



Poly ATA 400 Series Parameter Reference Guide

SUMMARY

This guide provides administrators with information about the parameters and configuration options for the featured product.

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Table of contents

| | |
|--|-----------|
| 1 About this guide..... | 1 |
| Audience, purpose, and required skills | 1 |
| Icons used in Poly documentation | 1 |
| 2 Getting Started..... | 2 |
| Supported Devices..... | 2 |
| Poly ATA Device Features..... | 2 |
| Key Features..... | 2 |
| Robust Telephony Features | 3 |
| Powerful Call Routing and Voice Service Features..... | 4 |
| LED Description and LED Behaviour | 4 |
| Poly ATA 400 LED Status Indicators..... | 4 |
| Poly ATA 402 LED Status Indicators | 5 |
| Port Connections..... | 6 |
| Poly ATA 400 Port Connections..... | 6 |
| Poly ATA 402 Port Connections | 7 |
| Accessibility Features..... | 9 |
| 3 Parameters..... | 10 |
| SP Services Status..... | 10 |
| Port Status Parameters | 12 |
| WAN Settings Parameters | 13 |
| WiFi Settings Parameters..... | 17 |
| Device Admin Parameters | 19 |
| Auto Provisioning Parameters..... | 30 |
| \$MACRO Expansion Supported by the Device..... | 34 |
| ITSP Profile (General and SP Info Settings) Parameters | 35 |
| ITSP SIP Settings Parameters | 37 |
| ITSP RTP Settings Parameter..... | 47 |
| SPn Services Parameters | 49 |
| PDMS-SP Service Settings Parameters | 56 |
| PDMS-SP Calling Features Parameters..... | 58 |
| PDMS-SP Inbound Direct Dialing Authentication Parameters | 59 |
| PDMS-SP Jitter Buffer Parameters | 60 |
| User Prompts Parameters..... | 61 |
| Auto Attendant Parameters..... | 61 |
| Auto Attendant Prompt Parameters | 63 |

| | |
|--|------------|
| Voice Gateway Parameters..... | 64 |
| Trunk Group Parameters | 64 |
| Page Group Parameters | 65 |
| PHONE Port Parameters | 66 |
| Tone Profile A Parameters | 75 |
| Tone Profile B Parameters | 80 |
| Codec Profile Parameters | 84 |
| Ring Profile A Call Waiting Tone Parameters..... | 91 |
| Ring Profile A Ring Pattern Parameters..... | 93 |
| Ring Profile B Ring Pattern Parameters..... | 97 |
| Ring Profile B Call Waiting Tone Parameters..... | 100 |
| Star Code Profile Parameters..... | 102 |
| Speed Dial Parameters..... | 106 |
| User-Defined Digit Maps Parameters..... | 107 |
| 4 Getting help..... | 108 |

1 About this guide






This section provides clarifying information about this guide.

Audience, purpose, and required skills

This guide is intended for the following users.

Icons used in Poly documentation

This section describes the icons used in Poly Documentation and what they mean.

-
-  **WARNING!** Indicates a hazardous situation that, if not avoided, **could** result in serious injury or death.
 -  **CAUTION:** Indicates a hazardous situation that, if not avoided, **could** result in minor or moderate injury.
 -  **IMPORTANT:** Indicates information considered important but not hazard-related (for example, messages related to property damage). Warns the user that failure to follow a procedure exactly as described could result in loss of data or in damage to hardware or software. Also contains essential information to explain a concept or to complete a task.
 -  **NOTE:** Contains additional information to emphasize or supplement important points of the main text.
 -  **TIP:** Provides helpful hints for completing a task.
-

2 Getting Started

Understand how to administer, configure, and provision Poly ATA devices.

As you read this guide, keep in mind that certain features are configurable by your system provider, configurable by your system administrator, or determined by your network environment. As a result, some features may not be enabled or may operate differently on your device. Additionally, the examples in this guide may not directly reflect what is available on your device.



NOTE: The terms *the device* and *your device* refer to any of the Poly ATA devices. Unless specifically noted in this guide, all device models operate in similar ways.

Supported Devices

The following table lists the product names, model names, and part numbers for Poly ATA devices.

The Poly ATA 400 series consists of the following device models:

- Poly ATA 400
- Poly ATA 402

Table 2-1 Poly ATA Device Product Name, SKU, and Part Number

| Product Name | SKU | Item Number | Item Description |
|--------------|-------------|-------------|--|
| Poly ATA 400 | NA | 8F3H4AA#ABA | Poly ATA 400 1FXS VP VoIP Adptr SIP US |
| Poly ATA 400 | ANZ, EU, UK | 8F3H4AA#AC3 | Poly ATA 400 1FXS VP VoIP Adptr SIP WW |
| Poly ATA 402 | NA | 8F3H5AA#ABA | Poly ATA 402 2FXS VP VoIP Adptr SIP US |
| Poly ATA 402 | ANZ, EU, UK | 8F3H5AA#AC3 | Poly ATA 402 2FXS VP VoIP Adptr SIP WW |

Poly ATA Device Features

Built with a high-performance, system-on-a-chip platform to ensure high-quality voice conversations, Poly ATA devices are dedicated systems targeted at applications for VoIP services.

Poly ATA devices have high availability and reliability because they're always-on to make or receive calls. If you use a Poly ATA device, you don't need to use a computer, or have a computer turned on, to talk to people. To get started, all you need is a phone, power, and a connection to the internet.

Key Features

Poly ATA devices implement the following features and functionalities.

Table 2-2 Poly ATA Devices

| Poly ATA Device Model | VoIP Account Support | No. of Phone Ports | No. of Ethernet Ports | No. of USB Ports |
|-----------------------|----------------------|--------------------|-----------------------|------------------|
| Poly ATA 400 | 4 | 1 | 1 | 1 |

Table 2-2 Poly ATA Devices (continued)

| Poly ATA Device Model | VoIP Account Support | No. of Phone Ports | No. of Ethernet Ports | No. of USB Ports |
|-----------------------|----------------------|--------------------|-----------------------|------------------|
| Poly ATA 402 | 4 | 2 | 2 | 1 |

The key features of the Poly ATA are:

- SIP Service Provider support for up to four SIP accounts
- Four SIP accounts on Poly ATA 400 and Poly ATA 402
- Any available service is accessible from each **Phone** port independently
- Automatic Attendant (AA) for simplified call routing
- Callback service: automatic callback to connect you to the AA to make a new call or call you back on the attached phone later

Your device is configurable to work with any SIP-compliant internet telephone service (ITSP).

The device supports using the Poly Device Management Service for Service Providers (PDMS-SP) web portal. PDMS-SP is the customer portal for device management allowing administrators to remotely inventory, monitor, and troubleshoot Poly devices.



IMPORTANT: *PDMS-SP* and *OBiTALK* are both terms used in the system web interface and the documentation to refer to the same functionality.

Using the PDMS-SP web portal integration lets you:

- Configure and manage your Poly ATA 400 series devices.
- Upgrade your Poly ATA 400 series devices.
- Troubleshoot and capture additional logs for your Poly ATA 400 series devices.

Robust Telephony Features

Connect an analog phone to one of the **Phone** ports on your device to access a robust set of telephony features.

Poly ATA 400 series devices provide the following telephony features:

- Message waiting indication—visual and tone based
- Speed dialing of 99 Poly endpoints or numbers
- Three-way conference calling with local mixing
- Hook flash event signaling
- Caller ID—name and number
- Call waiting
- Call forward unconditional
- Call forward on busy

- Call forward on no answer
- Call transfer
- Anonymous call
- Block anonymous call
- Do Not Disturb
- Call return
- Repeat dialing

Powerful Call Routing and Voice Service Features

Poly ATAs offer voice service features and call routing.

The Poly ATA 400 series devices give you the following features:

- SIP support for voice and fax over IP from internet telephony service providers (ITSPs)
- PDMS-SP managed VoIP network for Poly endpoint devices and applications
- High-quality voice encoding using G.711, G.722, G.726, G.729, iLBC, and Opus algorithms
- Recursive digit maps and associated call routing for outbound and inbound calls

LED Description and LED Behaviour

Each Poly ATA device has LEDs on the front which give indicators of the device status.

Poly ATA 400 LED Status Indicators

There are three LEDs on the front of the Poly ATA 400 device.

The LEDs give you a visual indication of the working order and general status of key functional aspects of the device. Under normal operating conditions, the LEDs show green (solid, flashing, or blinking) signals.

The following figure displays the LEDs on the front of the Poly ATA 400. The table lists each LED numbered in the figure.

Figure 2-1 Poly ATA 400 LEDs

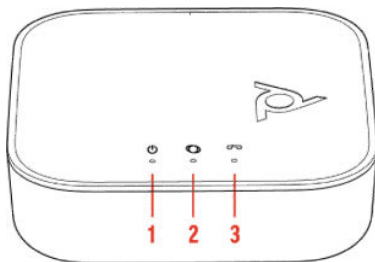


Table 2-3 LED Status Indicators

| Reference Number | LED | LED Description |
|------------------|-----------------------------|---|
| 1 | Power indicator | <p>The color and pattern of the power indicator shows the following states:</p> <p>LED off: No power</p> <p>Solid green: Powered on and working</p> <p>Flashing green: Searching for a DHCP IP address</p> <p>Flashing orange: Upgrading (Do not unplug power.)</p> <p>Solid red: Not operational</p> |
| 2 | LAN Ethernet port indicator | Blinking green: Data activity on the LAN Ethernet port |
| 3 | Phone port indicator | <p>The color and pattern of the phone port indicator shows the following states:</p> <p>LED off: Port not enabled</p> <p>Solid green: Phone ready (standby)</p> <p>Flashing green: Phone in use</p> |

Poly ATA 402 LED Status Indicators

There are five LEDs on the front of the Poly ATA 402 device.

The LEDs give you a visual indication of the working order and general status of key functional aspects of the device. Under normal operating conditions, the LEDs show green (solid, flashing, or blinking) signals.

The following figure displays the LEDs on the front of the Poly ATA 402. The table lists each LED numbered in the figure.

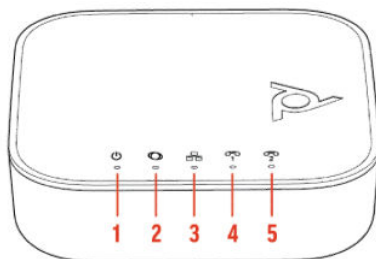
Figure 2-2 Poly ATA 402 LEDs

Table 2-4 LED Status Indicators

| Reference Number | LED | LED Description |
|------------------|--|---|
| 1 | Power indicator | <p>The color and pattern of the power indicator shows the following states:</p> <p>LED off: No power</p> <p>Solid green: Powered on and working</p> <p>Flashing green: Searching for a DHCP IP address</p> <p>Flashing orange: Upgrading (Do not unplug power.)</p> <p>Solid red: Not operational</p> |
| 2 | LAN Ethernet port indicator | Blinking green: Data activity on the LAN Ethernet port |
| 3 | PC Ethernet port indicator | Blinking green: Data activity on the PC Ethernet port |
| 4, 5 | Phone ports indicators: Phone 1, Phone 2 | <p>The color and pattern of the phone ports indicators show the following states:</p> <p>LED off: Port not enabled</p> <p>Solid green: Phone ready (standby)</p> <p>Flashing green: Phone in use</p> |

Port Connections

Learn about the different port connections on the devices in the Poly ATA 400 series.

Poly ATA 400 Port Connections

Become familiar with the physical ports on your Poly ATA 400 device.

The following figures display the ports on the back and side of the Poly ATA 400. The tables list each port numbered in the figure.

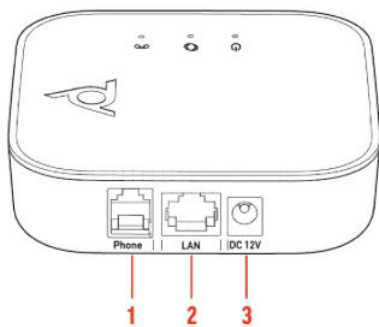
Figure 2-3 Poly ATA 400 Ports

Table 2-5 Port Connections

| Reference Number | Port | Port Description |
|------------------|-----------------------|---|
| 1 | Phone Port connection | <p>The Phone port on the Poly ATA supports input and output signaling and control messages.</p> <p>You can only connect a touch-tone land line phone to your device's Phone port. Phones that use pulse dialing aren't supported.</p> |
| 2 | LAN Ethernet port | <p>Use an Ethernet cable to connect the LAN port on your device to an Ethernet port on your internet router or switch.</p> <p>By default, the device requests an IP address, DNS, and Internet (WAN) Gateway IP addressing via DHCP.</p> |
| 3 | Power connection | <p>Use only the 12-volt power adapter supplied with the original packaging to power the device.</p> <p>The use of a power adapter other than the one given with the device voids the warranty. It might also cause the unit not to work or malfunction.</p> |

Figure 2-4 Poly ATA 400 Side View

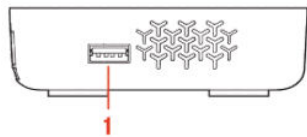


Table 2-6 Port Connections

| Reference Number | Port | Port Description |
|------------------|----------|------------------------------|
| 1 | USB port | Connect an OBiWiFi5G dongle. |

Poly ATA 402 Port Connections

Become familiar with the physical ports on your Poly ATA 402 device.

The following figures display the ports on the back and side of the Poly ATA 402. The tables list each port numbered in the figure.

Figure 2-5 Poly ATA 402 Ports

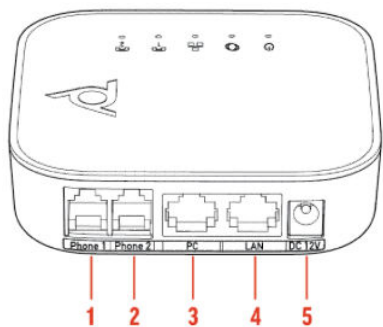


Table 2-7 Port Connections

| Reference Number | Port | Port Description |
|------------------|------------------------|---|
| 1, 2 | Phone Port connections | <p>The Phone ports on the Poly ATA support input and output signaling and control messages.</p> <p>You can only connect a touch-tone land line phone to your device's Phone ports. Phones that use pulse dialing aren't supported.</p> <p>You can connect a second analog phone, or another analog device such as a fax machine or an alarm panel, to the second Phone port.</p> |
| 3 | PC Ethernet port | <p>The device PC port enables you to daisy-chain a local device such as a computer.</p> |
| 4 | LAN Ethernet port | <p>Use an Ethernet cable to connect the LAN port on your device to an Ethernet port on your internet router or switch.</p> <p>By default, the device requests an IP address, DNS, and Internet (WAN) Gateway IP addressing via DHCP.</p> |
| 5 | Power connection | <p>Use only the 12-volt power adapter supplied with the original packaging to power the device.</p> <p>The use of a power adapter other than the one given with the device voids the warranty. It might also cause the unit not to work or malfunction.</p> |

Figure 2-6 Poly ATA 402 Side Callout

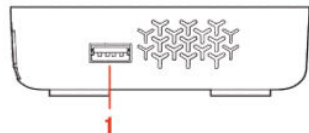


Table 2-8 Port Connections

| Reference Number | Port | Port Description |
|------------------|----------|------------------------------|
| 1 | USB port | Connect an OBiWiFi5G dongle. |

Accessibility Features

Poly products include a number of features to accommodate users with disabilities.

Table 2-9 Accessibility Features

| Accessibility Feature | Description |
|----------------------------|--|
| Hardware status indicators | LED indicators provide status and functionality information on the hardware interface. |

3 Parameters

This section provides information about the parameters and configurations for Poly ATA devices.

SP Services Status

This table lists services, phone, and line status.

Reset Statistics

ResetStatistics

Resets the statistics for this voice service.

Default Value:

N/A

RTP Statistics

PacketsSent

Total RTP packets sent on this line.

Default Value:

N/A

PacketsReceived

Total RTP packets received on this line.

Default Value:

N/A

BytesSent

RTP payload bytes sent for this line.

Default Value:

N/A

BytesReceived

RTP payload bytes received for this line.

Default Value:

N/A

PacketsLost

Number of RTP packets lost on this line.

Default Value:

N/A

Overruns

Number of times receive jitter buffer overrun on this line.

Default Value:

N/A

Underruns

Number of times receive jitter buffer underrun on this line.

Default Value:

N/A

ReceivePacketLossRate

Percentage of audio packet loss

Default Value:

0

ReceiveInterarrivalJitter

Current receive interarrival jitter in milliseconds

Default Value:

0

AverageReceiveInterarrivalJitter

Average receive interarrival jitter in milliseconds

Default Value:

0

SPn Service

SessionStartTime

The time that the session started, in UTC

Default Value:**SessionEndTime**

The time that the session ended, in UTC. 0 implies session is still on-going if SessionStartTime is not 0

Default Value:**SessionDuration**

Duration time of the current session, in seconds

Default Value:

0

FarEndIPAddress

IP address of the far end peer

Default Value:**FarEndUDPPort**

UDP port used by the far end peer

Default Value:

0

LocalUDPPort

Local UDP port for this call

Default Value:

0

PacketsSent

Total RTP packets sent in this session

Default Value:

0

PacketsReceived

Total RTP packets received in this session

Default Value:

0

BytesSent

RTP payload bytes sent in this session

Default Value:

0

BytesReceived

RTP payload bytes received in this session

Default Value:

0

PacketsLost

Number of RTP packets lost in this session

Default Value:

0

PacketsLostRate

RTP packet lost rate as a percentage

Default Value:

0

ReceiveInterarrivalJitter

Current receive interarrival jitter in milliseconds

Default Value:

0

MOS

Mean Opinion Score {1.00-5.00(Best)}

Default Value:

X_SessionID

A random session id for each session

Default Value:

X_Direction

Call Direction (inbound|outbound)

Default Value:

X_PeerNumber

Far end phone number

Default Value:

X_PeerName

Far end user ID

Default Value:

Port Status Parameters

This table lists port status parameters.

Port Status

State

Port status, such as on-hook, off-hook, ringing.

Default Value:

N/A

LoopCurrent

Loop current in mA.

Default Value:

N/A

VBAT

PHONE port battery voltage in volts. Not applicable for LINE port.

Default Value:

N/A

TipRingVoltage

Sensed differential Tip/Ring voltage in volts.

Default Value:

N/A

LastCallerInfo

Caller ID of previous call.

Default Value:

N/A

WAN Settings Parameters

This table lists WAN settings parameters.

Internet Settings

AddressingType

The method used to assign an IP address.

Default Value:

DHCP

IPAddress

IP address to assign to the device when **AddressingType** is set to *Static*.

SubnetMask

Subnet mask to use when **AddressingType** is set to *Static*.

DefaultGateway

Default gateway IP address to assign to the device when **AddressingType** is set to *Static*.

DNSServer1

IP address of the first DNS server to use, in addition to the ones obtained from the DHCP server when DHCP is also enabled. If **AddressingType** is set to *Static*, the device only uses **DNSServer1** and **DNSServer2** for DNS lookup. It tries as many as five DNS servers when attempting to resolve a domain name. **DNSServer1** and **DNSServer2** are tried first, whichever is specified, and then the addresses obtained from the DHCP Server if available.

DNSServer2

IP address of the second DNS server to use, in addition to the ones obtained from the DHCP server when DHCP is also enabled. If **AddressingType** is set to *Static*, the device only uses **DNSServer1** and **DNSServer2** for DNS lookup. It tries as many as five DNS servers when attempting to resolve a domain name.

DNSServer1 and **DNSServer2** are tried first, whichever is specified, and then the addresses obtained from the DHCP Server if available.

MACAddressClone

Use this MAC address for WAN interface. MAC address MUST be in six groups of two hexadecimal digits, separated by colons(:) such as "9c:ad:ef:00:00:00"

Default Value:**MTUSize**

MTU size (in bytes) of this interface

Default Value:

1500

VLANEnable

When enabled, only allow packets with the VLANID to pass and insert 802.1Q header according to VLANID and VLANPriority to all outbound packets

Default Value:

false

VLANID

A 12-bit field (0 to 4095) value specifying the VLAN to which the frame belongs. A value of 0 means that the frame does not belong to any VLAN.

The value 4095 is also reserved.

Default Value:

0

VLANPriority

A 3-bit field which refers to IEEE 802.1p priority. It indicates the frame priority level. Value are from 0 (lowest) to 7 (highest). It is valid when VLANID is set to non-zero value.

Default Value:

0

VLANDiscovery

Mode of VLAN ID assignment using DHCP.

Default Value:

Disabled

VLANDiscoveryOption

When VLANDiscovery set to Custom, this specified DHCP option is examined for a valid DVD string

Default Value:

129

802_1XMode

Authentication is disabled or enabled with the specified authentication scheme

Default Value:

Disable

802_1XIdentity

The user name for 802.1x authentication

Default Value:**802_1XPassword**

Password for EAP-MD5, EAP-TTLS Private key, EAP-TTLS/MSCHAPv2

Default Value:**802_1XPrivatekeyPassword**

Password for the private key used in EAP-TLS mode

Default Value:**802_1XAnonymousID**

Anonymous ID, if it is empty, anonymous_identity will not used in authentication

Default Value:**802_1XTLSecurityProfile**

Security profile for 802.1x authentication

Default Value:

1

LLDP-MED

Enable LLDP-MED discovery.

Default Value:

true

LLDP-MEDExclusivePeriod

Number of seconds for LLDP-MED discovering exclusively before the IP is established according to AddressType without Network Policy

Default Value:

5

LLDP-MEDAssetID

The device identity to be used in the LLDP-MED Inventory Management record

Default Value:

true

CDP

Enable CDP discovery

Default Value:

true

Switch Port

Name

(Read-only) Descriptive name for this switch port

Default Value:

SW

Speed

Speed of this switch port

Default Value:

Auto

PC Port

Name

(Read-only) Descriptive name for this PC port

Default Value:

PC

Speed

Speed of this PC port

Default Value:

Auto

Local Time

CurrentLocalTime

Current local date and time of the device (read-only parameter).

CurrentNTPServer1

(Read-only) Hostname or IP address of first NTP server#1 being used

Default Value:**CurrentNTPServer2**

(Read-only) Hostname or IP address of first NTP server#2 being used

Default Value:

LocalTimeZoneTR
Default Value:

Time Service Settings

NTPServer1
Host name or IP address of the first NTP server.
Default Value:
ntp.polycom.com

NTPServer2
Host name or IP address of the second NTP server.

LocalTimeZone
Default Value:
GMT-08:00 (Pacific Time)

DaylightSavingTimeEnable
Enables daylight saving time on the unit.
Default Value:
true

DaylightSavingTimeStart
Daylight Saving Time Start Date. Format: month/day/weekday/hh:mm:ss, where month=1-12, day=±(1-31), weekday=0, 1-7 (0=special, 1=Monday, 7=Sunday), hh=0-23, mm=0-59, ss=0-59.
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-given-month+day+1 (note that day = -1 equivalent to last day of the month).
:ss can be omitted if the value is 0.
:mm:ss can be omitted if mm and ss are both 0.
Default Value:
3/8/7/2

DaylightSavingTimeEnd
Daylight Saving Time End Date. Same format as Start Date.
Default Value:
11/1/7/2

DaylightSavingTimeDiff
Amount of time to add to current time during Daylight Saving Time. Format: [-] hh:mm:ss.
:ss can be omitted if it is 0.
:mm:ss can be omitted if both are 0.
Default Value:
1

DHCP Client Settings

ExtraOptions
Comma separated list of extra DHCP options to be requested.
Default Value:
66, 42, 160, 150, 15

DNS Control

DNSQueryOrder

The order to query available DNS servers when there is a non-zero delay (set in `DNSQueryDelay`) before trying the next DNS server.

Default Value:

DNS Server1, DNS Server2, DHCP Offered DNS Servers

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 query made by the unit for a given domain name. Choose from 0 to 5 seconds.

Default Value:

2

LocalDNSRecordMode

This mode defines how the local DNS records are handled by device.

Default Value:

Persistent Cache

LocalDNSRecordTTL

Time to Live in seconds for local DNS record in Backup Record mode

Default Value:

120

AllowCacheToBeCleared

When enabled, the application may clear negative DNS caches to handle certain failures

Default Value:

false

Local DNS Records

N

One of 32 **Local DNS Records** (numbered 1 - 32). Each record is a mini script of the following format:

Name=A,A,A,...

or

Name=R,R,R,...

where *Name* represents the domain name to be resolved locally, and has the format `prefix+domain` (such as `machine.sip+poly.com` or `_sip._udp.poly.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 that 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 the hostname of the machine providing the service, `port` is the port where the service is found, `pri` is the priority of the target host, `wt` is the relative weight for records with the same priority.

NOTE: If the *A* record of a given host name can't be found in any of the **Local DNS Records**, the device attempts to resolve it using external DNS queries. Any change applied to **Local DNS Record** needs a reboot in order to take effect.

WiFi Settings Parameters

This table lists WiFi settings parameters.

Basic Settings

Enable

Enables OBiWiFi. You must have an OBiWiFi dongle attached to the device to use the feature.

Default Value:

true

PreferredAccessPoint

This value is automatically populated with the last AP that the device's user chose to connect explicitly from the device web page.

Default Value:

None

ShowAccessPointPassword

Check this box and press submit to show all the AP passwords in (unmasked) plain text (no reboot required). The passwords are masked again following a reboot of the device.

Default Value:

false

ScanResultWaitDuration

Wait number of seconds (min=3, max=30) before processing the WiFi scan result.

Default Value:

10

Internet Settings

AddressingType

Assigns an IP address to this interface. Choose from DHCP or Static.

Default Value:

DHCP

IPAddress

The IP address to use if **AddressingType** is Static.

SubnetMask

The subnet mask to use if **AddressingType** is Static.

DefaultGateway

The default gateway to use if **AddressType** is Static.

DNSServer1

Additional DNS Server to use besides the ones received from DHCP.

DNSServer2

Additional DNS Server to use besides the ones received from DHCP.

802_1XMode

Authentication is disabled or enabled with the specified authentication scheme.

Default Value:

Disable

802_1XIdentity

802.1x identity.

Default Value:**802_1XPassword**

Password for EAP-MD5, EAP-TTLS Private key, EAP-TTLS/MSCHAPv2.

Default Value:**802_1XPrivatekeyPassword**

Password for the private key used in EAP-TLS mode.

Default Value:

802_1XAnonymousID

Anonymous ID, if it is "", anonymous_identity will not used in authentication

Default Value:

802_1XTLSSecurityProfile

Security profile for 802.1x authentication

Default Value:

1

Access Point N (N=1,2,...,20)**SSID**

SSID of the access point.

Password

The HEX digits can be upper or lower case.

SecurityEnabled

This read-only parameter indicates whether the AP has security enabled.

Device Admin Parameters

This table lists device admin parameters.

Web Server**Port**

Web Server Port Number

Default Value:

80

TLSProtocol

Configures the lowest TLS/SSL version to use for handshake negotiation when using HTTPS

Default Value:

TLSv1.2

TLSCipherSuite

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 DEFAULT: or HIGH:

Default Value:

PasswordMinimumLength

Minimum length of AdminPassword and UserPassword

Default Value:

0

ExpirationPeriod

Session expiration period in minutes. 0 indicates no expiration.

Default Value:

0

AdminPassword

Administrator Password, case sensitive

Default Value:

admin

UserPassword

User Password, case sensitive

Default Value:

user

CheckWebCookie

Check cookie against the Set-Cookie for the web login session

Default Value:

false

CheckHttpPostToken

Check POST token against session token assigned to the configuration web pages

Default Value:

false

CheckHttpOriginHeader

Check the Origin header in HTTP POST against the expected origin of the device web pages

Default Value:

false

LockOut

Enable or Disable Webserver Lock-out feature

Default Value:

true

LockOutPeriod

60-300 seconds. The period of time the user is locked out of the Webserver

Default Value:

60

LockOutInvalidAttempts

3 - 20. Specify the maximum number of failed login attempts after which the user is locked out from the Webserver.

Default Value:

5

LockOutInvalidAttemptsDuration

60- 300 seconds. After a user reaches the maximum failed login attempts within this time duration, the user is locked out from webserver

Default Value:

60

CustomLogoDownload

Enable or disable displaying custom log on device web page

Default Value:**CustomLogoURL**

The URL of the image of the custom logo

Default Value:**CustomLogoMD5Checksum**

Enter the MD5 Checksum value to compare against the downloaded custom logo file

Default Value:**CustomLogoTag**

The HTML tag for custom logo which is included in the custom web page

Default Value:

UserMenuDeviceAdmin

Whether to show the "Device Admin" menu item when logged in as "user"

Default Value:

true

UserMenuDeviceUpdate

Whether to show the "Device Update" menu item when logged in as "user"

Default Value:

true

IVR

Enable

Enable IVR

Default Value:

true

Password

IVR access password (must be all digits)

Default Value:

Media Loopback

AcceptMediaLoopback

Allow incoming Media Loopback Call

Default Value:

true

MediaLoopbackAnswerDelay

Delay in ms before answering a Media Loopback call

Default Value:

0

MediaLoopbackMaxDuration

Maximum duration in seconds to allow for a Media Loopback call; 0 implies unlimited duration

Default Value:

0

Syslog

Server

IP address of the Syslog Servers

Default Value:**Port**

Syslog Server Port Number

Default Value:

514

Level

Syslog Message Level

Default Value:

7

TAG

Syslog message TAG field. This TAG is a string of alphanumeric characters that MUST NOT exceed 32 characters.

Default Value:

ReportingEnable

Buffer syslog locally and upload it periodically to a server according to ReportingInterval and ReportingURL

Default Value:

false

ReportingInterval

Periodic buffered syslog upload interval in seconds. Minimum is 30s. Maximum is 120s.

Default Value:

60

ReportingURL

URL to upload locally buffered syslog

Default Value:

ReportingUTCTimeStamp

include UTC TimeStamp (TR69 ISO8601) in each message

Default Value:

true

LevelCCTL

Log Level for CCTL

Default Value:

Event 3

LevelXML

Log Level for XML parsing

Default Value:

Event 3

LevelSIP

Log Level for SIP

Default Value:

Event 3

LevelRTP

Log Level for RTP

Default Value:

Event 3

LevelMOH

Log Level for MOH Service

Default Value:

Event 3

LevelFxsTerm

Log Level for DECT terminal

Default Value:

Event 3

LevelParkTerm

Log Level for Park terminal

Default Value:

Event 3

LevelSpTerm

Log Level for SP terminal

Default Value:

Event 3

LevelAnrTerm

Log Level for ANR terminal

Default Value:

Event 3

LevelPageGroup

Log Level for Page Groups

Default Value:

Event 3

LevelDspObi

Log Level for OBi/DSP interface

Default Value:

Event 3

DebugChanTxMask

DSP out-bound channel mask for debug level logging

Default Value:

255

DebugChanRxMask

DSP in-bound channel mask for debug level logging

Default Value:

255

LevelLibTDM

Log Level for LibTDM

Default Value:

Warning

LevelLibDUA

Log Level for LibDUA

Default Value:

Info

LevelLibRTP

Log Level for LibRTP

Default Value:

Warning

LevelLibMedExt

Log Level for Libmediaext

Default Value:

Warning

HTTP Client

UserAgent

The value of User-Agent request-header field

Default Value:

Poly/\${DM}-\${FWV} (\${MAC})

TimeOut

HTTP request time out setting in seconds. This value shall be greater than 60 seconds

Default Value:

600

ProxyServer

Host name or IP address of the HTTP proxy server

Default Value:

ProxyServerPort

Destination port to connect to the HTTP proxy server

Default Value:

80

ProxyAuthUsername

Username of Proxy authentication

Default Value:

ProxyAuthPassword

Password of Proxy authentication

Default Value:

BypassProxyServerForLocalAddresses

Bypass the HTTP proxy server when connecting to local addresses

Default Value:

false

BypassProxyForSubnets

List of intranet subnets which bypass the proxy server. For example: 10.10.10.0/24,192.168.0.0/16

Default Value:

External Port Security

USB

When port security is enabled (On), USB port is disabled. false

Default Value:

false

PCPort

When port security is enabled (On), the PC port is disabled.

Default Value:

false

Emergency Geolocation Settings

E911Enable

Whether to include Geolocation header (and related headers) in the INVITE of emergency calls

Default Value:

false

GeolocationRoutingEnable

Whether Geolocation-Routing header is set to yes or no

Default Value:

false

UsageRuleRetransmission

Set the usage-rule.retransmission-allowed value

Default Value:

false

PEmergencyInfoHeader

Whether to include P-Emergency-Info header in emergency calls

Default Value:

false

ResourcePriorityHeader

Whether to include Resource-Priority header in emergency calls

Default Value:

true

Location Information Service

Enable

Enable the location service

Default Value:

false

PreferredSource

Specify the precedence of the source of the location information

Default Value:

LLDP

CurrentLocation

(Read only) Current location information of device

Default Value:

HTTP-Enabled Location Delivery

Enable

Enable HELD in Location Service

Default Value:

true

RequestType**Default Value:**

Any

Identity

Set the vendor-specific element to include in a location request message. For example, companyID

Default Value:**IdentityValue**

Set the vendor-specific element to include in a location request message

Default Value:**NAI**

Omit or Include the specified NAI (Network Access Identifier) in a location request message

Default Value:

Omit

NAICustomValue

Custom NAI (Network Access Identifier) value to be included in location request message, when NAI is set to 'Custom Value'

Default Value:**PrimaryServer**

Set the IP address or hostname of the location server

Default Value:**PrimaryServerUsername**

Username to authenticate with the HELD server

Default Value:**PrimaryServerPassword**

Password to authenticate with the HELD server

Default Value:**SecondaryServer**

Set the IP address or hostname of the location server

Default Value:**SecondaryServerUsername**

Username to authenticate with the HELD server

Default Value:**SecondaryServerPassword**

Password to authenticate with the HELD server

Default Value:**TLSecurityProfile**

Security profile for HELD

Default Value:

1

X_VerifyServerDomain

Enable verification of server domain against its certificate on HTTPS connection

Default Value:

true

RetryTimer

Specify the retry timeout value in seconds for the location request sent to the location information server

Default Value:

60

Enter Device Location Information

URI

Semicolon-separated location URI list

Default Value:

Country

Enter the country where the phone is located

Default Value:

A1

Enter the national subdivision where the phone is located. For example, a state or province

Default Value:

A3

Enter the city where the phone is located

Default Value:

PRD

Enter the leading direction of the street location

Default Value:

RD

Enter name of the road or street

Default Value:

STS

Enter the suffix name used in RD. For example, street or avenue

Default Value:

POD

Enter the trailing street direction

Default Value:

HNO

Enter street address number

Default Value:

HNS

Enter a suffix for the street address used in HNO

Default Value:

LOC

Enter any additional information that identifies the location

Default Value:

NAM

Enter a proper name to associate with the location

Default Value:

PC

Enter the ZIP or postal code

Default Value:

label

Enter a Label for the location

Default Value:

Remote PCAP Server

Enable

Enable this feature

Default Value:

false

Port

Server Port Number

Default Value:

2002

Clients

List of clients which are allowed to connect to this server. Empty list means everyone is allowed

Default Value:

Packet Capture

On

On/Off

Default Value:

false

Status

Status of capturing

Default Value:

Storage

The name of the interface to listen to

Default Value:

Internal

RestartCaptureOnReboot

Automatically restart packet capture after device is rebooted

Default Value:

false

PromiscuousMode

Capture packets received on the network interface

Default Value:

true

WebAccessExcluded

Do not capture web interface access packets

Default Value:

true

PostponeFirmwareUpdate

If packet capture is running, do not perform a firmware update

Default Value:

true

Platform CA N (N=1, 2)

DownloadURL

URL of certificate to be downloaded

Default Value:

MD5Checksum

If entered, the MD5 checksum to check against the downloaded file's MD5 checksum

Default Value:

CommonName

(Read only) The Common Name attribute of the certificate

Default Value:

FingerPrint

(Read Only) The fingerprint value of the certificate

Default Value:

Obsolete

Remove the certificate if there is one and ignore downloading

Default Value:

Custom Device Certificate N(N=1, 2)

DownloadURL

URL of certificate to be downloaded

Default Value:

MD5Checksum

If entered, the MD5 checksum to check against the downloaded file's MD5 checksum

Default Value:

CommonName

(Read only) The Common Name attribute of the certificate

Default Value:

FingerPrint

(Read Only) The fingerprint value of the certificate

Default Value:

Obsolete

Remove the certificate if there is one and ignore downloading

Default Value:

TLS Platform Profile N (N=1,2)

Protocol

Configure the lowest TLS/SSL version to use for handset negotiation

Default Value:

SSLv2v3

OCSP

Online Certificate Status Protocol Stapling

Default Value:

false

CipherSuite

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 DEFAULT: or HIGH:

Default Value:**CACertList**

The CA cert list to check against for server authentication

Default Value:

Default

DeviceCert

The device certificate to use for device authentication

Default Value:

Polycom

Auto Provisioning Parameters

This table lists auto provisioning parameters.

System Info

X_EnableTR69

Enable TR69 Provisioning

Default Value:

false

X_LinesAllocation

A multi-digit value such that the nth significant digit indicates number of SP services (a.k.a. lines) to allocate to VoiceProfile.{n}. For example 13 means SP1-SP3 to VoiceProfile.1., SP4 to VoiceProfile.2., no lines to the rest.

Default Value:

1;2;3;0,4;0;0;0,0;0;0;0,0;0;0;0

Auto Firmware Update

Method

Current operational method to poll and update firmware. Choose from:

- Disabled: Do not 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: Download from ConfigURL on system start, and then at the specified time of day.

Default Value:

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`).

Default Value:

0

TimeOfDay

Time of the day in `""hh:mm[+rr]""` format, valid when method is set to `""Time of Day""`

Default Value:

00:00+30

RandomDelayRange

The range of delay in seconds inserted before the first attempt only. The minimum value shall be 0

Default Value:

30

FirmwareURL

URL of firmware package. URL must include scheme. Supported schemes are `http://` and `tftp://`

TLSecurityProfile

Security profile to use for SIP TLS connection

Default Value:

1

DnsLookupType

Default Value:

A Record Only

DnsSrvPrefix

Default Value:

No Prefix

Username

Username for authentication, if needed, if scheme is `http://`

Password

Password for authentication, if needed, if scheme is `http://`

Suspend

Suspend Firmware update until cancelled

Default Value:

false

ITSP Provisioning

Method

Current optional method to poll and update configuration from the provisioning server. Choose from:

- Disabled: Do not 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: Download from ConfigURL on system start, and then at the specified time of day.

Default Value:

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*).

Default Value:

0

TimeOfDay

Time of the day in "hh:mm[+rr]" format, valid when method is set to "Time of Day"

Default Value:

00:00+30

ProvisioningOption

Select whether to use UCSSserver (UC Software provisioning method) or ConfigURL (OBI provisioning method).

Default Value:

UCSSServer then ConfigURL

UCSServer

Whether this SP line is enabled for TR69 Provisioning

Default Value:

\$DHCP OPT160; \$DHCP OPT66

ConfigURL

URL of config file.

Default Value:

\$DHCP OPT160/\$MAC.xml; \$DHCP OPT160/\$DM.xml; \$DHCP OPT160;tftp://\$DHCP OPT66/\$DM.xml; \$DHCP OPT66/\$DM.xml; \$DHCP OPT66

Username

The username to login into the provisioning server

Default Value:

PlcmSpIp

Password

The password to login into the provisioning server

Default Value:

DnsLookupType**Default Value:**

A Record Only

DnsSrvPrefix**Default Value:**

No Prefix

Override

Define what local settings can be overridden by this provisioning

Default Value:

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**. Please refer to device provisioning guide for examples on how to use these variables.

OBiTalk (PDMS-SP) Provisioning

Method

Current operational method of OBiTALK (PDMS-SP) provisioning. Choose from:

- Disabled: Do not 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: Download from ConfigURL on system start, and then at the specified time of day.

Default Value:

Disabled

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**).

Default Value:

0

TimeOfDay

Time of the day in ""hh:mm[+rr]"" format, valid when method is set to ""Time of Day""

Default Value:

00:00+30

ConfigURL

URL of config file.

Default Value:

DnsLookupType

Default Value:

A Record Only

DnsSrvPrefix

Default Value:

No Prefix

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**. Please refer to the *Poly OBi ATA Device Deployment Guide* for examples on how to create these variables.

User-Defined Macro 0-3 (\$UDM0 - \$UDM3)

Value

The value can be any plain text or a valid canonical parameter name preceded by a \$ sign. For example:

\$X_DeviceManagement.WebServer.Port

Note: Here you MUST NOT enclose the parameter name following the \$ sign with braces or parentheses.

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.

Statistics Reporting

SyncCQM

Sync call quality metrics

Default Value:

\$MACRO Expansion Supported by the Device

This table lists \$MACRO expansion supported by the device.

MAC

Device MAC address, such as 9CADEF000000

Default Value:

ANY

MACC

Device MAC address with colons, such as 9C:AD:EF:00:00:00

Default Value:

ANY

mac

Device MAC address in lower case with colons, such as 9c:ad:ef:00:00:00

Default Value:

ANY

FWV

Firmware version, such as 1.0.3.1626

Default Value:

ANY

HWV

Hardware version, such as 2.8

Default Value:

ANY

IPA

Current device IP address, such as 192.168.15.100

Default Value:

ANY

DM

Device Model Name, such as ATA402

Default Value:

ANY

DMN

Device model number, such as 402

Default Value:

ANY

OBNI

Device OBI number, such as 200123456

Default Value:

ANY

DSN

Device S/N, such as 88B01NA00000

Default Value:

ANY

GPRMValue Auto Provisioning::GPRM*n***Default Value:***Auto Provisioning* ::ConfigURL,*Auto Firmware Update* ::FirmwareURL**TPRM**Value of *Auto Provisioning* ::TPRM *n***Default Value:***Auto Provisioning* ::ConfigURL,*Auto Firmware Update*::FirmwareURL**UDM**Value of *User-Defined Macro n* ::Value**Default Value:**The value of *User-Defined Macro n* ::ExpandIn

ITSP Profile (General and SP Info Settings) Parameters

This table lists ITSP Profile A, B, C, and D (general and SP info settings) parameters.

General ITSP Settings

Name

Human-readable string to identify the profile instance. Maximum length is 127 characters.

Enable

Enable or Disable this ITSP Profile

Default Value:

Enabled

NumberOfLines

(Ready-only) Display the number of SP lines allocated to this particular ITSP Profile with TR69 Provisioning

Default Value:

X_ToneProfile

Specify which internal tone profile (A or B) to associate with this ITSP Profile

Default Value:

A

SignalingProtocol

Signaling protocols for this ITSP.

Default Value:

SIP

DTMFMethod

Default Value:

Auto

InbandDTMFVolume

DTMF tone volume when sending inband DTMF

Default Value:

-15dB

X_UseFixedDurationRFC2833DTMF

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.

Default Value:

FALSE

X_FixedDurationRFC2833DTMF

The fixed duration (in units of 10ms) to use when X_UseFixedDurationRFC2833DTMF is true

Default Value:

DigitMap

A digit map to restrict the numbers that can be dialed or called with this service. See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of digit map syntaxes. Maximum length is 511 characters.

Default Value:

(1xxxxxxxxx|<1>[2-9]xxxxxxxx|011xx.|xx.|(Mipd)|[^*#]@@.))

STUNEnable

Enables device to send a STUN binding request for its RTP port prior to every call.

Default Value:

false

STUNServer

IP address of domain name of the STUN Server to use.

X_STUNServerPort

UDP listen port of the STUN Server.

Default Value:

3478

X_ICEEnable

Enables device to use ICE algorithm to find the best peer RTP address to forward RTP traffic for every call.

Default Value:

false

X_EarlyICEEnable

Enable starting ICE upon 18x response when possible for outbound calls

Default Value:

false

X_EarlyICEEnableIn

Enable sending 183 with SDP and start ICE on INVITE when possible for inbound calls

Default Value:

false

X_ICEExpires

ICE Expires in milliseconds for outgoing calls

Default Value:

2000

X_ICEExpiresIn

ICE Expires in milliseconds for incoming calls

Default Value:

10

X_IgnoreSTUNCheckError

Ignore STUN check Error while the call is connected

Default Value:

false

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.

Default Value:

false

Service Provider Info

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.

ITSP SIP Settings Parameters

This table lists ITSP SIP settings parameter.

ProxyServer

Host name or IP address of the SIP proxy server.

ProxyServerPort

Destination port to connect to the SIP server.

Default Value:

5060

ProxyServerTransport

Default Value:

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**.

X_NoSIPS

Do not use the "sips" scheme when TLS is used as transport

Default Value:

false

RegistrarServerPort

Destination port to connect to SIP registrar.

Default Value:

5060

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 *SIP 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.

Default Value:

5060

X_OutboundProxyTransport

Transport protocol to connect to OutboundProxy server.

Default Value:

Follow ProxyServerTransport

X_TLSSecurityProfile

Security profile to use for SIP TLS connection

Default Value:

1

X_UserAgentContactFollowProxyServerTransport

If enabled, the user agent should use a Contact and Via transport that agrees with ProxyServerTransport

Default Value:

false

X_BypassOutboundProxyInCall

Enables bypassing the **OutboundProxy** inside the SIP dialog.

Default Value:

false

RegistrationPeriod

Nominal interval between device register in seconds.

Default Value:

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.

Default Value:

500

TimerT2

Value of SIP timer T2 in ms.

Default Value:

4000

TimerT4

Value of SIP timer T4 in ms.

Default Value:

5000

TimerA

Value of SIP timer A in ms.

Default Value:

500

TimerB

Value of SIP timer B in ms.

Default Value:

32000

TimerD

Value of SIP timer D in ms.

Default Value:

32000

TimerE

Value of SIP timer E in ms.

Default Value:

500

TimerF

Value of SIP timer F in ms.

Default Value:

32000

TimerG

Value of SIP timer G in ms.

Default Value:

500

TimerH

Value of SIP timer H in ms.

Default Value:

32000

TimerI

Value of SIP timer I in ms.

Default Value:

5000

TimerJ

Value of SIP timer J in ms.

Default Value:

32000

TimerK

Value of SIP timer K in ms.

Default Value:

5000

InviteExpires

Invite request Expires header value in seconds.

Default Value:

60

ReInviteExpires

Re-invite Expires header value in seconds.

Default Value:

10

RegisterMinExpires

Register Min-Expires header value in seconds (not used at the moment).

Default Value:

15

RegisterRetryInterval

Register retry interval in seconds.

Default Value:

30

X_RegisterRetryResponseCodes

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.

Default Value:

(<40[17]:w120>|<40[34]:w120>|<99[01]:w120-200>|[4-9]xx)

X_RegisterIncludeInstance

Include instance parameter in Register Contact

Default Value:

true

DSCPMark

Diffserv code outgoing SIP packets.

Default Value:

46

X_SpoofCallerID

Allow outbound Caller ID spoofing. If set to Yes, device attempts to set the caller-id name and userid field in the FROM header to that of a remote caller in the case of a bridged call (from another trunk, such as PSTN Line or another SP Service).

Otherwise, 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 does not match the account credentials. Enable this option only if you are sure that the service provider allows it.

Default Value:

false

X_SpoofRemotePartyID

Allow outbound Remote-Party-ID spoofing. X_SpoofCalledID is ignored if this is enabled.

Default Value:

false

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).

Default Value:

false

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.

Default Value:

true

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).

Default Value:

true

X_HoldReferee

Hold the Referee before a blind transfer if the call is not placed on hold. This may allow reconnecting with the Referee if the blind call transfer fails

Default Value:

false

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.

Default Value:

Poly/\${DM}-\${FWV} (\${MAC})

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.

Default Value:

true

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).

Default Value:

true

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 does not support Re-INVITE sent from the client device.

Default Value:

true

X_SessionTimer

Enable standard session refresh protocol based on RFC4028

Default Value:

false

X_SessionExpires

Session expires default value in seconds. If enabled, session refresh is performed half-way before expiration

Default Value:

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_EnforcePAssertedIdentity

Take Caller ID from P-Asserted-Identity header only

Default Value:

false

X_InsertPPreferedIdentity

Insert P-Preferred-Identity header in all outbound INVITE

Default Value:

false

X_InsertPAccessNetworkInfo

Insert P-Access-Network-Info header in REGISTER and INVITE requests

Default Value:

false

X_InsertPrivacyHdr

Insert a 'Privacy:id' header in INVITE for anonymous calls

Default Value:

false

X_UseAnonymousFROM

Use 'sip:anonymous@localhost' in FROM header of INVITE to block outbound Caller ID

Default Value:

false

X_SwitchInfoHeader

Include X-switch-info header (if info available) in the REGISTER requests

Default Value:

false

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.

Default Value:

true

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.

Default Value:

false

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 does not contain '@', it is user as the userid field in TO/FROM header as well as the Request-URI, which are otherwise same as REGISTER.

If the URI contains '@', it is used in the TO and FROM header as 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.

Default Value:

3600

X_RegSubscribe

Enables subscription to the "reg" event package.

Default Value:

false

X_RegSubscribeExpires

Expires value for subscription to the "reg" event package.

Default Value:

3761

X_NoNonceAuth

Enable sending Authorization header without being challenged first

Default Value:

false

X_BackupProxyServers

Additional list (host names or IP addresses separated by comma) of secondary SIP proxy servers for X_ProxyServerRedundancy.

Default Value:**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.

Default Value:

false

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.

Default Value:

false

X_CheckPrimaryFallbackInterval

Interval in seconds at which the device checks the primary fallback list of candidate servers.

Default Value:

60

X_CheckSecondaryFallbackInterval

Interval in seconds at which the device checks the secondary fallback list of candidate servers.

Default Value:

60

X_UnregisterOnFallback

Whether to send unREGISTER to the current active server when fallback to a higher priority server

Default Value:

true

X_ProxyFailoverResponseCodes

A comma separated list of digit maps where each map is a list of SIP response codes that trigger proxy failover. The first map is for REGISTER response codes; the second map is for INVITE response codes. If the 2nd map is not specified, INVITE will follow the first map.

Default Value:

([5-9]xx)

X_InviteFailoverWaitRegTimer

Maximum time (in milliseconds) to wait for successful register failover to retry INVITE after a failure

Default Value:

32000

X_ProxyRequire

If this parameter is not 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.

Default Value:

70

X_AcceptLanguage

If this parameter is not 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

Default Value:

true

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.

Default Value:

false

X_DnsSrvSipsTcpPrefix

Whether to use `""_sips._tcp.""` instead of `""_sip._tls""` for TLS transport when sending DNS SRV queries

Default Value:

false

X_DnsNAPTR

Enable DNS NAPTR Lookup for the Proxy Server or the Outbound Proxy Server

Default Value:

false

X_Support100rel

Enable support of 100rel (RFC3262)

Default Value:

false

X_UserEqPhone

Includes the parameter 'user=phone' in Request-URI and To-URI of outbound INVITE.

Default Value:

false

X_UseTelURI

Use tel: URI in outbound INVITE in Req-URI and To-URI

Default Value:

false

X_CallWaitingIndication

Enables including an indication in an 18x response to the calling peer if this is a call- waiting situation.

Default Value:

false

X_DiscoverPublicAddress

Enables letting the device use the public IP address and port it has discovered as its SIP Contact address.

Default Value:

false

X_UsePublicAddressInVia

Enables using the discovered external IP address (instead of the unit's assigned local IP address) in outbound Via header.

Default Value:

false

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 are using port forwarding on the external router to route inbound SIP messages to the device.

Default Value:

true

X_UseCompactHeader

Enables using compact form SIP message header names.

Default Value:

false

X_OmitContentLength

Omit Content-Length header if ProxyServerTransport and X_OutboundProxyTransport are both UDP

Default Value:

false

X_FaxPassThroughSignal**Default Value:**

ReINVITE

X_IncludeMessageHash

Include a MD5 hash of all the SIP message headers in a X-MD5-Hash header. Also include a hash of the SDP in the x-md5-hash attribute

Default Value:

false

X_EnableRFC2543CallHold

Enables interpretation of call hold indication per RFC2543.

X_VerifyServerDomain

Enable verification of server domain against its certificate on a SSL/TLS connection

Default Value:

false

X_RejectKeyResponseCode

SIP Response Code and Phrase to inbound INVITE when user presses the 'Reject' key

Default Value:**X_SupportOutbound**

Enable Support for SIP Outbound (RFC 5626)

Default Value:

false

X_SupportPath

Enable support for the path extension

Default Value:

true

X_SupportServiceRoute

Enable handling of the Service-Route header

Default Value:

true

X_SupportPAssociatedURI

Enable handling of the P-Associated-URI header

Default Value:

true

X_SupportGRUU

Support Global Routable User Agent URIs in SIP (RFC5627)

Default Value:

false

X_Sticky18x

Ignore further 18x responses w/o SDP upon receiving the first 18x w/ SDP to INVITE

Default Value:

true

Feature Configuration

X_ShareLineMethod

Select the signaling method for share line operation

Default Value:

call-info

X_CallInfoSubscribeExpires

CallInfo (SCA) Subscription Renewal interval in seconds. Set the value to 0 to disable subscription renewal

Default Value:

3600

X_LineSeizeSubscribeExpires

Line-seize event subscription renewal interval in seconds.

Default Value:

15

ITSP RTP Settings Parameter

This table lists ITSP RTP settings parameter.

RTP

LocalPortMin

Base of port range for tx/rx RTP with this SP.

Default Value:

16600

LocalPortMax

Top of port range for tx/rx RTP with this SP.

Default Value:

16798

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.

Default Value:

0

DSCPMark

Diffserv code for outgoing RTP packets with this SP.

Default Value:

46

X_UseSSL

Enables forcing the device to send RTP over an SSL channel when the ITSP is Google Voice.

Default Value:

false

X_RefreshSession

Allow incoming RTP packets to refresh session

Default Value:

true

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.

Default Value:

true

RTCP

Enable

Enables RTCP.

Default Value:

false

TxRepeatInterval

RTCP packet transmission interval in milliseconds.

Default Value:

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).

Default Value:

false

X_VqPublishEnable

Enable VQ report sent to the proxy server using Publish method

Default Value:

false

X_VqPublishUrl

A Username or URL to send Voice Quality Report using Publish method

Default Value:

X_VqPublishInterval

Interval in seconds between VQ reports; 0 or {blank} disables periodic reports

Default Value:

0

X_VqPublishOnSSRCChange

Enable VQ report when SSRC changes

Default Value:

true

Jitter Buffer

Adaptive

Enable jitter buffer adaptation

Default Value:

true

MaximumSize

Maximum jitter buffer size in milliseconds

Default Value:

250

SetPoint

Initial playout delay in milliseconds

Default Value:

60

Target

Target playout delay in milliseconds

Default Value:

20

AdaptationSlope

Maximum adaptation slope in samples per 10ms

Default Value:

16

TargetFax

Target playout delay in milliseconds for fax calls

Default Value:

200

SPn Services Parameters

This table lists SPn services parameters.

SPnService

Enable

Enables this line.

Default Value:

true

EnableTR

Whether this SP line is enabled for TR69 Provisioning

Default Value:

Enabled

X_ReportSessionStats

Report session statistics to TR69/ACS server when a call ends on this line

Default Value:

false

X_ReportSessionStart

Report when a session starts

Default Value:

false

X_DisplayNumber

A number to represent this service on the phone screen

Default Value:

X_ServProvProfile

Selects a Service Provider Profile for this service.Choose from A or B.

Default Value:

A

X_RingProfile

Selects a Ring Profile to ring the PHONE port with for incoming calls on this service that are routed to the PHONE port. The ringing pattern is taken from the given profile. Choose from A or B.

Default Value:

A

X_CodecProfile

Selects a Codec Profile for all calls on this service. Choose from A or B.

Default Value:

A

X_InboundCallRoute

Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the PHONE port (`ph`). See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes for specifying this parameter.

Default Value:

`ph, ph2`

X_RegisterEnable

Enables registration for this line. If set to true, device sends periodic SIP REGISTER to the service provider according to the settings in the ITSP Profile. Otherwise, device does not send any SIP REGISTER for the service.

Default Value:

true

X_AcceptSipFromRegistrarOnly

Accept SIP packets coming from the current registrar IP address only

Default Value:

false

X_NoRegNoCall

Enables blocking making or receiving calls on this service unless registration with the SIP server is successful.

Default Value:

false

X_KeepAliveExpiresFollowServer

Follow the keep alive expires value from server in the VIA-header keep parameter

Default Value:

true

X_KeepAliveEnable

Enables sending keep alive message. If set to true, device sends periodic keep-alive messages to the destination specified in **X_KeepAliveServer** and **X_KeepAliveServerPort**, at the interval specified in **X_KeepAliveExpires**. The content of this message is the ASCII string "keep-alive\r\n".

Default Value:

false

X_KeepAliveExpires

Keep-alive period in seconds.

Default Value:

15

X_KeepAliveServer

Host name or IP address of keep-alive server.

X_KeepAliveServerPort

UDP port of the keep-alive server.

Default Value:

5060

X_KeepAliveMsgType**Default Value:**

keep-alive

X_CustomKeepAliveMsg

Defines the custom message to be used when **X_KeepAliveMsgType** is "custom". The value should have the following format:

mtid=NOTIFY;event= *<whatever>* ;user= *<anyone>*

where

SIP messages for keep-alive are sent only once without retransmission. The device ignores responses to the SIP messages.

X_UserAgentPorts

UDP port where the device sends and listens for SIP messages.

Default Value:

5060

DirectoryNumber

Directory number associated with this service.

X_UserAgentPorts

A comma separated list of (up to 10) alternative UDP Ports for tx/rx SIP packets

Default Value:**X_DefaultRing**

Default ring pattern number to ring the PHONE port for incoming calls on this trunk that are routed to the PHONE port according to the **InboundCallRoute** of this service. The ring pattern is taken from the selected Ring Profile. Choose from 1 through 10.

Default Value:

1

X_CallOnHoldRing

Pattern to ring PHONE port when holding a call on this trunk that has been connected to the PHONE port. Typically this is a very short distinctive ring pattern that serves as a reminder to the user that a call is being on hold. The ring pattern is taken from the selected Ring Profile. Choose from No Ring, or 1 through 10.

Default Value:

8

X_RepeatDialRing

The ring pattern number to use to ring the PHONE port when a repeat dial operation on this trunk is successful as the called party is either ringing or answered.

Default Value:

5

X_BargeInRing

Call Waiting Ring pattern to ring the PHONE port when the incoming call is requesting to barge-in. This is applicable in a call-waiting scenario on the PHONE port.

Default Value:

4

X_CallParkedRing

Pattern to ring PHONE when one or more calls are parked

Default Value:

10

Debug Options

X_SipDebugOption

Default Value:

Disable

X_SipDebugExclusion

A list of SIP methods to exclude from the syslog for this SP service. For example: notify, subscribe.

X_Proxy

Enables proxy mode operation on this SP service. If enabled, the SP accepts SIP Registration from one client device from the LAN side, which must be using the same user-id and password as this SP's **AuthUserName** and **AuthPassword** parameters for authentication. The client device, known as the *local_client*, may 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 local client should be set to:

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

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

For example, SP1 has a *local_client* with the userid 4086578118 and the client wants to make and receive calls using SP3, which is set up for Google Voice. The SP1 **InboundCallRoute** shall include the following rule:

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

The SP3 **InboundCallRoute** shall be:

```
{sp1(408657118@local_client)}
```

Default Value:

false

X_ProxyClientConfig

A list of device attributes separated by a space or newline character for provisioning a device with the given MAC address and model number. Each attribute has the syntax <attribute-name>=<attribute-value> with no white space before and after the '=' sign. Every character within the pair of double quotes is taken as the attribute's value.

X_AcceptResync

Default Value:

yes without authentication

SIP Credentials

AuthUserName

The User ID to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the device is challenged by the UAS with a 401 or 407 Response.

AuthPassword

The password (corresponding to **AuthUserName**) to authenticate to a SIP UAS (User Agent Server) when an outbound SIP request sent by the device is challenged by the UAS with a 401 or 407 Response.

X_EnforceRequestUserID

Enforce incoming INVITE request userid to match AuthUserName or ContactUserID

Default Value:

false

DTLSPeerCertAltName

Altname for verifying peer DTLS Certificate

Default Value:

URI

This parameter affects the way the AOR is formed by the device in outbound SIP Requests. The AOR has the format: `user@domain`

If the value of URI is empty, device gets the user portion of its AOR from the **AuthUserName**, and the domain portion the value of ITSP Profile's **UserAgentDomain** if it is not empty, or that of the **ProxyServer** otherwise.

If the value URI is not empty and does not 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 device takes it as the AOR as is.

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 device uses **AuthUserName** and **AuthUserPassword** to compute authorization if challenged by a 401 or 407 response.

X_ContactUserID

An alternative userid to be used in Contact header. Enter ""Random"" to let the device generate a random one.

Default Value:

false

Share Line Features

X_ShareLine

Check this if this account is a share line

Default Value:

false

X_ShareLineUserID

A third-party userid to register with for the Share Line

Default Value:

X_ShareLineBargeIn

Check this if the share line supports barge-in function

Default Value:

false

Calling Features

CallerIDName

Displays 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.

MaxSessions

The maximum number of simultaneous calls that can be established on this service.

Default Value:

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 phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

CallForwardOnBusyEnable

Note: Users can set this parameter from the phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

CallForwardOnNoAnswerEnable

Enables call forwarding of all incoming calls when the call is not 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 phone with a Star Code.

Default Value:

false

CallForwardOnNoAnswerNumber

Directory number to forward all incoming calls when the call is not answered after a period specified in **CallForwardNoAnswerRingCount**.

Note: Users can set this parameter from the phone 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.

Default Value:

2

X_BlockedCallers

A comma-separated list of as many as 10 caller numbers to block from calling this service.

X_MailboxID

The mailbox ID to subscribe MWI with

Default Value:**X_CheckVoiceMailNumber**

The number to call to check voicemail

Default Value:**MWIEnableMask**

Message Waiting Indication Enable Mask for this service (1,2 for Handset 1,2, and 3 for both)

Default Value:

3

X_VMWIEnableMask

Visual Message Waiting Indication Enable Mask for this service (1,2 for Handset 1,2, and 3 for both)

Default Value:

3

X_MWIRoute

Rules to enable MWI signals on MWI Notifications

Default Value:**MessageWaiting**

This state parameter indicates if there are any new messages for this subscriber on the service provider's voicemail system.

Default Value:

false

MessageCount

Messages count. Format: new/old (urgent-new/urgent-old)

Default Value:**AnonymousCallBlockEnable**

Enables blocking Anonymous Calls on this service. Anonymous calls are rejected with a SIP 486 (Busy) response and Call Forward On Busy service is not applied.

Note: Users can set this parameter from the phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

Default Value:

false

X_BridgedOutboundCallMaxDuration

Limit on the call duration in seconds for all outbound calls that are bridged from the same or another trunk. A blank or 0 value implies the call duration is not limited.

X_AcceptDialogSubscription

Enables the device to accept SUBSCRIBE to this trunk's dialog event package.

Default Value:

false

X_AcceptLinePortStatusSubscription

Accept subscription to line port status

Default Value:

false

X_SkipCallScreening

Enables the device to automatically skip call screening when the underlying ITSP is Google Voice.

Default Value:

true

X_SMSNotify

Ring the phone on SMS reception from Google Voice and display the first few characters of the message as Caller-ID.

Default Value:

false

X_XMPPPriority

XMPP Priority to assume by this client for Google Voice when there are multiple clients using the same account. Valid values are 0 (highest) or 3 through 127.

Default Value:

0

X_GTalkSimultaneousRing

Ring all other clients using the same Google Voice account at present.

Default Value:

true

X_SRTP

Default Value:

Disable SRTP

X_SRTPCryptos

Comma separated list of cryptos to offer. For example: AES_CM_256_HMAC_SHA1_80, AES_128_CM_HMAC_SHA1_80

Default Value:

AES_CM_128_HMAC_SHA1_80

X_ConferenceBridge

The number of an external conference bridge to use for conference calls

Default Value:

cbridge

PDMS-SP Service Settings Parameters

This table lists PDMS-SP service settings parameters.

Enable

Enables the PDMS-SP Service (the built-in free voice service that comes with every OBi Device).

Default Value:

true

DisplayNumber

A number to represent this service on the phone screen

Default Value:

LocalPort

The UDP or TCP port used by the device to send and listens for PDMS-SP messages.

Default Value:

10000

TryMultiplePorts

Enables the device to try a few random UDP ports until it can successfully join the PDMS-SP network.

Default Value:

true

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. See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes for specifying a Digit Map.

Default Value:

(<ob>xxxxxxxx | obxxxxxxxx)

InboundCallRoute

Routing rule for directing incoming calls on this service. The default rule is to send all incoming calls to the PHONE port (ph). See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes for specifying this parameter.

Default Value:

ph, ph2

RingProfile

Selects a Ring Profile to ring the PHONE port with when an incoming call is routed to the PHONE port. Choose from A or B.

Default Value:

A

CodecProfile

Selects a Codec Profile to be used for all calls on this service. Choose from A or B.

Default Value:

A

DefaultRing

Default ring pattern number to ring the PHONE port for incoming calls on this trunk that are routed to the PHONE port according to the **InboundCallRoute** of this service. The ring pattern is taken from the selected Ring Profile. Choose from 1 through 10.

Default Value:

2

CallOnHoldRing

Pattern to ring PHONE port when holding a call on this trunk that has been connected to the PHONE port.

Typically this is a very short distinctive ring pattern that serves as a reminder to the user that a call is being on hold. The ring pattern is taken from the selected Ring Profile. Choose from No Ring, or 1 through 10.

Default Value:

8

RepeatDialRing

The ring pattern number to use to ring the PHONE port when a repeat dial operation on this trunk is successful as the called party is either ringing or answered.

Default Value:

4

DTMFMethod
Method to pass DTMF to the peer.
Default Value:
Auto

FixedDurationRFC2833DTMF
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.
Default Value:
16

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.
Default Value:
true

PDMS-SP Calling Features Parameters

This table lists PDMS-SP calling features parameters.

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 phone with a Star Code.
Default Value:
false

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 phone with a Star Code.

CallForwardOnBusyEnable
Directory number to forward all incoming calls on this line when busy.
NOTE: Users can set this parameter from the phone with a Star Code.

Default Value:
false

X_MissedCallNotificationMask
Missed Call Notification Mask for this service (1,2,4,8,...for Handset 1,2,3,4,...)
Default Value:
1023

X_ParkedCallNotificationMask
Parked Call Notification Mask for this service (1,2,4,8,...for Handset 1,2,3,4,...). Note that a handset will only get the notification if it has this line included in its OutboundServices.
Default Value:
1023

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 phone with a Star Code.

CallForwardOnNoAnswerEnable

Enables call forwarding of all incoming calls when the call is not 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 phone with a Star Code.

Default Value:

false

CallForwardOnNoAnswerNumber

Directory number to forward all incoming calls when the call is not answered after a period specified in *CallForwardNoAnswerRingCount*.

Note: Users can set this parameter from the phone 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.

Default Value:

2

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.

Default Value:

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 is not applied.

Note: Users can set this parameter from the phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

Default Value:

false

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 phone with a Star Code.

Default Value:

false

PDMS-SP Inbound Direct Dialing Authentication Parameters

This table lists PDMS-SP inbound direct dialing authentication parameters.

AuthMethod

Method to authenticate inbound direct dialing callers

Default Value:

HTTP Digest

AuthUserID1

One of 4 user IDs for authenticating direct dialing callers.

AuthPassword1

One of 4 passwords for authenticating direct dialing callers.

AuthUserID2

One of 4 user IDs for authenticating direct dialing callers.

AuthPassword2

One of 4 passwords for authenticating direct dialing callers.

AuthUserID3

One of 4 user IDs for authenticating direct dialing callers.

AuthPassword3

One of 4 passwords for authenticating direct dialing callers.

AuthUserID4

One of 4 user IDs for authenticating direct dialing callers.

AuthPassword4

One of 4 passwords for authenticating direct dialing callers.



NOTE: If **AuthPassword** is specified, **AuthUserID** can be set to blank to let the device use the default value, which is a special hash of the **AuthPassword**. This is only applicable if the external gateway is also an OBi device that understands how to generate the default **AuthUserID** using the same hash function.

PDMS-SP Jitter Buffer Parameters

This table lists PDMS-SP jitter buffer parameters.

Adaptive

Enable jitter buffer adaptation

Default Value:

true

MaximumSize

Maximum jitter buffer size in milliseconds

Default Value:

250

SetPoint

Initial playout delay in milliseconds

Default Value:

60

Target

Target playout delay in milliseconds

Default Value:

20

AdaptationSlope

Maximum adaptation slope in samples per 10ms

Default Value:

16

TargetFax

Target playout delay in milliseconds for fax calls

Default Value:

200

User Prompts Parameters

This table lists user prompts parameters.

User<N>Description, <N> = 1, 2, ..., 20

A text string that describes the contents of this user prompt.

User<N>Length, <N> = 1, 2, ..., 20

This is a read-only status parameter. It shows the space occupied by this prompt in number of milliseconds.

EnableDownloadURL

Enable the use of DownloadURL to download user prompt package.

Default Value:

False

DownloadURL

A URL to download a user prompt package. WARNING: This will overwrite all existing user prompts)

Default Value:**SpacedUsed**

This is a read-only status parameter. It shows the amount of recording space used in number of milliseconds.

SpaceAvailable

This is a read-only status parameter. It shows the amount of recording space remaining in number of milliseconds.

Auto Attendant Parameters

This table lists auto attendant parameters.

Enable

Enables AA. If enabled, the AA answers an incoming call that has been routed to it after a period as specified in **AnswerDelay**. If disabled, the AA won't attempt to answer any incoming call.

Default Value:

true

DigitMap

Once the AA answers an incoming call, it presents the caller with an option to make a further call using one of the available voice services on the device. This Digit map serves to restrict the numbers that can be dialed or called via this AA option.

See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes to specify a digit map.

Default Value:

```
([1-9]x?* (Mpli) | [1-9] | [1-9] [0-9] |
<00:$1>|0|**1 (Msp1) |
**2 (Msp2) | **3 (Msp3) |
**4 (Msp4) | **70 (Mli) | **8 (Mbt) | *
*81 (Mbt) | **82 (Mbt2) |
**9 (Mpp) | (Mpli) )
```

OutboundCallRoute

After the caller dials a number that is acceptable by the AA (according to its **DigitMap**) to make a further call, the device uses this outbound call routing rule to determine the service with which to make this call.

See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes to specify this parameter.

Forking to multiple numbers in an AA outbound call is supported on the OBi202.

For example, on the OBi202 you may have a rule like this: {0:ph, ph2}, which forks to ring both PHONE1 and PHONE2. You can have as many as four destinations in a forking rule.

Default Value:

For ATA400/402:

```
{ ([1-9]x?* (Mpli) ) :pp},
{0:ph},
{(<**1:> (Msp1) ) :sp1},
{(<**2:> (Msp2) ) :sp2},
{(<**3:> (Msp3) ) :sp3},
{(<**4:> (Msp4) ) :sp4},
{(<**70:> (Mli) ) :li},
{(<**82:> (Mbt2) ) :bt2},
{(<**81:> (Mbt) ) :bt},
{(<**8:> (Mbt) ) :bt},
{(<**9:> (Mpp) ) :pp},
{ (Mpli) :pli}
```

For OBi:202/OBi302:

```
{ ([1-9]x?* (Mpli) ) :pp},
{0:ph.ph2},
{(<**1:> (Msp1) ) :sp1},
{(<**2:> (Msp2) ) :sp2},
{(<**3:> (Msp3) ) :sp3},
{(<**4:> (Msp4) ) :sp4},
{(<**70:> (Mli) ) :li},
{(<**82:> (Mbt2) ) :bt2},
{(<**81:> (Mbt) ) :bt},
{(<**8:> (Mbt) ) :bt},
{(<**9:> (Mpp) ) :pp},
{ (Mpli) :pli}
```

PrimaryLine

The device processes the parameter by substituting the occurrences of *pli* and *(Mpli)* in **DigitMap** and **OutboundCallRoute** with the corresponding *code* and *(Mcode)*. For example, if **PrimaryLine** = PSTN Line, then all occurrences of *pli* and *(Mpli)* are substituted internally with *li1* and *(Mli1)*.

Default Value:

SP1 Service

AnswerDelay

Period of time in milliseconds that the AA waits before answering an incoming call that has been routed to it.

Default Value:

4000

CallbackAnswerDelay

Delay in ms the AA answers after ring when the AA should callback if the caller hangs up before it answers

Default Value:

10000

NumberOnNoInput

In the case that the caller does not enter any option from the top level menu after the menu has been announced for 3 times, the AA directs the caller to the number specified in this parameter. If this number is not specified, the AA terminates the current call.

Note: According to the default **DigitMap** and **OutboundCallRoute**, calling 0 means calling the PHONE port.

Default Value:

0

UsePIN

Enables using a PIN to authenticate callers when they select the option to make a further call. If PIN1, PIN2, PIN3, and PIN4 are all empty, device treats it as if UsePIN is set to No. Otherwise, the caller must enter one of the non-empty PIN in order to proceed.

Default Value:

false

PIN1

PIN code to make a call (must be all digits). Maximum length = 15 digits.

PIN2

PIN code to make a call (must be all digits). Maximum length = 15 digits.

PIN3

PIN code to make a call (must be all digits). Maximum length = 15 digits.

PIN4

PIN code to make a call (must be all digits). Maximum length = 15 digits.

Auto Attendant Prompt Parameters

This table lists auto attendant prompt parameters.

Welcome

Prompt List to replace the system's **Welcome** message.

InvalidPin

Prompt List to replace the system's **InvalidPin** message.

EnterPin

Prompt List to replace the system's **EnterPin** message.

MenuTitle

Prompt List to replace the system's **MenuTitle** message.

Menu

Prompt List to replace the system's **Menu** message.

PleaseWait

Prompt List to replace the system's **PleaseWait** message.

EnterNumber

Prompt List to replace the system's **EnterNumber** message.

Bye

Prompt List to replace the system's **Bye** message.

Voice Gateway Parameters

This table lists voice gateway parameters.

Voice Gateway n ($n=1-8$)

Enable

Enables this voice gateway.

Default Value:

true

Name

An arbitrary user-friendly name to identify this gateway (optional).

AccessNumber

The gateway's PDMS-SP 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).

Default Value:

(xx.)

AuthUserID

A user ID to authenticate with the gateway.

AuthPassword

A password to authenticate with the gateway.

Trunk Group Parameters

This table lists trunk group parameters.

Trunk Group n ($n=1-4$)

Enable

Enables this trunk group.

Default Value:

true

Name

An arbitrary user friendly name to identify this trunk group (optional).

TrunkList

A comma-separated list of names of trunks to include in this trunk group.

Default Value:

For TG1 and TG2, the default is: sp1, sp2, sp3, sp4

For other TG, the default is (blank)

DigitMap

Digit map associated with this trunk group. It can be referenced as (Mtn).

Default Value:

For TG1 and TG2, the default is (Msp1). For TG3 and TG4, the default is (xx.)

Page Group Parameters

This table lists page group parameters.

GroupName

The name for the page group

Default Value:

Emergency

Designate group as "Single". Page on the group is auto answered by handsets belonging to the group when possible, regardless of their "Single" setting.

Default Value:

false

MulticastAddress

The IP Multicast Address used by this Page group.

Default Value:

224.1.1.100

MulticastPort

The Multicast port number used by this Page group

Default Value:

The default setting for MulticastPort is different for each Page Group (increments by 2):

1: 65322

2: 65324

...

10: 65340

TTL

TTL of multicast packet (1-255)

Default Value:

2

AudioCodec

Codec to use for paging. Note that with Polycast, only G.711 and G.722 are supported.

Default Value:

G711U

TxPacketSize

RTP transmission packet size in milliseconds

Default Value:

20

RTCPtxInterval

RTCP transmission interval when talking in milliseconds

Default Value:

0

SilenceSuppression

Enable silence suppression when talking

Default Value:

false

PlayToneOnIncomingPage

Play a short Paging Tone on receiving a new incoming page

Default Value:

true

StartTalkingOnJoin

Start talking immediately when joining the group

Default Value:

true

TalkingAlertTone

A short Call Waiting Tone to play periodically to remind user it is currently in the talking mode

Default Value:

CWT10

SwitchToTalkModeDigit

Digit to switch from listening mode to talking mode

Default Value:

*

SwitchToListenModeDigit

Digit to switch from talking mode to listening mode

Default Value:

#

Polycast

Enable Polycom Multicasting.

Default Value:

false

PolycastListen

Enable Polycom Multicast Listening.

Default Value:

true

PolycastGroup

Polycom Group to multicast with

Default Value:

1

PHONE Port Parameters

This table lists PHONE port parameters.

PHONE Port

Enable

Enables the PHONE port.

Default Value:

true

DigitMap

Restricts the numbers that can be dialed or called from the PHONE port. If the caller dials a number that is not 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.

See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes to specify a digit map.

Default Value:

```
([1-9]x?* (Mpli) | [1-9]S9 | [1-9] [0-9]S9 | 911 | **0 | *** | # | **1 (Msp1) | **2 (Msp2) | **3 (Msp3) | **4 (Msp4) | **9 (Mpp) | (Mpli) )
```

OutboundCallRoute

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.

See the **Poly ATA Call Routing and Digit Map** section in the [Poly ATA Administrator Guide](#) for a description of the syntaxes to specify this parameter.

Default Value:

```
{ ([1-9]x?* (Mpli) ) : pp }, { (<#:>) : ph2 }, { **0 : aa }, { *** : aa2 },  
{ (<**1:> (Msp1) ) : sp1 }, { (<**2:> (Msp2) ) : sp2 }, { (<**3:> (Msp3) ) : sp3 },  
{ (<**4:> (Msp4) ) : sp4 }, { (<**9:> (Mpp) ) : pp }, { (Mpli) : pli }
```

CallReturnDigitMaps

Call Return is the service where the user 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 does not 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.

Default Value:

```
{ pli : (xx.) }, { sp1 : (<**1>xx.) }, { sp2 : (<**2>xx.) }, { sp3 : (<**3>xx.) }, { sp4 :  
(<**4>xx.) }, { bt : (<**8>xx.) }, { pp : (<**9>xx.) }
```

PrimaryLine

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** = PSTN Line, then all occurrences of pli and (Mpli) are substituted internally with li1 and (Mli1).

Default Value:

SP1 Service

ToneOnPrimaryServiceDown

Default Value:

Normal Dial Tone

Ringer

RingFrequency

Ringer frequency in Hz (14 to 68) to apply to the PHONE port when ringing.

Default Value:

20

RingVoltage

Peak ringer voltage in volts (55 to 82) to apply to the PHONE port when ringing.

Default Value:

72

RingWaveform

Default Value:

Sinusoidal

InterleavedRing

When both PHONE ports are ringing, enabling this option causes the device to interleave the ring signal applied to each port to reduce the chance of overloading the power supply.

Default Value:

true

Port Settings

OnHookTipRingVoltage

Tip/Ring Voltage when the attached phone is on hook (30 V to 52 V).

Default Value:

48

OffHookCurrentMax

Maximum supported current (15 mA to 45 mA) when the attached phone is off-hook.

Default Value:

20

Impedance**Default Value:**

600

OnHookPowerSaveMode

Enable Power Save Mode in on-hook state. This is a global setting and is applied to all phone ports.

Default Value:

false

HVICPowerThreshold

HVIC power alarm threshold in WATT

Default Value:

2.5 W

DTMFDetectMinLength

Minimum duration for a DTMF tone to be considered valid. Actual threshold in ms is loosely given by $30 + \text{value} * 10$. So for a value of 3, the detection threshold is around 60ms.

Default Value:

3

DTMFDetectMinGap

Minimum gap between DTMF digits in units of 10ms for the purpose of detection. If a new digit starts too soon after the previous digit ends, it will be dropped.

Default Value:

7

CidNoNameFormat

How to format Caller ID NAME field when name information is not available

Default Value:

Show Code 0

CidNumberTransform

Digit map to transform Caller ID Number before sending it to the phone

Default Value:

(<+:>xx.)

CallerIDMethod

Default Value:

FSK (Bell202)

CallerIDTrigger**Default Value:**

After First Ring

ChannelTxGain

Transmit gain in dB (-12 to 12) to apply to signal sent from the device to the attached phone(s).

Default Value:

0

ChannelRxGain

Receive gain in dB (-12 to 12) to apply to signal received by the device from the attached phone(s).

Default Value:

0

SilenceDetectSensitivity

PHONE port silence detection servers the purpose of driving silence suppression in RTP transmission when the phone Call terminates on SP1/2 or PDMS-SP Service and silence suppression is enabled.

Default Value:

Medium

Calling Features

CallCommandSignalMethod

Method for user to signal a call command during a call.

Default Value:

N. America

CallerIDEnable

Enables Caller ID Signal generation. This option can be set to Yes even if the attached phone is not capable of displaying Caller ID. There is no harm in sending Caller ID signal while the phone is in the on hook state.

Default Value:

true

CallWaitingCallerIDEnable

Enables Call Waiting Caller ID (CWCID) Signal generation.

The CWCID signal is sent to the phone when it is in the off hook state. It starts with a handshake between the device and the attached phone, by exchanging audible short tones. The device proceeds with the transmission of the remaining Caller ID signal only if the handshake succeeds (with a phone is capable of displaying CWCID). In that case the phone mutes the handset earpiece until the CWCID signal is complete. Some users however may still find the audible handshake tones objectionable, especially if their phones do not support CWCID. Set this option to No if you don't want the CWCID feature, or don't have phones that can display CWCID.

Default Value:

true

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.

Default Value:

true

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.

Default Value:

true

CallTransferEnable

If Call Transfer is disabled, hanging up the phone in the above scenarios ends all the calls except for the one that is holding, which remains on hold (Cases 1 and 2).

Default Value:

true

ConferenceCallEnable

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, the user 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 phone 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).

Default Value:

true

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 PHONE 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.

Default Value:

false

StartConfOnPeerRing

Allow 3-way Conference when the 3rd party is still ringing

Default Value:

true

CallWaitingEnable

Enables call waiting service. Call Waiting is the situation where a new incoming call is routed to the PHONE port when there is already another call connected. If this service is enabled, the device plays call-waiting tone to alert the user, as well as generates 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 phone with a Star Code.

Default Value:

true

DoNotDisturbEnable

Enable Do Not Disturb on this phone port

Default Value:

false

ToneProfile

Selects a Tone Profile for call progress tone generation. Choose from A or B.

Default Value:

A

StarCodeProfile

If set to `None`, no star code is recognized by the device.

Default Value:

A

LastDialedNumber

(Read only) Last number dialed out on the PHONE port.

LastCallerNumber

(Read only) Last caller's number that rings the PHONE port.

AcceptMediaLoopback

Enables the device to accept incoming media loopback calls.

Default Value:

true

MediaLoopbackAnswerDelay

Delay in milliseconds before the device answers an incoming media loopback call.

Default Value:

0

MediaLoopbackMaxDuration

Maximum duration in seconds to allow for an inbound media loopback call. Set the value to blank or 0 to make it unlimited.

Default Value:

0

RepeatDialInterval

Interval in seconds between retry in a repeat dial operation.

Default Value:

30

RepeatDialExpires

Duration of time in seconds when a repeat dial operation remains active.

Default Value:

1800

GenerateCPCSignal

Control when to generate CPC Signal to the PHONE Port.

Default Value:

For Inbound and Outbound Calls

EnablePHONEPortBargeIn

Enables the caller to barge in when he calls the other PHONE port from this PHONE port while the other PHONE port has an active call in progress, on-hold, or ringing.

Default Value:

true

UseForPagingOnly

Enables the device to be used for paging only when the PHONE port is connected to an external PA system (via a RJ11 to line out connector, available from many electronics shops). In such configuration the PHONE port is expected to be "off-hook" all the time. The device automatically answers an incoming call and won't accept call-waiting.

Default Value:

false

RemovePowerWhenNotPaging

Remove power to the phone port when not paging if UseForPagingOnly is enabled

Default Value:

false

IncomingCallDurationLimit

If not zero, this is the call duration limit in seconds to apply to all incoming calls that are answered

Default Value:

0

UseForFaxOnly

Assume all calls are for FAXing

Default Value:

false

DisableEcanForAllFax

Disable echo cancellation for fax calls

Default Value:

false

FaxTxGainOffset

FXS transmit (from ATA to the phone) gain offset for fax calls in dB

Default Value:

-15

FaxRxGainOffset

FXS receive (from the phone to the ATA) gain offset for fax calls in dB

Default Value:

-6

UseForModemOnly

Assume modem calls only

Default Value:

false

FaxDetectionMethod

Method for detecting FAX calls for the purpose of determining if T.38 is appropriate

Default Value:

CNG or CED

TransferWhenHolding

This option provides a short cut to transfer a call to a fixed preconfigured number without dialing it. If a valid number is specified for this parameter, the device transfers the call to the given number when the phone hook flashes and then on-hook (which normally leaves the call holding if this parameter is not specified). The valid number should be a complete number with trunk information, such as SP1 (14083334567).

EndConfWhenHangUp

End 3-way (local mixing) conference call when hang up, instead of transferring one remote party to the other

Default Value:

false

EndHoldingCallWhenHangUp

If enabled, when a user hangs up while a call is still on hold, the device ends that call instead of alerting the same to the user (with a short ring).

Default Value:

false

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 response codes on outbound calls to trigger SIT w/ 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.

Default Value:

([4-9]xx)

PlaySITWithAnnouncement

Enables including announcement of the error when an outbound call has failed.

Default Value:

true

AcceptIncomingPage

If set to ""Only when Idle"", non-emergency incoming pages are ignored when the handset is on a call. If set to ""Never"", non-emergency pages are ignored even when the handset is idle.

Default Value:

Always

JoinPageGroup N (N=1, 2, ..., 10)

Join Page Group

Default Value:

false

Call Forwarding

ForwardAll

Enable call forward immediately at all time

Default Value:

false

ForwardAllNumber

Directory number to forward for all incoming call to this phone port

Default Value:

ForwardOnBusy

Enable call forward when the phone port is busy

Default Value:

true

ForwardOnBusyNumber

Directory number to forward when this phone port is busy

Default Value:

ForwardOnNoAnswer

Enable call forward when call is not answered after a certain number of rings

Default Value:

true

ForwardOnNoAnswerNumber

Directory number to forward when there is no answer from this phone port

Default Value:

ForwardOnNoAnswerRingCount

Number of rings to trigger call-forward-on-no-answer

Default Value:

4

Timers

HookFlashTimeMax

Hook Flash is a quick transition of the phone's hook switch from Off-Hook state to On-Hook state, and back to Off-Hook state.

This parameter specifies the upper time limit in milliseconds such that if the hook switch stays at the intermediate On-Hook state for longer than this time limit, the device won't recognize the state transition as a HOOK FLASH event, but instead as an ON HOOK event followed by an OFF HOOK event.

Default Value:

900

HookFlashTimeMin

Hook Flash is a quick transition of the phone's hook switch from Off-Hook state to On-Hook state, and back to Off-Hook state.

This parameter specifies the lower time limit in milliseconds such that if the hook switch stays at the intermediate On-Hook state for less than this time limit, the device won't recognize the state transition as a HOOK FLASH event, but consider the hook switch remains at Off-Hook state throughout the transition (in other words, the transition is discarded as a glitch if it happens too quickly).

Default Value:

70

ReorderDelayTime

Delay in ms to start reorder tone after peer hangs up

Default Value:

5500

CPCDelayTime

A short delay in milliseconds before the device generates a CPC signal to the PHONE port after the far end has hung up during a call.

Default Value:

2000

CPCDuration

The device generates CPC (Calling Party Control) Signal by removing power from the PHONE port for a short period. This parameter specifies the length of this period in milliseconds. CPC signal tells the attached phone equipment that the far end has ended the call.

Default Value:

500

DigitMapLongTimer

Value of the long inter-digit timer (in seconds) when collecting dialed digits according to the **DigitMap** on this PHONE port. This timer governs the timeout when one or more patterns are

partially matched or a variable length pattern (that can accommodate one or more optional digits) is matched.

Default Value:

10

DigitMapShortTimer

Value of the short inter-digit timer (in seconds) when collecting dialed digits according to the **DigitMap** on this PHONE port. This timer governs the timeout when a fixed length pattern has been matched while one or more other patterns can be potentially matched with more input digits.

Default Value:

2

MWILedTimer

Interval in seconds between blinks on the Phone LED for MWI (when phone is onhook). Setting the value to 0 disables LED blinking for MWI

Default Value:

0

Tip Ring Voltage Polarity

IdlePolarity

Tip/Ring voltage polarity the line is idle, before a call is connected, or after one side hangs up.

Choose from `Forward` or `Reverse`.

Default Value:

`Forward`

ConnectPolarity

Tip/Ring voltage polarity when the line is connected on a call. Choose from `Forward` or `Reverse`.

Note: By using a different polarity for an Idle and a Connected line, the device effectively generates a polarity reversal signal to the PHONE port, which signals the attached phone equipment that the call is either connected or ended.

Default Value:

`Forward`

OriginatingSeizurePolarity

The polarity when making an outgoing call

Default Value:

`Same As IdlePolarity`

TerminatingSeizurePolarity

The polarity when ringing the phone on incoming call

Default Value:

`Same As IdlePolarity`

Tone Profile A Parameters

This table lists phone profile A parameters.

Dial Tone

ToneName

(Read-only) Dial Tone

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

350-18, 440-18; 20

Ringback Tone

ToneName

(Read-only) Ringback Tone.

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18, 480-18; -1; (2+4)

Busy Tone

ToneName

(Read-only) Busy Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

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

Reorder Tone

ToneName

(Read-only) Reorder tone or Fast busy.

TonePattern

Obihai Tone Pattern Script.

Default Value:

480-18, 620-18; 10; (.25+.25)

Confirmation Tone

ToneName

Confirmation Tone.

Default Value:

Not configurable.

TonePattern

(Read-only) Obihai Tone Pattern Script.

Default Value:

600-18; 1; (.2+.2)

Holding Tone

ToneName

(Read-only) Holding Tone played when peer holding the call.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:
800-18;30;(.1+10)

Second Dial Tone

ToneName
(Read-only) Second Dial Tone played when dialing second call in a 3-way call.

Default Value:
Not configurable.

TonePattern
Obihai Tone Pattern Script.

Default Value:
385-18,484-18;20

Stutter Tone

ToneName
(Read-only) Stutter Dial Tone.

TonePattern
Obihai Tone Pattern Script.

Default Value:
350-18,440-18;20;2(.1+.1);()

Howling Tone

ToneName
(Read-only) Howling Tone for off-hook warning.

Default Value:
Not configurable.

TonePattern
Obihai Tone Pattern Script.

Default Value:
350-18,440-18;20;2(.1+.1);()

Prompt Tone

ToneName
(Read-only) Prompt Tone to prompt user to enter a number for configuration, such as speed dial.

TonePattern
Obihai Tone Pattern Script.

Default Value:
480-16;20

Call Forwarded Dial Tone

ToneName
(Read-only) Call Forwarded Dial Tone: A special dial tone to indicate call-forward-all is active.

Default Value:
Not configurable.

TonePattern
Obihai Tone Pattern Script.

Default Value:

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

DND Dial Tone

ToneName

(Read-only) DND Dial Tone: A special dial tone to indicate DND is active.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

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

Conference Tone

ToneName

(Read-only) Conference Tone (indicates a 3-way conference call has started).

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

350-16;10;(.1+.1,.1+9.7)

SIT Tone 1

ToneName

(Read-only) Special Information Tone 1.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

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

SIT Tone 2

ToneName

(Read-only) Special Information Tone 2.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

914-16,1371-16,1777-16;20;(1/.274+0,2/.274+0,4/.380+0,0/0+4)

SIT Tone 3

ToneName

(Read-only) Special Information Tone 3.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

914-16,1428-16,1777-16;20;(1/.380+0,2/.380+0,4/.380+0,0/0+4)

SIT Tone 4

ToneName

(Read-only) Special Information Tone 4.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

985-16,1371-16,1777-16;20;(1/.380+0,2/.380+0,4/.380+0,0/0+4)

Outside Dial Tone

ToneName

(Read-only) Outside Dial Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

385-16;10

R-Command Tone

ToneName

(Read-only) R-Command Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

400-16;5

Paging Tone

ToneName

(Read-only) Paging Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

480-16;1;(.2+.2)

Callback Tone

ToneName

(Read-only) Callback Tone

Default Value:
Not configurable

TonePattern
OBIIHAI Tone Pattern Script
Default Value:
Profile A: 440-18, 480-18; 30; (2+4)

Tone Profile B Parameters

This table lists profile B parameters.

Dial Tone

ToneName
(Read-only) Dial Tone
Default Value:
Not configurable.

TonePattern
Obihai Tone Pattern Script.
Default Value:
400-18,425-18,450-18;20

Ringback Tone

ToneName
(Read-only) Ringback Tone.

TonePattern
Obihai Tone Pattern Script.
Default Value:
400-18,425-18,450-18;-1;(.4+.2,.4+2)

Busy Tone

ToneName
(Read-only) Busy Tone.
Default Value:
Not configurable.

TonePattern
Obihai Tone Pattern Script.
Default Value:
425-18;10; (.4+.4)

Reorder Tone

ToneName
(Read-only) Reorder tone or Fast busy.

TonePattern
Obihai Tone Pattern Script.
Default Value:
425-18;10; (.2+.2)

Confirmation Tone

ToneName

Confirmation Tone.

Default Value:

Not configurable.

TonePattern

(Read-only) Obihai Tone Pattern Script.

Default Value:

600-18;1; (.2+.2)

Holding Tone

ToneName

(Read-only) Holding Tone played when peer holding the call.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

800-18;30; (.1+10)

Second Dial Tone

ToneName

(Read-only) Second Dial Tone played when dialing second call in a 3-way call.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

425-18;20

Stutter Tone

ToneName

(Read-only) Stutter Dial Tone.

TonePattern

Obihai Tone Pattern Script.

Default Value:

400-18,425-18,450-18;20;2(.1+.04);()

Howling Tone

ToneName

(Read-only) Howling Tone for off-hook warning.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

480+3,620+3;10; (.125+.125)

Prompt Tone

ToneName

(Read-only) Prompt Tone to prompt user to enter a number for configuration, such as speed dial.

TonePattern

Obihai Tone Pattern Script.

Default Value:

480-16;20

Call Forwarded Dial Tone

ToneName

(Read-only) Call Forwarded Dial Tone: A special dial tone to indicate call-forward-all is active.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

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

DND Dial Tone

ToneName

(Read-only) DND Dial Tone: A special dial tone to indicate DND is active.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

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

Conference Tone

ToneName

(Read-only) Conference Tone (indicates a 3-way conference call has started).

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

425-16;10;(1+15,.36+15)

SIT Tone 1

ToneName

(Read-only) Special Information Tone 1.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

425-16;20;(2.5+.5)

SIT Tone 2

ToneName

(Read-only) Special Information Tone 2.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

914-16,1371-16,1777-16;20;(1/.274+0,2/.274+0,4/.380+0,0/0+4)

SIT Tone 3

ToneName

(Read-only) Special Information Tone 3.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

914-16,1428-16,1777-16;20;(1/.380+0,2/.380+0,4/.380+0,0/0+4)

SIT Tone 4

ToneName

(Read-only) Special Information Tone 4.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

985-16,1371-16,1777-16;20;(1/.380+0,2/.380+0,4/.380+0,0/0+4)

Outside Dial Tone

ToneName

(Read-only) Outside Dial Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

385-16;10

R-Command Tone

ToneName

(Read-only) R-Command Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

400-16;5

Paging Tone

ToneName

(Read-only) Paging Tone.

Default Value:

Not configurable.

TonePattern

Obihai Tone Pattern Script.

Default Value:

480-16;1;(.2+.2)

Callback Tone

ToneName

(Read-only) Callback Tone

Default Value:

Not configurable

TonePattern

OBIHAI Tone Pattern Script

Default Value:

Profile A: 440-18,480-18;30;(2+4)

Codec Profile Parameters

This table lists codec profile parameters.

G711U Codec

Codec

Codec name.

Default Value:

PCMU

BitRate

Bit rate in bits/sec.

Note: Informational only, not configurable.

Default Value:

64000

Enable

Enables this codec.

Default Value:

true

SilenceSuppression

Enables silence suppression for this codec.

Default Value:

false

PacketizationPeriod

Packet size in ms.

Default Value:

20

Priority

Priority assigned to this codec (1 is the highest).

Default Value:

3

PayloadType

Standard payload type for this codec.

Note: Informational only, not configurable.

Default Value:

0

FaxPayloadType

Special payload type reserved for G711U during up-speed for FAX pass through

Default Value:

0

G726R32 Codec

Codec

Codec Name

Default Value:

G726-32

BitRate

Bit rate in bits/sec

Default Value:

32000

Enable

Enable this codec

Default Value:

true

SilenceSuppression

Enable silence suppression for this codec

Default Value:

false

PacketizationPeriod

Packet size in ms

Default Value:

20

Priority

Priority assigned to this codec

Default Value:

7

G711A Codec

Codec

Codec name.

Default Value:

PCMA

BitRate

Bit rate in bits/sec.

Note: Informational only, not configurable.

Default Value:

64000

Enable

Enables this codec.

Default Value:

true

SilenceSuppression

Enables silence suppression for this codec.

Default Value:

false

PacketizationPeriod

Packet size in ms.

Default Value:

20

Priority

Priority assigned to this codec (1 is the highest).

Default Value:

4

PayloadType

Standard payload type for G711-alaw.

Note: Informational only, not configurable.

Default Value:

8

FaxPayloadType

Special payload type reserved for G711A during up-speed for FAX pass through

Default Value:

8

G729 Codec

Codec

Codec name.

Default Value:

G729

BitRate

Bit rate in bits/sec.

Note: Informational only, not configurable.

Default Value:

8000

Enable

Enables this codec.

Default Value:

true

SilenceSuppression

Enables silence suppression for this codec.

Default Value:

false

PacketizationPeriod

Packet size in ms.

Default Value:

20

Priority

Priority assigned to this codec (1 is the highest).

Default Value:

5

PayloadType

Standard payload type for G729.

Note: Informational only, not configurable.

Default Value:

18

G722 Codec

Codec

Codec Name

Default Value:

G722

BitRate

Bit rate in bits/sec

Default Value:

64000

Enable

Enable this codec

Default Value:

true

SilenceSuppression

Enable silence suppression for this codec

Default Value:

false

PacketizationPeriod

Packet size in ms

Default Value:

20

Priority

Priority assigned to this codec (1 is the highest)

Default Value:

1

PayloadType

Standard payload type for this codec

Default Value:

9

OPUS Codec

Codec

Codec Name

Default Value:

opus

BitRate

Bit rate in bits/sec

Default Value:

20000

Enable

Enable this codec

Default Value:

true

SilenceSuppression

Enable silence suppression for this codec

Default Value:

false

PacketizationPeriod

Packet size in ms

Default Value:

20

Priority

Priority assigned to this codec (1 is the highest)

Default Value:

2

PayloadType

Standard payload type for OPUS

Default Value:

109

UseInbandFEC

Use inband FEC when appropriate .

Default Value:

False

iLBC Codec

Codec

Codec name.

Default Value:

iLBC

BitRate

Bit rate in bits/sec.

Note: Informational only, not configurable.

Default Value:

13333

Enable

Enables this codec.

Default Value:

false

SilenceSuppression

Enables silence suppression for this codec.

Default Value:

false

PacketizationPeriod

Packet size in ms.

Default Value:

30

Priority

Priority assigned to this codec (1 is the highest).

Default Value:

6

PayloadType

Dynamic Payload type for this codec. Valid range is 96-127.

Default Value:

98

FAX Event

Codec

Codec name. This codec can be used for relaying FAX tone event using RTP.

Default Value:

fax-event

Enable

Enables this codec.

Default Value:

false

PayloadType

Dynamic Payload type to be used to indicate this event.

Default Value:

100

FaxEvents

Comma-separated list of event IDs to indicate (such as CED, CNG).

Default Value:

32

Telephone Event

Codec

Codec Name. This codec is for relaying DTMF events using RTP.

Default Value:

telephone-event

Enable

Enables this codec.

Default Value:

true

PayloadType

Dynamic Payload type to be used for RFC2833 telephone (DTMF) events. Valid range is 96-127.

Default Value:

101

Encap RTP

Codec

Codec Name. This codec is used to encapsulate RTP packets during a packet loopback call.

Default Value:

encaprtp

PayloadType

Dynamic Payload type for this codec. Valid range is 96-127.

Default Value:

107

Loopback Primer

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.

Default Value:

loopbkprimer

PayloadType

Dynamic Payload type for this codec. Valid range is 96-127.

Default Value:

108

Codec Settings

G726BitPacking

G726 bitstream packing order.

Default Value:

big-endian

T38Enable

Enables the use of T38 (FAX Relay).

Default Value:

true

T38Redundancy

Select T.38 FAX Relay Packet Redundancy .

Default Value:

2

T38MaxBitRate

Select T.38 FAX relay maximum bit rate

Default Value:

14400

T38ECM

Enable ECM in T.38 FAX Relay

Default Value:

false

T38Reinvite

Send T38 reinvoke as callee only

Default Value:

Caller or callee

T38ReinviteDelay

As caller, delay sending T.38 re-invoke upon receiving CED.

Default Value:

1

FaxPassThroughCodec

Codec to use for FAX Pass Through.

Default Value:

G711U

Ring Profile A Call Waiting Tone Parameters

This table lists ring profile A call waiting tone parameters.

Call Waiting Tone 1

ToneName

Distinctive Call Waiting Tone 1. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr1

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30; (.25+10)

Call Waiting Tone 2

ToneName

Distinctive Call Waiting Tone 2. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr2

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30; (.1+.1, .3+.1, .1+10)

Call Waiting Tone 3

ToneName

Distinctive Call Waiting Tone 3. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr3

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30; (.1+.1, .1+10)

Call Waiting Tone 4

ToneName

Distinctive Call Waiting Tone 4. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr4

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.1+.1,.1+.1,.1+10)

Call Waiting Tone 5

ToneName

Distinctive Call Waiting Tone 5. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr5

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+.1,.1+.1,.3+10)

Call Waiting Tone 6

ToneName

Distinctive Call Waiting Tone 6. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr1

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.1+.1,.3+.2,.3+10)

Call Waiting Tone 7

ToneName

Distinctive Call Waiting Tone 7. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr2

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+.1,.3+.1,.1+10)

Call Waiting Tone 8

ToneName

Distinctive Call Waiting Tone 8. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:`User-dr3`**TonePattern**

Obihai Tone Pattern Script.

Default Value:`440-18;30;(.3+2)`

Call Waiting Tone 9

ToneName

Distinctive Call Waiting Tone 9. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:`User-dr4`**TonePattern**

Obihai Tone Pattern Script.

Default Value:`440-18;30;(.3+2)`

Call Waiting Tone 10

ToneName

Distinctive Call Waiting Tone 10. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:`User-dr5`**TonePattern**

Obihai Tone Pattern Script.

Default Value:`Profile A: 440-24;-1;(.1+10)`

Ring Profile A Ring Pattern Parameters

This table lists ring profile A ring pattern parameters.

Ring Pattern 1

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:`Bellcore-dr1`**RingPattern**

Obihai tone cadence script.

Default Value:`60;(2+4)`**Type**

(Ready-only) Handset ring type

Default Value:`Handset Default`

Ring Pattern 2

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr2

RingPattern

Obihai tone cadence script.

Default Value:

60; (.3+.2, 1+.2, .3+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 2

Ring Pattern 3

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr3

RingPattern

Obihai tone cadence script.

Default Value:

60; (.8+.4, .8+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 3

Ring Pattern 4

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr4

RingPattern

Obihai tone cadence script.

Default Value:

60; (.4+.2, .3+.2, .8+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 4

Ring Pattern 5

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr5

RingPattern

Obihai tone cadence script.

Default Value:

60; (.2+.2, .2+.2, .2+.2, 1+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 5

Ring Pattern 6

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr1

RingPattern

Obihai tone cadence script.

Default Value:

60; (.2+.4, .2+.4, .2+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 6

Ring Pattern 7

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr2

RingPattern

Obihai tone cadence script.

Default Value:

60; (.4+.2, .4+.2, .4+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 8

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr3

RingPattern

Obihai tone cadence script.

Default Value:

60; (.25+9.75)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 9

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr4

RingPattern

Obihai tone cadence script.

Default Value:

60; (.25+9.75)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 10

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr5

RingPattern

Obihai tone cadence script.

Default Value:

60; (.25+9.75)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 11

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

ntt-car

RingPattern

Obihai tone cadence script.

Default Value:

5; (.4+.6)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Profile B Ring Pattern Parameters

This table lists ring profile B ring pattern parameters.

Ring Pattern 1

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr1

RingPattern

Obihai tone cadence script.

Default Value:

60; (.4+.2, .4+2)

Type

(Ready-only) Handset ring type

Default Value:

Handset Default

Ring Pattern 2

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr2

RingPattern

Obihai tone cadence script.

Default Value:

60; (.3+.2, 1+.2, .3+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 2

Ring Pattern 3

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr3

RingPattern

Obihai tone cadence script.

Default Value:

60; (.8+.4, .8+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 3

Ring Pattern 4

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr4

RingPattern

Obihai tone cadence script.

Default Value:

60; (.4+.2, .3+.2, .8+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 4

Ring Pattern 5

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr5

RingPattern

Obihai tone cadence script.

Default Value:

60; (.2+.2, .2+.2, .2+.2, 1+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 5

Ring Pattern 6

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr1

RingPattern

Obihai tone cadence script.

Default Value:

60; (.2+.4,.2+.4,.2+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 6

Ring Pattern 7

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr2

RingPattern

Obihai tone cadence script.

Default Value:

60; (.4+.2,.4+.2,.4+4)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 8

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr3

RingPattern

Obihai tone cadence script.

Default Value:

60; (.25+9.75)

Type

(Ready-only) Handset ring type

Default Value:

Type 7

Ring Pattern 9

RingName

Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr4

RingPattern

Obihai tone cadence script.

Default Value:

60; (.25+9.75)

Type

(Ready-only) Handset ring type
Default Value:
Type 7

Ring Pattern 10

RingName
Name of the ring. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.
Default Value:
User-dr5

RingPattern
Obihai tone cadence script.
Default Value:
60; (.25+9.75)

Type
(Ready-only) Handset ring type
Default Value:
Type 7

Ring Profile B Call Waiting Tone Parameters

This table lists ring profile B call waiting tone parameters.

Call Waiting Tone 1

ToneName
Distinctive Call Waiting Tone 1. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.
Default Value:
Bellcore-dr1

TonePattern
Obihai Tone Pattern Script.
Default Value:
425-18;30; (.2+.2, .2+4.4)

Call Waiting Tone 2

ToneName
Distinctive Call Waiting Tone 2. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.
Default Value:
Bellcore-dr2

TonePattern
Obihai Tone Pattern Script.
Default Value:
440-18;30; (.1+.1, .3+.1, .1+10)

Call Waiting Tone 3

ToneName

Distinctive Call Waiting Tone 3. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr3

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.1+.1,.1+10)

Call Waiting Tone 4

ToneName

Distinctive Call Waiting Tone 4. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr4

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.1+.1,.1+.1,.1+10)

Call Waiting Tone 5

ToneName

Distinctive Call Waiting Tone 5. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

Bellcore-dr5

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+.1,.1+.1,.3+10)

Call Waiting Tone 6

ToneName

Distinctive Call Waiting Tone 6. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr1

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.1+.1,.3+.2,.3+10)

Call Waiting Tone 7

ToneName

Distinctive Call Waiting Tone 7. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr2

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+.1,.3+.1,.1+10)

Call Waiting Tone 8

ToneName

Distinctive Call Waiting Tone 8. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr3

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+2)

Call Waiting Tone 9

ToneName

Distinctive Call Waiting Tone 9. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr4

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-18;30;(.3+2)

Call Waiting Tone 10

ToneName

Distinctive Call Waiting Tone 10. An incoming SIP INVITE may include the same name in an Alert-Info header to choose this ring.

Default Value:

User-dr5

TonePattern

Obihai Tone Pattern Script.

Default Value:

440-24;-1;(.1+10)

Star Code Profile Parameters

This table lists star code profile parameters.

Code1

Default = Redial Star Code

Default Value:

*07,Redial,call(\$Ldn)

Code2

Default = Call Return Star Code

Default Value:

*69, Call Return, call(\$Lcn)

Code3

Default = Block Caller ID (Persistent) Star Code

Default Value:

*81, Block Caller ID, set(\$Bci,1)

Code4

Default = Unblock Caller ID (Persistent) Star Code

Default Value:

*82, Unblock Caller ID, set(\$Bci,0)

Code5

Default = Block Caller ID Once Star Code

Default Value:

*67, Block Caller ID Once, set(\$Bci1,1)

Code6

Default = Unblock Caller ID Once Star Code

Default Value:

*68, Unblock Caller ID Once, set(\$Ubc1,1)

Code7

Default = Call Forward Unconditional Star Code

Default Value:

*72, Cfwd All, coll(\$Cfan), set(\$Cfa,1)

Code8

Default = Disable Call Forward Unconditional Star Code

Default Value:

*73, Disable Cfwd All, set(\$Cfa, 0)

Code9

Default = Call Forward on Busy Star Code

Default Value:

*60, Cfwd Busy, coll(\$Cfbn), set(\$Cfb,1)

Code10

Default = Disable Call Forward on Busy Star Code

Default Value:

*61, Disable Cfwd Busy, set(\$Cfb, 0)

Code11

Default = Call Forward on No Answer Star Code

Default Value:

*62, Cfwd No Ans, coll(\$Cfnn), set(\$Cfn,1)

Code12

Default = Disable Call Forward on No Answer Star Code

Default Value:

*63, Disable Cfwd No Ans, set(\$Cfn,0)

Code13

Default = Block Anonymous Calls Star Code

Default Value:

*77, Block Anonymous Call, set(\$Bac,1)

Code14

Default = Unblock Anonymous Calls Star Code

Default Value:

*87, Unblock Anonymous Call, set(\$Bac,0)

Code15

Default = Enable Call Waiting Star Code

Default Value:

*56, Enable Call Waiting, set(\$Cwa,1)

Code16

Default = Disable Call Waiting Star Code

Default Value:

*57, Disable Call Waiting, set(\$Cwa,0)

Code17

Default = Do Not Disturb Star Code

Default Value:

*78, Do Not Disturb, set(\$Dnd,1)

Code18

Default = Disable Do Not Disturb Star Code

Default Value:

*79, Disable DND, set(\$Dnd,0)

Code19

Default = Repeat Dial Star Code

Default Value:

*66, Repeat Dial, rpdi(\$Ldn)

Code20

Default = Disable Repeat Dial Star Code

Default Value:

*86, Cancel Repeat Dial, rpdi()

Code21

Default = Set Speed Dial Star Code

Default Value:

*74([1-9][1-9]x), Set Speed Dial, coll(\$Spd[\$Code])

Code22

Default = Check Speed Dial Star Code

Default Value:

*75([1-9][1-9]x), Check Speed Dial, say(\$Spd[\$Code])

Code23

Default = Loopback Media Star Code

Default Value:

*03, Loopback Media, set(\$Lbm,1)

Code24

Default = Loopback RTP Star Code

Default Value:

*04, Loopback RTP Packet, set(\$Lbp,1)

Code25

Default = Force G711u Codec Star Code

Default Value:

*4711, Use G711 Only, set(\$Cdm,3)

Code26

Default = Force G729 Codec Star Code

Default Value:

*4729, Use G729 Only, set(\$Cdm1,4)

Code27

Default = Clear Speed Dial Star Code

Default Value:

*76([1-9][1-9]x), Clear Speed Dial, set(\$Spd[\$Code],)

Code28

Default = Blind Transfer Star Code

Default Value:

*98, Blind Transfer, coll(\$Bxrn)

Code29

Default = Barge In Star Code

Default Value:

*96, Barge In, set(\$Bar1,1)

Code30

Default = Page Group 1 Star Code *01, Page Group 1 Talk, pg1tx

Default Value:

Default = Page Group 1 Star Code *01, Page Group 1 Talk, pg1tx

Code 31

Default = Page Group 2 Star Code

*02, Page Group 2 Talk, pg2tx

Default Value:

Default = Use G722 Only

*4722, Use G722 Only, set(\$Cdm1,512)

Code32

Default = Use OPUS Only

*4678, Use OPUS Only, set(\$Cdm1,1024)

Default Value:

Default = Use OPUS Only

*4678, Use OPUS Only, set(\$Cdm1,1024)

Code33

Default = Set OBiPLUS to Night Mode Star Code (Requires OBiPLUS Subscription)

Default Value:

*11, Night Mode, set(\$Opm,1)

Code34**Code35**

Default = Page Group 1 Listen

Default Value:

*11, Page Group 1 Listen, pg1rx

Code36

Default = Page Group 2 Listen

Default Value:

*12, Page Group 2 Listen, pg2rx

Code37

Code38
Block laster caller
Default Value:
*86, Block last Caller, blst

Code39

Code40

Code41
Default Value:

Code42
Default Value:

Speed Dial Parameters

This table lists speed dial parameters.

Number or URL
Number for the speed dial entry
Default Value:

1
Speed Dial 1

2
Speed Dial 2

3
Speed Dial 3

. . .
...

99
Speed Dial 99

Name
Name for the speed dial entry
Default Value:

1
Name 1
Default Value:

2
Name 2
Default Value:

3
Name 3
Default Value:

. . .
Default Value:

99

Name 99
Default Value:

User-Defined Digit Maps Parameters

This table lists user-defined digit maps parameters.

User-Defined Digit Map 1

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.

Default Value:

ipd

DigitMap

A valid digit map.

Default Value:

(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.

User-Defined Digit MapN(N= 2 to 10)

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.

4 Getting help

Poly is now a part of HP. The joining of Poly and HP will pave the way for us to create the hybrid work experiences of the future.

During the merge of our two organizations, information about Poly products will transition from the [Poly Support](#) site to the [HP® Support](#) site.

The [Poly Documentation Library](#) will continue to host the installation, configuration, and administration guides for Poly products in HTML and PDF format. In addition, the Poly Documentation Library will provide Poly customers with up-to-date status information about the transition of Poly content from the [Poly Support](#) site to the [HP® Support](#) site.

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