Poly ATA 400 Series Parameter Reference Guide

PVOS-L ATA 4.0.1

SUMMARY

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

Legal information

Privacy policy

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Table of contents

1 About this guide

This section provides clarifying information about this guide.

Audience, purpose, and required skills

This guide provides administrators with information about configuring, maintaining, and troubleshooting the featured product.

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.

- Poly ATA 402
- Poly ATA 400

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

Product Name	SKU	Item Number	Item Description
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
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

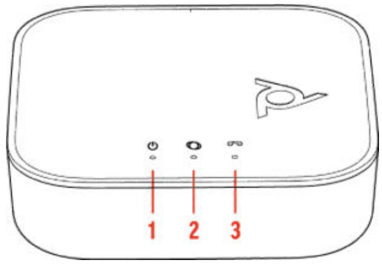

Table 2-3 LED Status Indicators

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

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

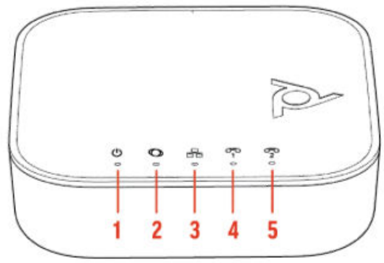

Table 2-4 LED Status Indicators

Reference Number	LED	LED Description
1	Power indicator	The color and pattern of the power indicator shows the following states: LED off: No power Solid green: Powered on and working Flashing green: Searching for a DHCP IP address Flashing orange: Upgrading (Do not unplug power.) Solid red: Not operational
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	The color and pattern of the phone ports indicators show the following states: LED off: Port not enabled Solid green: Phone ready (standby) Flashing green: Phone in use

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

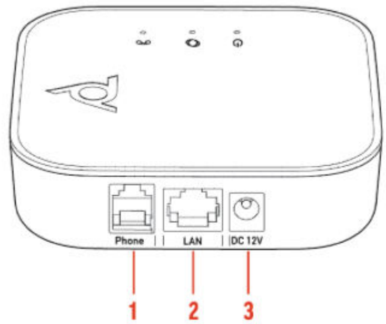

Table 2-5 Port Connections

Reference Number	Port	Port Description
1	Phone Port connection	The **Phone** port on the Poly ATA supports input and output signaling and control messages. You can only connect a touch-tone land line phone to your device's **Phone** port. Phones that use pulse dialing aren't supported.
2	LAN Ethernet port	Use an Ethernet cable to connect the **LAN** port on your device to an Ethernet port on your internet router or switch. By default, the device requests an IP address, DNS, and Internet (WAN) Gateway IP addressing via DHCP.
3	Power connection	Use only the 12-volt power adapter supplied with the original packaging to power the device. 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.

Figure 2-4 Poly ATA 400 Side View

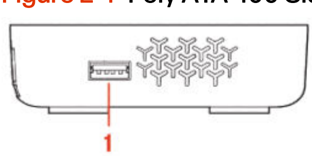

Table 2-6 Port Connections

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

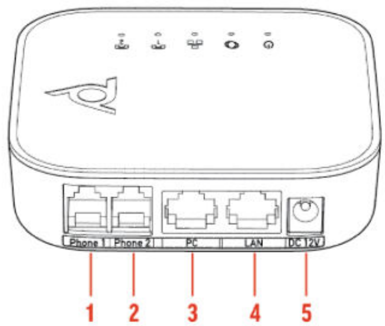

Table 2-7 Port Connections

Reference Number	Port	Port Description
1, 2	Phone Port connections	The **Phone** ports on the Poly ATA support input and output signaling and control messages. You can only connect a touch-tone land line phone to your device's **Phone** ports. Phones that use pulse dialing aren't supported. 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.
3	PC Ethernet port	The device **PC** port enables you to daisy-chain a local device such as a computer.
4	LAN Ethernet port	Use an Ethernet cable to connect the **LAN** port on your device to an Ethernet port on your internet router or switch. By default, the device requests an IP address, DNS, and Internet (WAN) Gateway IP addressing via DHCP.
5	Power connection	Use only the 12-volt power adapter supplied with the original packaging to power the device. 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.

Figure 2-6 Poly ATA 402 Side Callout

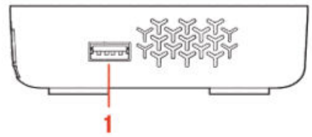

Table 2-8 Port Connections

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.
Connection status indicators	LED indicators provide status and functionality information on the connections between the device and the network or other devices.

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 twn hexdeciaml 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_1XTLSSecurityProfile
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 druation, 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:

TLSSecurityProfile
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://
`TLSSecurityProfile`
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:

`UCSServer then ConfigURL`

`UCSServer`
Whether this SP line is enabled for TR69 Provisioning
Default Value:

`$DHCPOPT160;$DHCPOPT66`

`ConfigURL`
URL of config file.
Default Value:

`$DHCPOPT160/$MAC.xml;$DHCPOPT160/$DM.xml;$DHCPOPT160;tftp://$DHCPOPT66/$DM.xml;$DHCPOPT66/$DM.xml;$DHCPOPT66`

`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 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

`OBN`
Device OBi number, such as `200123456`
Default Value:

ANY

`DSN`
Device S/N, such as `88B01NA00000`
Default Value:

ANY

`GPRM`
Value 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
Specifiy 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:

`(1xxxxxxxxxx|<1>[2-9]xxxxxxxxx|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 *SPn Service* ::URI is specified, additional rules applied in forming the AOR. See the description of the **URI**parameter for more details and examples.

`OutboundProxy`
Host name or IP address of the outbound proxy. Outbound proxying is disabled if this parameter is blank.
`OutboundProxyPort`
Destination port to be used in connecting to the outbound proxy.
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_InsertPPreferredIdentity
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 futher 18x responses w/o SDP upon reveiving 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:
`mtd=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 thrid-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:

`4`

`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>xxxxxxxxx|obxxxxxxxxxx)`

`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 Gatewayn(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 Groupn(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 (Mtg*n*).
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 talkng 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 + 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 O`

`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 transfering 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)`

`PlaySITWithAnnoucement`
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 Fowarding

`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 pone 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`
OBIHAI Tone Pattern Script
Default Value:

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

Tone Profile B Parameters

This table lists pone 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 reinvite as callee only
Default Value:

`Caller or callee`

`T38ReinviteDelay`
As caller, delay sending T.38 re-invite 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($Ubci1,1)

`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($Lbm1,1)

`Code24`
Default = Loopback RTP Star Code
Default Value:

*04, Loopback RTP Packet, set($Lbp1,1)

`Code25`
Default = Force G711u Codec Star Code
Default Value:

*4711, Use G711 Only, set($Cdm1,3)

`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

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