes an MD5 hash of all the SIP headers in an XMD5-Hash header. A hash of the SDP is also included in an x-md5-hash SDP attribute.	No
X_EchoServer	Name or IP address of an echo server for SIP ALG detection.	

ITSP Profile X – SIP Web Page (X = A, B, C, D, E, F, G, H) Settings Parameter Guide

Parameter	Description	Default Setting
X_EchoServerPort	Listening of the echo server for SIP ALG detection.	`5060`
X_EnableRFC2543CallHold	Enables interpretation of call hold indication per RFC2543.	No
X_VerifyServerDomain	Enable verification of server domain against its certificate on a SSL/TLS connection.	No
X_RejectKeyResponseCode	SIP response code and phrase to inbound INVITE, when the user presses the **End Call** key.	
X_Sticky18x	Ignore further 18x responses without SDP upon reviving the first 18x with SDP to INVITE.	Yes
Feature Configuration (VoiceService.1.VoiceProfile.n.SIP.), n = 1 - 8		
X_CallParkMethod	Select the method to use to park a call. Choice of: • `Feature Code` • `REFER`	`Feature Code`
X_DirectedCallPickupMethod	Select the method to use for directed call pickup. Choice of: • `Feature Code` • `INVITE+Replaces`	`Feature Code`
X_SharedLineMethod	Select the signaling method for shared line operation.	`call-info`
X_CallInfoSubscribeExpires	Call information subscription renewal interval in seconds. Set the value to `0` to disable subscription renewal.	`3600`
X_LineSeizeSubscribeExpires	Line-seize event subscription renewal interval in seconds.	`15`
Feature Codes (VoiceService.1.VoiceProfile.n.X_FeatureCode.), n = 1 - 8		
DirectedCallPickup	Code to invoke Directed Call Pickup feature on the ITSP.	`*97`
CallPickup	Code to retrieve a parked call from the ITSP.	`*88`
Park	Code to park a call on the ITSP.	`*68`

RTP Settings

The following configuration parameters are available on this page.

ITSP Profile *X* – RTP Web Page (*X* = A, B, C, D, E, F, G, H) Parameter Guide

Parameter	Description	Default Setting
ITSP Profile X – RTP (VoiceService.1.VoiceProfile.n.RTP.) for X = A, B, …, H corresponding to n = 1, 2, …, 8, respectively		
LocalPortMin	Base of port range for tx/rx RTP with this SP.	`16600` (X=A) `16800` (X=B) `17000` (X=C) `17200` (X=D) `17400` (X=E) `17600` (X=F) `17800` (X=G) `18000` (X=H)
LocalPortMax	Top of port range for tx/rx RTP with this SP.	`16798` (X=A) `16998` (X=B) `17198` (X=C) `17398` (X=D) `17598` (X=E) `17798` (X=F) `17998` (X=G) `18198` (X=H)
KeepAliveInterval	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`
DSCPMark	Diffserv code for outgoing RTP packets with this SP.	`0`
X_UseSSL	Enables forcing the device to send RTP over an SSL channel.	No
X_RefreshSession	Allow incoming RTP packets to refresh session.	Yes
X_SymmetricMedia	If incoming payload type changes unannounced, after 10 packets with the new payload type, decoding will switch to the new format. If symmetric media is enabled, outgoing packets will also be in the new format.	Yes
ITSP Profile X – RTCP (VoiceService.1.VoiceProfile.n.RTP.RTCP.) for X = A, B, …, H corresponding to n = 1, 2, …, 8, respectively		
Enable	Enables RTCP operation.	No
TxRepeatInterval	RTCP packet transmission interval in milliseconds.	`10000`
LocalCName	The canonical name to use in RTCP messages. If blank, the device uses `<userid>@<local_IP_address>` as its canonical name.	
X_RTCPMux	Enables using an rtcp-mux attribute in SDP (send and receive RTCP on the same port as RTP).	No

ITSP Profile *X* – RTP Web Page (*X* = A, B, C, D, E, F, G, H) Parameter Guide

Parameter	Description	Default Setting
X_VqPublishEnable	Enables VQ report sent to the proxy server using Publish method	false
X_VqPublishUrl	A Username or URL to send Voice Quality Report using Publish method	
X_VqPublishInterval	Interval in seconds between VQ reports; 0 or an empty value disables periodic reports	0
X_VqPublishOnSSRCChange	Enables VQ report when SSRC changes	true
ITSP Profile X - Jitter Buffer (VoiceService.1.VoiceProfile.n.RTP.JIB.) for X = A, B, ..., H corresponding to n=1, 2, ..., 8 respectively		
Adaptive	Enable jitter buffer adaptation.	Yes
MaximumSize	Maximum jitter buffer size in milliseconds.	`250`
SetPoint	Initial playout delay in milliseconds.	`60`
Target	Target playout delay in milliseconds.	`20`
AdaptationSlope	Maximum adaptation slope in samples per 10 milliseconds.	`16`

Voice Services

The Voice Services parameters web pages show parameters for the voice services that are bound to the service providers on your device. They include these sets of pages.

- SPn Service Settings (n = 1, 2, 3, 4, 5, 6, 7, 8)
- OBiTALK Service Settings
- Gateway Settings
- Page Group Settings

SPn Service Settings (n = 1, 2, 3, 4, 5, 6, 7, 8)

The following configuration parameters are available on this page.

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
SPn Service (VoiceService.1.VoiceProfile.1.Line.n.), n = 1 – 8		
Enable	Enables this line.	True
X_ServProvProfile	Select a Service Provider profile for this service. Choices are `A`, `B`, `C`, `D`, `E`, `F`, `G`, or `H`.	`A`

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
X_RingProfile	Selects a Ring Profile to ring the handset 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`
X_CodecProfile	Selects a Codec Profile for all calls on this service. Choices are `A` or `B`.	`A`
X_InboundCallRoute	Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the handset. See the Call Routing chapter for a description of the syntaxes to specify this parameter.	`DT1`
X_RegisterEnable	Enables registration for this line. If set to True, the handset sends periodic SIP REGISTER to the service provider according to the settings in the ITSP Profile. Otherwise, the handset doesn't send any SIP REGISTER for the service.	True
X_AcceptSipFromRegistrarOnly	Accept SIP packets coming from the current registrar IP address only.	False
X_NoRegNoCall	Enables this option to disallow incoming and outgoing calls if registration with the service provider isn't successful.	False
X_KeepAliveEnable	Enables sending keep alive message. If set to True, the handset 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 `keep-alive\r\n`.	False
X_KeepAliveExpires	Keep alive period in seconds.	`15`
X_KeepAliveMsgType	The type of keep alive messages to send out periodically if keep-alive is enabled. Choice of: • `keep-alive`: The string `keep-alive` • `empty`: A blank line • `stun`: A standard STUN binding request. The handset uses the binding response to form its contact address for REGISTRATION • `custom`: use the value of **X_CustomKeepAliveMsg** • `options`: A SIP OPTIONS message • `notify`: A SIP NOTIFY message	`keep-alive`

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	**Description**	**Default Setting**
X_CustomKeepAliveMsg	Defines the custom message to be used when **X_KeepAliveMsgType** is `custom`. The value has the following format: `mtd=NOTIFY;event=<whatever>;user=<anyone>` Where • `NOTIFY` can be replaced by any other SIP method, such as PING • `event` is optional and is only applicable if method is NOTIFY. If event isn't specified, the 'keep-alive' event is used with NOTIFY • `user` is optional. If not speciifed, 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 `user` is specified, it's used as the user ID in the Request-URI and TO header. SIP messages for keep-alive are sent only once without retransmission. Responses to the SIP messages are ignored by the handset.	None
X_UserAgentPort	UDP port where the handset sends and listens for SIP messages.	`5060` (*n*=1) `5061` (*n*=2) `5062` (*n*=3) `5063` (*n*=4) `5064` (*n*=5) `5065` (*n*=6) `5066` (*n*=7) `5067` (*n*=8)
X_UserAgentPorts	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
DirectoryNumber	Directory number associated with this service.	None
X_DefaultRing	Call waiting tone (as specified in the Ring Profiles) to play when there is a second incoming call.	1
X_AcceptResync	Control whether to accept a SIP NOTIFY request with event=resync to trigger a reboot of the handset so it can download new firmware or configuration upon boot up. Choice of: • `no` (don't accept resync trigger) • `yes with authentication` (accept after challenging the sender) • `yes without authentication` (accept without challenging the sender)	`yes without authenticati on`

SPn Service — Debug Options (VoiceService.1.VoiceProfile.1.Line.n.), n = 1 - 8

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
X_SipDebugOption	Enables sending of SIP signaling debug information to the syslog server (if one is configured on the device). Choice of: • `Disable` (do not send SIP signaling debug information) • `Log All Messages` (log all messages) • `Log All Except REGISTER Messages`	`Disable`
X_SipDebugExclusion	Comma-separated list of SIP methods to exclude from the debug log.	None
SPn Service — SIP Credentials (VoiceService.1.VoiceProfile.1.Line.n.SIP.), n = 1 – 8		
AuthUserName	The User ID to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the handset is challenged by the UAS with a 401 or 407 response.	None
AuthPassword	The Password (corresponding to **AuthUserName**) to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the handset is challenged by the UAS with a 401 or 407 response.	None
URI	This parameter affects the way the AOR is formed by the handset in outbound SIP Requests. The AOR has the format: `user@domain`. If the value of **URI** is empty, the handset gets the user portion of its AOR from the **AuthUserName**, and the domain portion the value of ITSP Profile's **UserAgentDomain** if it isn't empty, or that of the **ProxyServer** otherwise. If the value of **URI** isn't empty and doesn't contain "@", it is used as the user portion of the AOR while the domain portion is formed the usual way. If the value of **URI** contains `@`, it is interpreted as a full AOR and handset takes it as the AOR as is. Some examples: • Let **ProxyServer** = `sip.myitsp.com`, **AuthUserName** = `4089991123`, **URI**=[empty], **UserAgentDomain**=[empty], then AOR = `4089991123@sip.myitsp.com` • Change **UserAgentDomain** to `users.myitsp.com`, then AOR = `4089991123@users.myitsp.com` • Change **URI** to `bobdylan`, then AOR = `bobdylan@users.myitsp.com` • Change **URI** to `bobdylan@superusers.myitsp.com`, then AOR = `bobdylan@superusers.myitsp.com` Note: In all cases, the handset uses **AuthUserName** and **AuthUserPassword** to compute authorization if challenged by a 401 or 407 response.	None

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
X_XsiUserName	Username to authenticate a Broadsoft XSI application server. If not specified '`sip-userid@sip-proxy`' is used.	Yes
X_XsiPassword	Password to authenticate a Broadsoft XSI application server. If not specified, the SIP password is used.	Yes
X_ContactUserID	An alternative user ID to be used in Contact header. Enter `Random` to let the handset generate a random one.	None
X_EnforceRequestUserID	Enforce incoming INVITE request user ID to match **AuthUserName** or **X_ContactUserID**.	False
SPn Service — Calling Features (VoiceService.1.VoiceProfile.1.Line.n.CallingFeatures.), n = 1 – 8		
CallerIDName	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
MaxSessions	The maximum number of simultaneous calls that can be established on this service.	`4`
CallForwardUnconditionalEnable	Enables call forwarding of all calls unconditionally by the handset. If **CallForwardUnconditionalNumber** is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the handset using a Star Code.	False
CallForwardUnconditionalNumber	Directory number to forward all incoming calls on this service unconditionally. Maximum Length is 127 characters. Note: You can set this parameter from the handset using a Star Code.	None
CallForwardOnBusyEnable	Enables call forwarding of all incoming calls when the handset is busy. If **CallForwardOnBusyNumber** is blank, this parameter is treated as if it has been set to False. The handset is considered busy if one of the following conditions holds: • This service already reaches the limit of simultaneous calls as specified in **MaxSessions** • DND (Do Not Disturb) Service is enabled on this service if the call is routed to the handset where the handset is in a busy state (such as ringing, dialing, playing reorder, or already having 2 calls in progress) Note: You can set this parameter from the handset using a Star Code.	False
CallForwardOnBusyNumber	Directory number to forward all incoming calls on this service when the handset is busy. Maximum Length is 127 characters. Note: You can set this parameter from the handset using a Star Code.	None

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
CallForwardOnNoAnswerEnable	Enables call forwarding of all incoming calls when the call isn't answered after a period as specified in **CallForwardOnNoAnswerRingCount**. If **CallForwardOnNoAnswerNumber** is blank, this parameter is treated as if it has been set to False. Note: You can set this parameter from the handset using a Star Code.	False
CallForwardOnNoAnswerNumber	Directory number to forward all incoming calls when the call isn't answered after a period specified in **CallForwardNoAnswerRingCount**. Note: You can set this parameter from the handset using a Star Code.	None
CallForwardOnNoAnswerRingCount	Number of rings to be considered by the handset as no answer to an incoming call. Note: 1 ring is approximately 6 seconds.	2
X_BlockedCallers	A comma-separated list of as many as 10 caller numbers to block from calling this service.	None
X_MailboxID	The mailbox ID to subscribe MWI with.	None
X_CheckVoiceMailNumber	The number to call to check voicemail.	None
MWIEnableMask	The set of handsets that are to receive the MWI or VMWI notifications.	
X_VMWIEnableMase	It is a bit mask. Each bit represents a handset. So 1023 (0x3ff) represents all 10 handsets. 4 (0x4) represents handset 3.	
X_MWIRoute	SIP/NOTIFY Routing Rules to enable MWI signals on MWI Notifications.	
X_VMWIRoute	SIP/NOTIFY Routing Rules to enable VMWI signals on MWI Notifications.	
MessageWaiting	This state parameter indicates if there are any new messages for this subscriber on the service provider's voicemail system.	No
MessageCount	Displays count of new messages, in format new/old (urgent – new/urgent – old).	None
AnonymousCallBlockEnable	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: Users can set this parameter from the handset with a Star Code.	No

SP*n* Services (*n* = 1, 2, 3, 4, 5, 6, 7, 8) Settings

Parameter	Description	Default Setting
AnonymousCallEnable	Enables masking Caller-ID information for all outgoing calls. If enabled, the called party sees the call as coming from an anonymous caller. Note: Users can set this parameter from the handset with a Star Code.	No
DoNotDisturbEnable	Enables Do Not Disturb Service. If enabled, all incoming calls on this service are treated as if the device is busy. Note: Users can set this parameter from the handset with a Star Code.	No
X_SRTP	Enables SRTP. Choose one of: • `Disable SRTP` = don't use SRTP for all calls. The call fails if the peer insists on using SRTP only. • `Use SRTP Only` = Require all calls to use SRTP. The call fails if the peer doesn't support SRTP. • `Use SRTP When Possible` = Use SRTP for a call if the peer supports SRTP. Otherwise, fall back to use regular unencrypted SRTP.	`Disable SRTP`
X_ASFeatureEventSubscribe	Enables subscription to the as-feature-event package.	`False`
X_ConferenceBridge	The number of an external conference bridge to use with calls on this service. 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.	`cbridge`

OBiTALK Service Settings

The following configuration parameters are available on this page.

OBiTALK Service Settings Parameter Guide

Parameter	Description	Default Setting
OBiTALK Service Settings (VoiceService.1.X_P2P.1.)		
Enable	Enables the OBiTALK Service (the built-in free voice service that comes with every OBi Device).	Yes
DisplayNumber	The number to display on the handset screen for a line or service.	
LocalPort	The UDP or TCP port used by the device to send and listens for OBiTALK messages.	`10000`

OBiTALK Service Settings Parameter Guide

Parameter	Description	Default Setting
TryMultiplePorts	Enables the device to try a few random UDP ports until it can successfully join the OBiTALK network.	No
Transport	Select the transport to connect the device to OBiTALK.	UDP/TCP
ServerAddress	OBiTALK server IP address (should not be empty in normal operation).	
LastRegistrarAddress	IP address and port number of the last registrar address used.	None
TLSServerPort	OBiTALK TLS server listen port (443 in normal operation).	443
DisplayName	Display name to identify the subscriber, for the purpose of displaying a Caller ID Name on the recipient's device.	
DigitMap	Digit map to restrict numbers that can be dialed or called with this service.	(<ob>xxxxxxxxx|obxxxxxxxxx)
InboundCallRoute	Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the handset(dt).	DT1
RingProfile	Selects a Ring Profile to ring the handset with when an incoming call is routed to the handset.Choose from A or B.	A
CodecProfile	Selects a Codec Profile to be used for all calls on this service. Choose from A or B.	A
DefaultRing	Call waiting tone (as specified in the Ring Profiles) to play when there is a second incoming call.	2
DTMFMethod	Method to pass DTMF digits to peer device. Choose from: • Inband: DTMF tone are sent as inband audio signal • RFC2833: DTMF tone events are relayed per RFC2833 • SIPInfo: DTMF tones are relayed with SIP INFO request • Auto: Method to use based on call setup negotiation (either Inband or RFC2833 can be negotiated)	Auto
FixedDurationRFC2833DTMF	The fixed duration (in groups of 10 milliseconds) to use when **UsedFixedDurationRFC2833DTMF** is True.	16
SymmetricMedia	If incoming payload type changes unannounced, after ten packets with the new payload type, decoding will switch to the new format. If symmetric media is enabled, outgoing packets also uses the new format.	Yes

OBiTALK Calling Features Parameter Guide

Parameter	Description	Default Setting
OBiTALK Service – Calling Features (VoiceService.1.X_P2P.1.CallingFeatures.)		
CallForwardUnconditionalEnable	Enables call forwarding of all calls unconditionally by the device. If **CallForwardUnconditionalNumber** is blank, this parameter is treated as if it has been set to *No*. Note: Users can set this parameter from the handset with a Star Code.	No
CallForwardUnconditionalNumber	Directory number to forward all incoming calls on this service unconditionally. Maximum length is 127 characters. Note: Users can set this parameter from the handset with a Star Code.	
CallForwardOnBusyEnable	Enables call forwarding of all incoming calls when the device is busy. If ***CallForwardOnBusyNumber*** is blank, this parameter is treated as if it has been set to *No*. Device is considered busy if one of the following conditions holds: • This service already reaches the limit of simultaneous calls as specified in ***MaxSessions*** • DND (Do Not Disturb) Service is enabled on this service If the call is routed to the handset port where the handset is in a busy state (such as ringing, dialing, playing reorder, or already having 2 calls in progress) Note: Users can set this parameter from the handset with a Star Code.	No
CallForwardOnBusyNumber	Directory number to forward all incoming calls on this service when the device is busy. Maximum length is 127 characters. Note: Users can set this parameter from the handset with a Star Code.	
CallForwardOnNoAnswerEnable	Enables call forwarding of all incoming calls when the call isn't answered after a period as specified in ***CallForwardOnNoAnswerRingCount***. If ***CallForwardOnNoAnswerNumber*** is blank, this parameter is treated as if it has been set to No. Note: Users can set this parameter from the handset with a Star Code.	No
CallForwardOnNoAnswerNumber	Directory number to forward all incoming calls when the call isn't answered after a period specified in ***CallForwardNoAnswerRingCount***. Note: Users can set this parameter from the phandset with a Star Code.	
CallForwardOnNoAnswerRingCount	Number of rings to be considered by the device as no answer to an incoming call. Note: 1 ring is approximately 6 seconds.	2

OBiTALK Calling Features Parameter Guide

Parameter	Description	Default Setting
BlockedCallers	A comma-separated list of as many as 10 caller numbers to block from calling this service.	
MaxSessions	The maximum number of simultaneous calls that can be established on this service.	2
AnonymousCallBlockEnable	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: Users can set this parameter from the handset with a Star Code.	No
AnonymousCallEnable	Enables masking Caller-ID information for all outgoing calls. If enabled, the called party sees the call as coming from an anonymous caller. Note: Users can set this parameter from the handset with a Star Code.	No
DoNotDisturbEnable	Enables Do Not Disturb Service. If enabled, all incoming calls on this service are treated as if the device is busy. Note: Users can set this parameter from the handset with a Star Code.	No
Jitter Buffer (VoiceService.1.X_P2P.1.JIB.)		
Adaptive	Enables jitter buffer adaptation.	True
MaximumSize	Maximum jitter buffer size in ms.	250
SetPoint	Initial play out delay in ms.	60
Target	Target play out delay in ms.	20
AdaptationSlope	Maximum adaptation slope in samples per 10 ms.	16

Gateway Settings

The following configuration parameters are available on this page.

Gateways Parameter Guide

Parameter	Description	Default Setting
Voice Gateway n (VoiceService.1.X_VoiceGateway.n.) for n = 1 - 8		
Enable	Enables this voice gateway.	Yes
Name	An arbitrary user-friendly name to identify this gateway (optional).	

Gateways Parameter Guide

Parameter	Description	Default Setting
AccessNumber	The gateway's OBiTALK number, including trunk information, such as: `PP(ob200112334)` or `PP(ob200112334)` If the value is blank, the device treats this VG as disabled. Starting with release 1.2, this can also be set to a SIP URL, such as: `SP1(sip.mycompany.com:5060)` or `SP2(192.168.15.113)`	
DigitMap	**DigitMap** for this VG. It can be referenced as (Mvg*n*).	`(x.x)`
AuthUserID	A user ID to authenticate with the gateway.	
AuthPassword	A password to authenticate with the gateway.	

Page Group Settings

The following configuration parameters are available on this page.

Page Groups Parameter Guide

Parameter	Description	Default Setting
Page Group 1 or 2 (VoiceService.1.X_PageGroup.n.), n = 1, 2		
GroupName	A user friendly name to label the group on the handset user interface.	None
MulticastAddress	Must be a valid IPv4 Multicast Address.	`224.1.1.100`
MulticastPort	Port to use for multicast.	`65322`
TTL	TTL value of outgoing (multicast) RTP packets.	`2`
ParticipantName	Name to identify this participant to the group.	None
AudioCodec	Audio codec to use for outgoing page.	`G711U`
TxPacketSize	RTP transmission packet size (in ms).	`20`
RTCPTxInterval	RTCP transmission interval (in ms) when talking.	`0`
SilenceSuppression	Enable silence suppression when talking.	False
PlayToneOnIncomingPage	Play a short paging tone on receiving an incoming page.	True
StartTalkingOnJoin	Start talking immediately when joining the group.	Ture
TalkingAlertTone	A short call waiting tone plays periodically to remind you the device is in talking mode.	`CWT10`
SwitchToTalkModeDigit	Digit to switch from listening mode to talking mode.	`*`
SwitchToListenModeDigit	Digit to switch from talking mode to listening mode.	`#`

DECT Wireless

The DECT Wireless web page shows parameters for the wireless handset. They include these sets of pages.

- System Settings
- Registration Settings
- Handsetn (n = 1, 2, ..., 9, 10) Settings

System Settings

The following configuration parameters are available on this page.

System Parameter Guide

Parameter	Description	Default Setting
System - DECT Base Information (VoiceService.1.X_HS.)		
TargetVersion	Firmware version.	
RFPI	Radio fixed part identity of the base.	
DectType	Displays the DECT device region.	
BaseName	Name of the base unit.	
TargetFW	Target firmware file to use. For debugging use only.	
TargetFWUpgrade	Enables target firmware upgrade.	Yes
TargetFWDowngradeAllowed	Allow the DECT target firmware to be downgraded when the full firmware bundle is an older version.	No
IntercomEnable	Enables handset intercom function.	Yes
Handset Status		
HandsetnStatus, n=1-10	Displays the handset status.	
Handset Information		
HandsetnType, n=1-10	Displays the handset type.	
HandsetnIPEI, n=1-10	Displays the handset's International Portable Equipment Identity (IPEI).	
Handset Firmware		
HandsetnFWVersion, n=1-10	Displays the handset firmware version.	
HandsetFW	Handset firmware file to use. For debugging use only.	
HandsetFWDowngradeAllowed	Allow handset firmware to be downgraded when the full firmware bundle is an older version.	No
HandsetnFWUpgrade, n=1-10	Enable handset firmware upgrade.	Yes

System Parameter Guide

Parameter	Description	Default Setting
Handset Locator		
FindHandsetAll	Locate all handsets. Equivalent to pressing the "FIND" button on the unit.	
FindHandsetn, n=1-10	Locate individual handset(s).	

Registration Settings

The following configuration parameters are available on this page.

Registration Parameter Guide

Parameter	Description	Default Setting
Registration - DECT Handset Registration (VoiceService.1.X_HS.)		
RegistrationWindow	Registration window status.	
OpenRegistration	Opens handset registration.	
RegisteredHandsets	Displays a list of registered handsets.	
DeleteHandsetAll	Deletes all registered handsets.	
DeleteHandsetn, n=1-10	Deletes the selected handset.	

Handsetn (n = 1, 2, ..., 9, 10) Settings

The following configuration parameters are available on this page.

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
Handsetn - Settings (VoiceService.1.X_HS.n.), n = 1, 2, ..., 9, 10		
Enable	Enables the handset.	Yes
Name	Sets the name of the handset.	

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

<table>
<tr><th>Parameter</th><th>Description</th><th>Default Setting</th></tr>
<tr><td>DigitMap</td><td>Restricts the numbers that can be dialed or called from the handset. If the caller dials a number that isn’t allowed by the digit map, the device plays a SIT tone followed by a short error message to let the caller know that the dialed number is invalid.</td><td>([1-9]x?*(Mpli)|[1-9]S9|[1-9][0-9]S9|911|[67]XX|**0|***|#S4|#[0-8]|#9x|**81(Mbt)|**82(Mbt2)|**1(Msp1)|**2(Msp2)|**3(Msp3)|**4(Msp4)|**9(Mpp)|(Mpli))</td></tr>
<tr><td>OutboundCallRoute</td><td>After the caller dials a number that is acceptable according to the DigitMap, the device uses this outbound call routing rule to determine that service to make this call with. If no appropriate call route is found, the device plays a SIT tone followed by a short error message to let the caller know that there is no call route to place the call.</td><td>{([1-9]x?*(Mpli)):pp},{#0:ao},{#1:dt1},{#2:dt2},{#3:dt3},{#4:dt4},{#5:dt5},{#6:dt6},{#7:dt7},{#8:dt8},{#*:dt1,dt2,dt3,dt4,dt5,dt6,dt7,dt8},{(<6:park>XX<;s=1>):pk},{(<7:pickup>XX<;d=0>):pk},{(<**82:>(Mbt2)):bt2},{(<**81:>(Mbt)):bt},{**0:aa},{***:aa2},{(<**1:>(Msp1)):sp1},{(<**2:>(Msp2)):sp2},{(<**3:>(Msp3)):sp3},{(<**4:>(Msp4)):sp4},{(<**9:>(Mpp)):pp},{(Mpli):pli}</td></tr>
<tr><td>OutboundServices</td><td>List of services available for dialing out.</td><td>sp1,pp</td></tr>
</table>

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
CallReturnDigitMaps	Call Return is the service where you can call the last caller by dialing a star code (*69 by default). The device implements this service by remembering the number of the last caller in memory. However, the stored information doesn’t include any dialing prefix to tell the device which voice service to use to call back the last caller. This list of digit maps serve the purpose of mapping a caller’s number to one that includes the desired dialing prefix used exclusively for call return service.	`{pli:(xx.)},{sp1:(<**1>xx.)},{sp2:(<**2>xx.)},{sp3:(<**3>xx.)},{sp4:(<**4>xx.)},{bt1:(<**81>xx.)},{bt2:(<**82>xx.)},{pp:(<**9>xx.)}`
PrimaryLine	The “primary line” is the service that doesn’t 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 device which voice service is considered as the primary line when dialing out from the handset. Choose from: • SP1 Service (code = sp1) • SP2 Service (code = sp2) • SP3 Service (code = sp3) • SP4 Service (code = sp4) • SP5 Service (code = sp5) • SP6 Service (code = sp6) • SP7 Service (code = sp7) • SP8 Service (code = sp8) • OBiTALK Service (code = pp1) The device process the parameter by substituting of the occurrences of pli and (Mpli) in **DigitMap, OutboundCallRoute**, and **CallReturnDigitMaps** with the corresponding code and (Mcode). For example, if **PrimaryLine** = sp3, then all occurrences of pli and (Mpli) are substituted internally with sp3and (Msp3).	`SP1 Service`

Handsetn - Calling Features (VoiceService.1.X_HS.n.CallingFeatures.), n = 1, 2, ..., 9, 10

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
CallerIDEnable	Enables Caller ID Signal generation. This option can be set to Yes even if the attached handset isn't capable of displaying Caller ID. There is no harm in sending Caller ID signal while the handset is in the on hook state.	Yes
CallWaitingCallerIDEnable	Enables Call Waiting Caller ID (CWCID) Signal generation. The CWCID signal is sent to the handset when it is in the off hook state. It starts with a handshake between the device and the attached handset, by exchanging audible short tones. The device proceeds with the transmission of the remaining Caller ID signal only if the handshake succeeds (with a handset is capable of displaying CWCID). In that case the handset mutes the handset earpiece until the CWCID signal is complete. Some users however may still find the audible handshake tones objectionable, especially if their handsets don't support CWCID. Set this option to No if you don't want the CWCID feature, or don't have handsets that can display CWCID.	Yes
MWIEnable	Enables MWI Signal (stutter dial tone) generation. If enabled, any SP voice service enabled on the device 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.	Yes
VMWIEnable	Enables VMWI Signal generation. If enabled, any SP voice service enabled on the device 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.	Yes

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
CallTransferEnable	Enables Call Transfer. If enabled, you initiate Call Transfer by hanging up the handset in one of the following scenarios: • One call on hold while a second • outgoing call ringing (Case 1) • One call on hold while a second • outgoing call connected (Case 2) • One call connected while a second • outgoing call ringing (Case 3) • 3-way conference with both calls • connected (Case 4) If Call Transfer is disabled, hanging up the handset in the above scenarios ends all the calls except for the one that is holding, which remains on hold (Cases 1 and 2).	Yes
ConferenceCallEnable	Enables 3-way Conference Call w/ local audio mixing. If enabled, you initiate Conference Call by hook flashing the handset in one of the following scenarios: • One call on hold while a second • outgoing call ringing (Case 1) • One call on hold while a second • outgoing call connected (Case 2) Case 1 is an early conference, where the second conferencee is still ringing. The other two parties may converse while hearing ringback tone in the back-ground until the third party answers. In either case, you can end the call with the second conferencee by hook flashing another time and the call reverts to a 2-way call. If Conference Call service is disabled, then hook flashing the handset resumes the holding call but ends the second outgoing call in Case 1, and swaps between the two calls in Case 2 (as in a call waiting situation).	Yes

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
UseExternalConferenceBridge	Enables using an external conference bridge for conference calls (SIP only). In addition, the following rule `{cbridge:SPx(bridge-userid)}` must also be added to the handset port's **OutboundCallRoute** parameter, where x=1,2,3,4, and bridge-userid the userid of the conference bridge SUA. Note that the keyword cbridge is hard-coded and must not be changed.	No
CallWaitingEnable	Enables call waiting service. Call Waiting is the situation where a new incoming call is routed to the handset port when there is already another call connected. If this service is enabled, the device plays the call-waiting tone to alert you, as well as generates the CWCID signal if CWCID is enabled. You can then swap between the two calls by hook flashing. If the service is disabled, the device rejects the incoming call as busy. Note: Users can set this parameter from the handset with a Star Code.	Yes
ToneProfile	Selects a Tone Profile for call progress tone generation.Choose from `A` or `B`.	`A`
StarCodeProfile	Selects a Star Code Profile for interpreting Star Codes you enter. Choose from: • `None` • `A` • `B` If set to `None`, no star code is recognized by the device.	`A`
LastDialedNumber	Last number dialed out on the handset.	
LastCallerNumber	Last caller’s number that rings the handset.	
RepeatDialInterval	Interval in seconds between retry in a repeat dial operation.	`30`
RepeatDialExpires	Duration of time in seconds when a repeat dial operation remains active.	`1800`

Handsetn Settings Parameter Guide (n = 1, 2, ..., 9, 10)

Parameter	Description	Default Setting
MOHServiceNumber	The number to call to get music streamed to the remote party when the remote party is placed on hold.	
PlaySITOnCallFailureCodes	A list of (3-digit) error respons codes on outbound calls to trigger SIT with optional announcement of the error. The device plays fast busy tone without any announcement for all other call failure codes. The codes must be specified collectively as a digit map.	`([4-9]xx)`
PlaySITWithAnnouncement	Enables including announcement of the error when an outbound call fails.	Yes
Timers (VoiceService.1.X_HS.n.Timer.), n = 1, 2, ..., 9, 10		
ReorderDelayTime	Delay (in ms) to start reorder tone after peer ends call.	`5500`
DigitMapLongTimer	Default number (in seconds) when the digit map processor should timeout waiting for more digits for matching patterns with an unspecified length.	`10`
DigitMapShortTimer	Default number (in seconds) when the digit map processor should timeout waiting for more digits when at least one pattern with specific length has matched.	`2`

Codec Profiles

The Codec Profiles parameters web pages include one page for each codec profile.

Codec Profile X Web Page (X = A, B)

The following configuration parameters are available on each page.

Codec Profile X Web Page (X = A, B)

Parameter	Description	Default Setting
Codec Profile X – G711U Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.1.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`PCMU`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`64000`

Codec Profile X Web Page (X = A, B)

Parameter	Description	Default Setting
Enable	Enables this codec.	Yes
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`20`
Priority	Priority assigned to this codec (1 is the highest).	`3`
PayloadType	Standard payload type for this codec.	`0`
Codec Profile X – G711A Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.2.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`PCMA`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`64000`
Enable	Enables this codec.	Yes
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`20`
Priority	Priority assigned to this codec (1 is the highest).	`4`
PayloadType	Standard payload type for G711-alaw. Note: Informational only, not configurable.	`8`
Codec Profile X – G729 Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.3.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`G729`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`8000`
Enable	Enables this codec.	Yes
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`20`
Priority	Priority assigned to this codec (1 is the highest).	`5`
PayloadType	Standard payload type for G729. Note: Informational only, not configurable.	`18`
Codec Profile X – G726R32 Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.4.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`G726-32`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`32000`

Codec Profile X Web Page (X = A, B)

Parameter	Description	Default Setting
Enable	Enables this codec.	Yes
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`20`
Priority	Priority assigned to this codec (`1` is the highest).	`7`
PayloadType	Dynamic Payload type for this codec. Valid range is 96–127.	`104`
Codec Profile X – iLBC Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.8.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`iLbc`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`13333`
Enable	Enables this codec.	No
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`30`
Priority	Priority assigned to this codec (`1` is the highest).	`6`
PayloadType	Dynamic Payload type for this codec. Valid range is 96–127.	`98`
Codec Profile X – *G722 Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.9.) n = 1 or 2 corresponding to X = A or B, respectively*		
Codec	Codec name.	`G722`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`64000`
Enable	Enables this codec.	Yes
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	`20`
Priority	Priority assigned to this codec (`1` is the highest).	`1`
PayloadType	Dynamic Payload type for this codec.	`9`
Codec Profile X – OPUS Codec (VoiceService.1.VoiceProfile.1.Line.n.Codec.List.10) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name.	`OPUS`
BitRate	Bit rate in bits/sec. Note: Informational only, not configurable.	`20000`
Enable	Enables this codec.	Yes

Codec Profile X Web Page (X = A, B)

Parameter	Description	Default Setting
SilenceSuppression	Enables silence suppression for this codec.	No
PacketizationPeriod	Packet size in ms.	20
Priority	Priority assigned to this codec (`1` is the highest).	`2`
PayloadType	Dynamic Payload type to be used to indicate this event.	`109`
UseInbandFEC	Enables use in band FEC when appropriate.	No
Codec Profile X – Telephone Event (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_TelephoneEvent.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec Name for this RTP event, as used in SDP.	`telephone-event`
Enable	Enables this codec.	Yes
PayloadType	Payload type to be used for RFC2833 telephone (DTMF) events. Valid range is 96–127.	`101`
Codec Profile X – Encap RTP (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_EncapRTP.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec Name. This codec is used to encapsulate RTP packets during a packet loopback call.	`encaprtp`
PayloadType	Dynamic Payload type for this codec. Valid range is 96–127.	`107`
Codec Profile X – Loopback Primer (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_LoopbackPrimer.) n = 1 or 2 corresponding to X = A or B, respectively		
Codec	Codec name. The device 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`
PayloadType	Dynamic Payload type for this codec. Valid range is 96–127.	`108`
Codec Profile X – Codec Settings (VoiceService.1.VoiceProfile.1.Line.n.Codec.X_Settings.) n = 1 or 2 corresponding to X = A or B, respectively		
G726BitPacking	Two values to choose from: • `big-endian` • `little-endian`	`big-endian`

Tone Settings

The Tone Profile parameters web pages include one page fore each tone profile.

Tone Profile X Web Page (X = A, B)

The following configuration parameters are available on each page.

Tone Profile A & B Parameter Guide

Parameter	Description	Default Setting
Tone Profile X – Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.1.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Dial Tone.	`Dial Tone`
TonePattern	Poly Tone Pattern Script.	`350-18,440-18;20`
Tone Profile X – Ringback Tone (VoiceService.1.VoiceProfile.n.Tone.Description.2.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Ringback Tone.	`Ringback Tone`
TonePattern	Poly Tone Pattern Script.	`440-18,480-18;-1;(2+4)`
Tone Profile X – Busy Tone (VoiceService.1.VoiceProfile.n.Tone.Description.3.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Busy Tone.	`Busy Tone`
TonePattern	Poly Tone Pattern Script.	`480-18,620-18;10;(.5+.5)`
Tone Profile X – Reorder Tone (VoiceService.1.VoiceProfile.n.Tone.Description.4.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Reorder tone or Fastbusy Tone.	`Reorder or Fastbusy Tone`
TonePattern	Poly Tone Pattern Script.	`480-18,620-18;10;(.25+.25)`
Tone Profile X – Confirmation Tone (VoiceService.1.VoiceProfile.n.Tone.Description.5.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Confirmation Tone.	`Confirmation Tone`
TonePattern	Obihai Tone Pattern Script.	`600-18;1;(.2+.2)`
Tone Profile X – Holding Tone (VoiceService.1.VoiceProfile.n.Tone.Description.6.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Holding Tone played when peer holding the call.	`Holding Tone`
TonePattern	Poly Tone Pattern Script.	`800-18;30;(.1+10)`
Tone Profile X – Second Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.7.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Second Dial Tone played when dialing second call in a 3-way call.	`Second Dial Tone`
TonePattern	Poly Tone Pattern Script.	`385-18,484-18;20` (*n* = 1) `400-18,425-18;20` (*n* = 2)

Tone Profile A & B Parameter Guide

Parameter	Description	Default Setting
Tone Profile X – Stutter Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.8.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Stutter Dial Tone.	`Stutter Dial Tone`
TonePattern	Poly Tone Pattern Script.	`350-18,440-18;20;2(.1+.1);()` (*n* = 1) `400-18,425-18,450-18;20;2(.1+.04);()` (*n* = 2)
Tone Profile X – Howling Tone (VoiceService.1.VoiceProfile.n.Tone.Description.9.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Howling Tone for off-hook warning.	`Howling Tone`
TonePattern	Poly Tone Pattern Script.	`480+3,620+3;10;(.125+.125)`
Tone Profile X – Prompt Tone (VoiceService.1.VoiceProfile.n.Tone.Description.10.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Prompt Tone to prompt user to enter a number for configuration, such as speed dial.	`Prompt Tone`
TonePattern	Poly Tone Pattern Script.	`480-16;20`
Tone Profile X – Call Forwarded Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.11.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Call Forwarded Dial Tone: A special dial tone to indicate call-forward-all is active.	`Call Forwarded Dial Tone`
TonePattern	Poly Tone Pattern Script.	`350-18,440-18;20;(.2+.2)`
Tone Profile X – Conference Tone (VoiceService.1.VoiceProfile.n.Tone.Description.12.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Conference Tone (indicates a 3-way conference call has started).	`Conference Tone`
TonePattern	Poly Tone Pattern Script.	`350-16;10;(.1+.1,.1+9.7)` (*n* = 1) `425-16;10;(1+15,.36+15)` (*n* = 2)
Tone Profile X – SIT Tone 1 (VoiceService.1.VoiceProfile.n.Tone.Description.13.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Special Information Tone 1.	`SIT Tone 1`
TonePattern	Poly Tone Pattern Script.	`985-16,1428-16,1777-16;20;(1/.380+0, 2/.380+0,4/.380+0,0/0+4)` (*n* = 1) `425-16;20;(2.5+.5)(n = 2)`
Tone Profile X – SIT Tone 2 (VoiceService.1.VoiceProfile.n.Tone.Description.14.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Special Information Tone 2.	`SIT Tone 2`

Tone Profile A & B Parameter Guide

Parameter	Description	Default Setting
TonePattern	Poly Tone Pattern Script.	`914-16,1371-16,1777-16;20;(1/.274+0, 2/.274+0,4/.380+0,0/0+4)`
Tone Profile X – SIT Tone 3 (VoiceService.1.VoiceProfile.n.Tone.Description.15.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Special Information Tone 3.	`SIT Tone 3`
TonePattern	Poly Tone Pattern Script.	`914-16,1371-16,1777-16;20;(1/.380+0, 2/.380+0,4/.380+0,0/0+4)`
Tone Profile X – SIT Tone 4 (VoiceService.1.VoiceProfile.n.Tone.Description.16.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Special Information Tone 4.	`SIT Tone 4`
TonePattern	Poly Tone Pattern Script.	`985-16,1371-16,1777-16;20;(1/.380+0, 2/.380+0,4/.380+0,0/0+4)`
Tone Profile X – Outside Dial Tone (VoiceService.1.VoiceProfile.n.Tone.Description.17.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Outside Dial Tone.	`Outside Dial Tone`
TonePattern	Obihai Tone Pattern Script.	`385-16;10`
Tone Profile X – R-Command Tone (VoiceService.1.VoiceProfile.n.Tone.Description.18.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	R-Command Tone.	`R-Command Tone`
TonePattern	Obihai Tone Pattern Script.	`400-16;5`
Tone Profile X – Paging Tone (VoiceService.1.VoiceProfile.n.Tone.Description.19.) for n = 1 or 2 corresponding to X = A or B, respectively		
ToneName	Paging Tone.	`Paging Tone`
TonePattern	Obihai Tone Pattern Script.	`480-16;1;(.2+.2)`

Ring Settings

The Ring Profile parameters web pages include one page for each ring profile.

Ring Profile X Web Page (X = A, B)

The following configuration parameters are available on each page.

Ring Profile A & B Parameter Guide

Parameter	Description	Default Setting
Ring Profile X – Call Waiting Tone 1 (VoiceService.1.VoiceProfile.1.Tone.Description.21.)		
ToneName	Distinctive Call Waiting Tone 1.	`Bellcore-dr1`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.25+10)` (*n* = 1) `425-18;30;(.2+.2,.2+4.4)` (*n* = 2)
Ring Profile X – Call Waiting Tone 2 (VoiceService.1.VoiceProfile.1.Tone.Description.22.)		
ToneName	Distinctive Call Waiting Tone 2.	`Bellcore-dr2`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.1+.1,.3+.1,.1+10)`
Ring Profile X – Call Waiting Tone 3 (VoiceService.1.VoiceProfile.1.Tone.Description.23.)		
ToneName	Distinctive Call Waiting Tone 3.	`Bellcore-dr3`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.1+.1,.1+10)`
Ring Profile X – Call Waiting Tone 4 (VoiceService.1.VoiceProfile.1.Tone.Description.24.)		
ToneName	Distinctive Call Waiting Tone 4.	`Bellcore-dr4`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.1+.1,.1+.1,.1+10)`
Ring Profile X – Call Waiting Tone 5 (VoiceService.1.VoiceProfile.1.Tone.Description.25.)		
ToneName	Distinctive Call Waiting Tone 5.	`Bellcore-dr5`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.3+.1,.1+.1,.3+10)`
Ring Profile X – Call Waiting Tone 6 (VoiceService.1.VoiceProfile.1.Tone.Description.26.)		
ToneName	Distinctive Call Waiting Tone 6.	`User-dr1`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.1+.1,.3+.2,.3+10)`
Ring Profile X – Call Waiting Tone 7 (VoiceService.1.VoiceProfile.1.Tone.Description.27.)		
ToneName	Distinctive Call Waiting Tone 7.	`User-dr2`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.3+.1,.3+.1,.1+10)`
Ring Profile X – Call Waiting Tone 8 (VoiceService.1.VoiceProfile.1.Tone.Description.28.)		
ToneName	Distinctive Call Waiting Tone 8.	`User-dr3`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.3+2)`
Ring Profile X – Call Waiting Tone 9 (VoiceService.1.VoiceProfile.1.Tone.Description.29.)		
ToneName	Distinctive Call Waiting Tone 9.	`User-dr4`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.3+2)`
Ring Profile X – Call Waiting Tone 10 (VoiceService.1.VoiceProfile.1.Tone.Description.30.)		

Ring Profile A & B Parameter Guide

Parameter	Description	Default Setting
ToneName	Distinctive Call Waiting Tone 10.	`User-dr5`
TonePattern	Obihai Tone Pattern Script.	`440-18;30;(.3+2)`

Star Codes

The Star Codes parameters web pages include one page for each star code profile.

Star Code Profile X Web Page (X = A, B)

The following configuration parameters are available on each page.

Star Code Profile Parameter Guide

Parameter	Description	Default Setting
Star Code Profile X – Star Codes (VoiceService.1.X_StarCode.n.) for n = 1, 2,..8 corresponding to X = A, B, …, H		
Code1	Default = Redial Star Code	`*07, Redial, call($Ldn)`
Code2	Default = Call Return Star Code	`*69, Call Return, call($Lcn)`
Code3	Default = Block Caller ID (Persistent) Star Code	`*81, Block Caller ID, set($Bci,1)`
Code4	Default = Unblock Caller ID (Persistent) Star Code	`*82, Unblock Caller ID, set($Bci,0)`
Code5	Default = Block Caller ID Once Star Code	`*67, Block Caller ID Once, set($Bci1,1)`
Code6	Default = Unblock Caller ID Once Star Code	`*68, Unblock Caller ID Once, set($Ubci1,1)`
Code7	Default = Call Forward Unconditional Star Code	`*72, Cfwd All, coll($Cfan), set($Cfa,1)`
Code8	Default = Disable Call Forward Unconditional Star Code	`*73, Disable Cfwd All, set($Cfa, 0)`
Code9	Default = Call Forward on Busy Star Code	`*60, Cfwd Busy, coll($Cfbn), set($Cfb,1)`
Code10	Default = Disable Call Forward on Busy Star Code	`*61, Disable Cfwd Busy, set($Cfb, 0)`
Code11	Default = Call Forward on No Answer Star Code	`*62, Cfwd No Ans, coll($Cfnn), set($Cfn,1)`
Code12	Default = Disable Call Forward on No Answer Star Code	`*63, Disable Cfwd No Ans, set($Cfn,0)`

Star Code Profile Parameter Guide

Parameter	Description	Default Setting
Code13	Default = Block Anonymous Calls Star Code	`*77, Block Anonymous Call, set($Bac,1)`
Code14	Default = Unblock Anonymous Calls Star Code	`*87, Unblock Anonymous Call, set($Bac,0)`
Code15	Default = Enable Call Waiting Star Code	`*56, Enable Call Waiting, set($Cwa,1)`
Code16	Default = Disable Call Waiting Star Code	`*57, Disable Call Waiting, set($Cwa,0)`
Code17	Default = Do Not Disturb Star Code	`*78, Do Not Disturb, set($Dnd,1)`
Code18	Default = Disable Do Not Disturb Star Code	`*79, Disable DND, set($Dnd,0)`
Code19	Default = Repeat Dial Star Code	`*66, Repeat Dial, rpdi($Ldn)`
Code20	Default = Disable Repeat Dial Star Code	`*86, Cancel Repeat Dial, rpdi()`
Code21	Default = Set Speed Dial Star Code	`*74([1-9]|[1-9]x), Set Speed Dial, coll($Spd[$Code])`
Code22	Default = Check Speed Dial Star Code	`*75([1-9]|[1-9]x), Check Speed Dial, say($Spd[$Code])`
Code23	Default = Loopback Media Star Code	`*03, Loopback Media, set($Lbm1,1)`
Code24	Default = Loopback RTP Star Code	`*04, Loopback RTP Packet, set($Lbp1,1)`
Code25	Default = Force G711u Codec Star Code	`*4711, Use G711 Only, set($Cdm1,3)`
Code26	Default = Force G729 Codec Star Code	`*4729, Use G729 Only, set($Cdm1,4)`
Code27	Default = Clear Speed Dial Star Code	`*76([1-9]|[1-9]x), Clear Speed Dial, set($Spd[$Code],)`
Code28	Default = Blind Transfer Star Code	`*98, Blind Transfer, coll($Bxrn)`
Code29	Default = Barge In Star Code	`*96, Barge In, set($Bar1,1)`
Code30		
Code31		
Code32	Default = Force G722 Codec Star Code	`*4722, Use G722 Only, set($Cdm1,512)`
Code33	Default = Force OPUS Codec Star Code	`*4678, Use OPUS Only, set($Cdm1,1024)`

User Settings

The User Settings parameters web pages include the following pages:

Speed Dial Settings

The following configuration parameters are available on this page.

Speed Dial Parameter Guide

Parameter	Description	Default Setting
Speed Dials (SpeedDial.)		
1	Enter the number or the URL for Speed Dial 1 with an associated name.	
2	Enter the number or the URL for Speed Dial 2 with an associated name.	
3	Enter the number or the URL for Speed Dial 3 with an associated name.	
...	...	
99	Enter the number or the URL for Speed Dial 99 with an associated name.	

User-Defined Digit Maps Settings

For an explanation of this feature, refer to the User-Defined Digit Maps section in the Device Settings chapter.

User-Defined Digit Maps Parameter Guide

<table>
<tr><th>Parameter</th><th>Description</th><th>Default Setting</th></tr>
<tr><td colspan="3">User-Defined Digit Map 1 (UserDigitMap.1.)</td></tr>
<tr><td>Label</td><td>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.</td><td>ipd</td></tr>
<tr><td>DigitMap</td><td>A valid digit map.</td><td>(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.</td></tr>
<tr><td colspan="3">User-Defined Digit Map N (N = 2 to 10)</td></tr>
</table>

User-Defined Digit Maps Parameter Guide

Parameter	Description	Default Setting
Label	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.	
DigitMap	A valid digit map.	