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Poly Voice Software

Trio Parameter Reference Guide

Contents

Before You Begin

The information in this guide applies to the following Poly devices except where noted:

- Poly Trio Visual+ system
- Poly Trio VisualPro system

Related Poly and Partner Resources

See the following sites for information related to this product.

- Poly Support is the entry point to online product, service, and solution support information. Find product-specific information such as Knowledge Base articles, Support Videos, Guide & Manuals, and Software Releases on the Products page, download software for desktop and mobile platforms from Downloads & Apps, and access additional services.
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- With Poly+ you get exclusive premium features, insights and management tools necessary to keep employee devices up, running, and ready for action.
- Poly Lens enables better collaboration for every user in every workspace. It is designed to spotlight the health and efficiency of your spaces and devices by providing actionable insights and simplifying device management.

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Using the Poly Trio Parameter Reference Guide

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

Understanding Device Parameters for Poly Trio

This section provides information about the device parameters for Poly Trio.

Analytics Parameters

This sections provides information about the device analytics parameters for Poly Trio.

Device Analytics Parameters

Use the following parameters to configure device analytics. You can configure the device analytics feature to only enable services of your choice.

`feature.da.enabled`

0 (default) - Disable device analytics.

1 - Enable device analytics.

Change causes system to restart or reboot.

`feature.da.enabled`

1 (default) – Enable the Device Analytics feature.

0 – Disable the Device Analytics feature.

Change causes system to restart or reboot.

`feature.obitalk.enabled`

0 (default) - Disable the connection to the OBiTALK cloud.

1 - Enable the connection to the OBiTALK cloud.

Change causes system to restart or reboot.

`obitalk.accountCode`

Null (default)

String (maximum of 256 characters).

Change causes system to restart or reboot.

`da.supported.services`

Specify the device analytics service to enable.

all (default)

Configure the following strings (maximum of 2048 characters) using a comma-separated list.

ni

service

tsid

pcap

log

config

core

vqmon

cdr

uptimeanalytics

hardwareanalytics

uianalytics

restart

reboot

resettofactory

restapi

Change causes system to restart or reboot.

`deviceAnalytics.note`

Sets the self-note value on the phone and sends to cloud with primary device information message.

Null (default)

String (maximum of 512 characters).

Poly Cloud Connector Parameters

To send device analytics details to Poly Cloud Services, you must enable the Poly Cloud Connector for Poly Trio systems.

`feature.pcc.enabled`

0 - Disables the Poly Cloud Connector for Poly Trio systems.

1 (default)- Enables the Poly Trio system to interface with the Poly Cloud Connector.

Change causes system to restart or reboot.

`pcc.url`

Set the URL for the Poly Cloud Connector interface.

https://api-global.plcm.cloud/globaldirectory (default)

0 - 256

Change causes system to restart or reboot.

`pcc.accountCode`

Enter the Poly Cloud Connector account code to connect your device with a Poly Cloud Services account.

Null (default)

0 - 256

Change causes system to restart or reboot.

Audio Parameters

This section provides information about the audio parameters for Poly Trio.

Acoustic Echo Cancellation Parameters

The following list includes the parameters you can use to set up Acoustic Echo Cancellation (AEC).

`voice.aec.hf.enable`

1 (default) - Enables the AEC function for hands-free options.

0 - Disables the AEC function for hands-free options.

Poly doesn’t recommend disabling this parameter.

`voice.aec.hs.enable`

0 - Disables the AEC function for the handset.

1 (default) - Enables the AEC function for the handset.

Audio Alert Parameters

Use the parameters in the following list to configure audio alerts and sound effects.

`se.appLocalEnabled`

Enables or disables audio alerts and sound effects.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`se.destination`

chassis (default) - Alerts and sound effects play through the phone's speakerphone.

headset - If connected, alerts and sounds play through the headset.

handset active - Alerts play from the destination that is currently in use. For example, if a user is in a call on the handset, a new incoming call rings through the handset.

`se.stutterOnVoiceMail`

1 (default) - A stuttered dial tone is used instead of a normal dial tone to indicate that one or more voicemail messages are waiting at the message center.

0 - A normal tone is used to indicate that one or more voicemail messages are waiting at the message center.

`se.touchFeedback.enabled`

Play sound effect when certain buttons are pressed, like microphone mute.
0 (default) - Does not play an alert tone when the mute status is changed.

1 - An alert tone is played when the mute status is changed.

`call.mute.reminder.period`

The time interval in seconds to play an alert tone periodically when the phone is in the mute state. The parameter `se.touchFeedback.enabled` must be set to 1 in order for this behavior to be applied.

Null (default) - No reminder tone is played.

5-3600

Audio Codec Parameters

You can configure a set of codec properties to improve consistency and reduce workload on the phones.

Use the following parameters to specify audio codec priority on your phones.

- Permitted values to set audio codec priority are 1 - 35
- A value of 1 is the highest priority, 35 the lowest.
- If 0 or Null, the codec is disabled.
- A change to the default value does not cause a phone to restart or reboot

If a phone does not support a codec, the phone treats the value as 0, does not offer or accept calls using that codec, and continues to the codec next in priority.

`voice.codecPref.G711_A`

7 (default)

`voice.codecPref.G711_Mu`

6 (default)

`voice.codecPref.G719.32kbps`

0 (default)

`voice.codecPref.G719.48kbps`

0 (default)

`voice.codecPref.G719.64kbps`

0 (default)

`voice.codecPref.G722`

4 (default)

`voice.codecPref.G7221.16kbps`

0 (default)

`voice.codecPref.G7221.24kbps`

0 (default)

`voice.codecPref.G7221_C.24kbps`

0 (default)

`voice.codecPref.G7221.32kbps`

5 (default)

`voice.codecPref.G7221_C.32kbps`

0 (default)

`voice.codecPref.G7221_C.48kbps`

2 (default)

`voice.codecPref.G729_AB`

8 (default)

`voice.codecPref.iLBC.13_33kbps`

0 (default)

`voice.codecPref.iLBC.15_2kbps`

0 (default)

`voice.codecPref.Lin16.8ksps`

0 (default)

`voice.codecPref.Lin16.16ksps`

0 (default)

`voice.codecPref.Lin16.32ksps`

0 (default)

`voice.codecPref.Lin16.44_1ksps`

0 (default)

`voice.codecPref.Lin16.48ksps`

0 (default)

`voice.codecPref.Siren7.16kbps`

0 (default)

`voice.codecPref.Siren7.24kbps`

0 (default)

`voice.codecPref.Siren7.32kbps`

0 (default)

`voice.codecPref.Siren14.24kbps`

0 (default)

`voice.codecPref.Siren14.32kbps`

0 (default)

`voice.codecPref.Siren14.48kbps`

3 (default)

`voice.codecPref.Siren22.32kbps`

0 (default)

`voice.codecPref.Siren22.48kbps`

0 (default)

`voice.codecPref.Siren22.64kbps`

1 (default)

`voice.codecPref.SILK.8ksps`

0 (default)

`voice.codecPref.SILK.12ksps`

0 (default)

`voice.codecPref.SILK.16ksps`

0 (default)

`voice.codecPref.SILK.24ksps`

0 (default)

Audio Output Options Parameters

The following table includes the parameters you can use to set the audio routing options for the Poly Trio solution.

`up.audio.networkedDevicePlayout`

PhoneOnly (default) - Use the Poly Trio system speakers.

TvOnly - Use the monitor speakers and external speakers (if present).

Auto - Audio-only calls use the Poly Trio system speakers. Video calls use the monitor speakers and external speakers (if present).

`feature.usb.device.hostOs`

Specify the operating system of the computer you are connecting by USB when using Poly Trio as an audio output device.

Windows (default) - The computer connected by USB to the Poly Trio uses a Windows operating system.

Other—The operating system of the computer connected via USB to the Poly Trio system is other than Windows or Mac.

Mac—The computer connected by USB to the Poly Trio uses a Mac operating system.

Confirm—The user is prompted to confirm the computer's operating system each time a USB cable is used to connect to the Poly Trio system.

Comfort Noise Payload Packets Parameters

The following list includes the parameters you can use to configure Comfort Noise payload packets.

`voice.CNControl`

Publishes support for Comfort Noise in the SDP body of the INVITE message and includes the supported comfort noise payloads in the media line for audio.

1 – Either the payload type 13 for 8 KHz sample rate audio codec is sent for Comfort Noise, or the dynamic payload type for 16 KHz audio codecs are sent in the SDP body.

0 (default) – Does not publish support or payloads for Comfort Noise.

`voice.CN16KPayload`

Alters the dynamic payload type used for Comfort Noise RTP packets for 16 KHz codecs.

96 to 127

122 (default)

DTMF Tone Parameters

The following list includes the parameters you can use to set up DTMF tones.

`reg.1.telephony`

Allow telephony services for inbound and outbound calls.

1 (default) – Allowed

0 – Disallowed

`tone.dtmf.chassis.masking`

0 (default) - DTMF tones play through the speakerphone in handsfree mode.

1 - Set to 1 only if `tone.dtmf.viaRtp` is set to 0. DTMF tones are substituted with non-DTMF pacifier tones when dialing in handsfree mode to prevent tones from broadcasting to surrounding telephony devices or inadvertently transmitted in-band due to local acoustic echo.

Change causes system to restart or reboot.

`tone.dtmf.level`

The level of the high frequency component of the DTMF digit measured in dBm0; the low frequency tone is two dB lower.

-15

-33 to 3

Change causes system to restart or reboot.

`tone.dtmf.offTime`

When a sequence of DTMF tones is played out automatically, specify the length of time in milliseconds (ms) the phone pauses between digits. This is also the minimum inter-digit time when dialing manually.

50 (default)

1 – Indefinite

Change causes system to restart or reboot.

`tone.dtmf.onTime`

Set the time in milliseconds (ms) DTMF tones play on the network when DTMF tones play automatically. The time you set is also the minimum time the tone plays when manually dialing.

50 (default)

1 - 65535

Change causes system to restart or reboot.

`tone.dtmf.rfc2833Control`

Specify if the phone uses RFC 2833 to encode DTMF tones.

1 (default) - The phone indicates a preference for encoding DTMF through RFC2833 format in its Session Description Protocol (SDP) offers by showing support for the phone-event payload type. This doesn't affect SDP answers and always honor the DTMF format present in the offer.

0 - The phone doesn't offer dynamic payload for RFC2833 phone-event.

Change causes system to restart or reboot.

`tone.dtmf.rfc2833Payload`

Specify the phone-event payload encoding in the dynamic range to be used in SDP offers.

Generic (default) -127

96 to 127

Change causes system to restart or reboot.

`tone.dtmf.rfc2833Payload_OPUS`

Sets the DTMF payload required to use Opus codec.

126 (default)

96 - 127

Change causes system to restart or reboot.

`tone.dtmf.viaRtp`

1 (default) - Encode DTMF in the active RTP stream. Otherwise, DTMF may be encoded within the signaling protocol only when the protocol offers the option.

0 – If you set this parameter to 0, you must set `tone.dtmf.chassis.masking` to 1.

Change causes system to restart or reboot.

`tone.localDtmf.onTime`

Set the time in milliseconds (ms) DTMF tones play for when the phone plays out a DTML tone sequence automatically.

50 (default)

1 - 65535

`tone.dtmf.rfc2833.SupportOpusClockRate`

1 – (default) Publishes the Telephone-event DTMF frequency as 48000 Hz along with 8000 Hz on Opus codec.

0 - Publishes the Telephone-event DTMF frequency as 8000 Hz on Opus codec.

IEEE 802.1p/Q Parameters

Use the following list to set values for IEEE 802.1p/Q parameters.

You can configure the user_priority specifically for RTP and call control packets, such as SIP signaling packets, with default settings configurable for all other packets.

The phone tags all Ethernet packets it transmits with an 802.1Q VLAN header when the following occurs:

- A valid VLAN ID specified in the phone's network configuration.
- The phone is instructed to tag packets through Cisco Discovery Protocol (CDP) running on a connected Ethernet switch.
- A VLAN ID is obtained from DHCP or CDP.

`qos.ethernet.other.user_priority`

Set user priority for packets without a per-protocol setting.

2 (Default)

0 - 7

`qos.ethernet.rtp.video.user_priority`

Set user-priority used for Video RTP packets.

5 (Default)

0 - 7

`qos.ethernet.rtp.user_priority`

Choose the priority of voice Real-Time Protocol (RTP) packets.

5 (Default)

0 - 7

`qos.ethernet.callControl.user_priority`

Set the user-priority used for call control packets.

5 (Default)

0 - 7

Opus Audio Codec Parameters

Use the following parameters to configure the Opus audio codec.

`voice.audioProfile.Opus.appType`

Assign the Opus encoder’s application type.

VoIP (Default) - process signal for improved speech intelligibility.

Audio - favors faithfulness to original input audio.

LowDelay - configures the minimum possible coding delay by disabling certain modes of operation.

`voice.audioProfile.Opus.BitrateMode`

Sets the preferred encoder transmit bit rate mode. Also controls what is sent in the SDP offer using the `CBR` parameter.

CVBR (default) – Constrained Variable Bit Rate

CBR – Constant Bit Rate

VBR - Variable Bit Rate

`voice.audioProfile.Opus.decInbandFECEnable`

Enables decoding of any received FEC information from the far end.

0 (default) - All FEC information is ignored.

1- All information is received and decoded.

`voice.audioProfile.Opus.encComplexity`

Sets the Opus encoder complexity. A higher value allows for greater encoder complexity. Increased complexity increases processing requirements.

7 (default)

0-10

`voice.audioProfile.Opus.encDTXEnable`

0 (default) – Disables the encoder discontinuous transmit (DTX) mode in the Opus codec.

1 – The encoder skips packet TX during periods of silence and only sends periodic frames with comfort noise information.

`voice.audioProfile.Opus.encExpectedPktLossPercent`

Helps the Opus encoder decide what amount of redundant information to send when in-band FEC is enabled using the parameter `voice.audioProfile.Opus.encInbandFECEnable`.

0 (default)

0 -100

`voice.audioProfile.Opus.encInbandFECEnable`

0 (default) - Disable encoder in-band FEC (Forward Error Correction) for the Opus codec.

1 - The encoder adds redundant information about the previous packet to the current output packet and determines whether to use FEC based on the expected packet loss percentage and the channel's capacity.

Configure the amount of redundant information to send using the parameter `voice.audioProfile.Opus.encExpectedPktLossPercent`.

`voice.audioProfile.Opus.encMaxAvgBitrateKbps`

Communicates to the far end the preferred maximum average bit rate (in kbps) for the Opus encoder.

24 (default)

8 - 510

`voice.audioProfile.Opus.MaxPTime`

Sets the maximum duration of media represented by a packet (in milliseconds).

10

20 (default)

`voice.audioProfile.Opus.pTime`

Sets the preferred duration of media represented by a packet (in milliseconds (ms)).

10

20 (default)

Polycom NoiseBlock Parameters

Use the following parameters to configure NoiseBlock.

`voice.ns.hf.block`

1 (default) - Enables NoiseBlock.

0 - Disables NoiseBlock.

Ringtone Parameters

Use the following parameters to configure ringtones.

`se.rt.enabled`

Enables or disables ringtone feature.

0 - Disabled

1 (default) - Enabled

`se.rt.modification.enabled`

Controls whether or not users are allowed to modify the predefined ringtone from the phone's user interface.

0 - Users not allowed.

1 (default) - Users allowed.

`se.rt.<ringClass>.callWait`

The call waiting tone used for the specified ring class. The call waiting pattern should match the pattern defined in Supported Ring Classes.

callWaiting (default)

callWaitingLong

precedenceCallWaiting

`se.rt.<ringClass>.name`

The answer mode for a ringtone, which is used to identify the ringtone in the user interface.

UTF-8 encoded string

0-127 characters

`se.rt.<ringClass>.ringer`

The ringtone used for this ring class. The ringer must match one listed in Ringtones.

default

ringer1 to ringer24

ringer2 (default)

`se.rt.<ringClass>.timeout`

The duration of the ring in milliseconds before the call is auto-answered, which only applies if the type is set to ring-answer.

1 to 60000

2000 (default)

`se.rt.<ringClass>.type`

Set the answer mode for a ringtone.

ring

visual

answer

ring-answer

Sampled Audio File Parameter

Your custom sampled audio files must be available at the path or URL specified in the parameter `saf.x` so the phone can download the files. Make sure to include the name of the file and the .wav extension in the path.

`saf.x`

Specify a path or URL for the phone to download a custom audio file (x).

To use a Welcome sound, enable the parameter `up.welcomeSoundEnabled` and specify a file in `saf.x`. The default Welcome sound file is `Welcome.wav`.

Null (default) – The phone uses a built-in file.

Path Name – During start-up, the phone attempts to download the file at the specified path in the provisioning server.

URL – During start-up, the phone attempts to download the file from the specified URL on the Internet. Must be a RFC 1738-compliant URL to an HTTP, FTP, or TFTP wave file resource.

Note: If using TFTP, the URL must be in the following format: `tftp://<host>/[pathname]<filename>`. For example: `tftp://somehost.example.com/sounds/example.wav`.

SILK Audio Codec Parameters

Use the following parameters to configure the SILK audio codec.

`voice.codecPref.SILK.8ksps`

Set the SILK audio codec preference for the supported codec sample rates.

0 (default)

`voice.codecPref.SILK.12ksps`

Set the SILK audio codec preference for the supported codec sample rates.

`voice.codecPref.SILK.16ksps`

Set the SILK audio codec preference for the supported codec sample rates.

0 (default)

`voice.codecPref.SILK.24ksps`

Set the SILK audio codec preference for the supported codec sample rates.

0 (default)

`voice.audioProfile.SILK.8ksps.encMaxAvgBitrateKbps`

Set the maximum average encoder output bitrate in kilobits per second (kbps) for the supported SILK sample rate.

20 kbps (default)

6 – 20 kbps

`voice.audioProfile.SILK.12ksps.encMaxAvgBitrateKbps`

Set the maximum average encoder output bitrate in kilobits per second (kbps) for the supported SILK sample rate.

25 kbps (default)

7 – 25 kbps

`voice.audioProfile.SILK.16ksps.encMaxAvgBitrateKbps`

Set the maximum average encoder output bitrate in kilobits per second (kbps) for the supported SILK sample rate.

30 kbps (default)

8 – 30 kbps

`voice.audioProfile.SILK.24ksps.encMaxAvgBitrateKbps`

Set the maximum average encoder output bitrate in kilobits per second (kbps) for the supported SILK sample rate.

40 kbps (default)

12 to 40 kbps

`voice.audioProfile.SILK.encComplexity`

Specify the SILK encoder complexity. The higher the number the more complex the encoding allowed.

2 (default)

0 to 2

`voice.audioProfile.SILK.encDTXEnable`

0 (default) – Disable Enable Discontinuous transmission (DTX).

1 - Enable DTX in the SILK encoder. Note that DTX reduces the encoder bitrate to 0bps during silence.

`voice.audioProfile.SILK.encExpectedPktLossPercent`

Set the SILK encoder expected network packet loss percentage.

A non-zero setting allows less inter-frame dependency to be encoded into the bitstream, resulting in increasingly larger bitrates but with an average bitrate less than that configured with voice.audioProfile.SILK.*.

0 (default)

0 to 100

`voice.audioProfile.SILK.encInbandFECEnable`

0 (default) - Disable inband Forward Error Correction (FEC) in the SILK encoder.

A non-zero value here causes perceptually important speech information to be sent twice: once in the normal bitstream and again at a lower bitrate in later packets, resulting in an increased bitrate.

`voice.audioProfile.SILK.MaxPTime`

Specify the maximum SILK packet duration in milliseconds (ms).

20 ms

`voice.audioProfile.SILK.MinPTime`

Specify the minimum SILK packet duration in milliseconds (ms).

20 ms

`voice.audioProfile.SILK.pTime`

The recommended received SILK packet duration in milliseconds (ms).

20 ms

Sound Effect Pattern Parameters

There are three categories of sound effect patterns that you can use to replace `cat` in the parameter names: `callProg` (Call Progress Patterns), `ringer` (Ringer Patterns) and `misc` (Miscellaneous Patterns).

Keep the following in mind when using the parameters:

- X is the pattern name.
- Y is the instruction number.
- Both x and y need to be sequential.
- `Cat` is the sound effect pattern category.

`se.pat.<cat>.x.inst.y.type`

Sound effects name, where <*cat*> is `callProg` , `ringer` , or `misc` .

sample

chord

silence

branch

`se.pat.<cat>.x.inst.y.value`

sampled – Sampled audio file number

chord – Type of sound effect

silence – Silence duration in milliseconds

branch – Number of instructions to advance

String

USB Audio Call Parameters

The following table includes the parameters you can use to configure USB audio calls for connected devices.

`feature.usb.device.audio`

0 – Disables the ability to use Trio as a USB speakerphone.

1 (default) – Enables the ability to use Poly Trio as a USB speakerphone.

Requires restart

`device.baseProfile`

NULL (default)

Generic - Disables the Skype for Business graphic interface.

Lync - Use this Base Profile for Skype for Business deployments.

USBOptimized - Use this Base Profile when you want to connect Poly Trio to a Microsoft Room System or a Microsoft Surface Hub.

`voice.usb.holdResume.enable`

0 (default) - The Hold and Resume buttons do not display during USB calls.

1 - The Hold and Resume buttons display during USB calls.

This parameter applies only when Poly Trio Base Profile is set to `USBOptimized`.

Voice Activity Detection Parameters

The following list includes the parameters you can use to configure Voice Activity Detection.

`voice.vad.signalAnnexB`

1 (default) - Annex B is used and a new line is added to SDP depending on the setting of `voice.vadEnable`. If `voice.vadEnable` is set to 1, add parameter line `a=fmtp:18 annexb="yes"` below `a=rtpmap` parameter line (where "18" could be replaced by another payload).

0 There is no change to SDP. If `voice.vadEnable` is set to 0, add parameter line – `a=fmtp:18 annexb="no"` below the `a=rtpmap`… parameter line (where "18" could be replaced by another payload).

`voice.vadEnable`

0 (Default) - Disable Voice activity detection (VAD).

1 - Enable VAD.

`voice.vadThresh`

The threshold for determining what is active voice and what is background noise in dB. Sounds louder than this set value are considered active voice, and sounds quieter than this threshold are considered background noise. This does not apply to G.729AB codec operation which has its own built-in VAD function.

15 (default)

Integer from 0 - 30

VQMon Parameters

The parameters listed in the following list configure Voice Quality Monitoring.

`voice.qualityMonitoring.collector.alert.moslq.threshold.critical`

Specify the threshold value of listening MOS score (MOS-LQ) that causes the phone to send a critical alert quality report. Configure the desired MOS value multiplied by 10.

For example, a value of 28 corresponds to the MOS score 2.8.

0 (default) - Critical alerts are not generated due to MOS-LQ.

0 - 40

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.alert.moslq.threshold.warning`

Specify the threshold value of listening MOS score (MOS-LQ) that causes phone to send a warning alert quality report. Configure the desired MOS value multiplied by 10.

For example, a configured value of 35 corresponds to the MOS score 3.5.

0 (default) - Warning alerts are not generated due to MOS-LQ.

0 - 40

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.alert.delay.threshold.critical`

Specify the threshold value of one way-delay (in milliseconds) that causes the phone to send a critical alert quality report.

One-way delay includes both network delay and end system delay.

0 (default) - Critical alerts are not generated due to one-way delay.

0 - 2000 ms

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.alert.delay.threshold.warning`

Specify the threshold value of one-way delay (in milliseconds) that causes the phone to send a critical alert quality report.

One-way delay includes both network delay and end system delay.

0 (default) - Warning alerts are not generated due to one-way delay.

0 - 2000 ms

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.enable.periodic`

0 (default) - Periodic quality reports are not generated.

1 - Periodic quality reports are generated throughout a call.

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.enable.session`

1 (default) - Reports are generated at the end of each call.

0 - Quality reports are not generated at the end of each call.

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.enable.triggeredPeriodic`

0 (default) - Alert states do not cause periodic reports to be generated.

1 - Periodic reports are generated if an alert state is critical.

2 - Period reports are generated when an alert state is either warning or critical.

Note: This parameter is ignored when `voice.qualityMonitoring.collector.enable.periodic` is 1, since reports are sent throughout the duration of a call.

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.period`

The time interval (in milliseconds) between successive periodic quality reports.

20 (default)

5 - 900 seconds

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.server.x.address`

The server address of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages.

Set x to 1 as only one report collector is supported at this time.

NULL (default)

IP address or hostname

Change causes system to restart or reboot.

`voice.qualityMonitoring.collector.server.x.outboundProxy.address`

This parameter directs SIP messages related to voice quality monitoring to a separate proxy. No failover is supported for this proxy, and voice quality monitoring is not available for error scenarios.

NULL (default)

IP address or FQDN

`voice.qualityMonitoring.collector.server.x.outboundProxy.port`

Specify the port to use for the voice quality monitoring outbound proxy server.

0 (default)

0 to 65535

`voice.qualityMonitoring.collector.server.x.outboundProxy.transport`

Specify the transport protocol the phone uses to send the voice quality monitoring SIP messages.

DNSnaptr (default)

TCPpreferred

UDPOnly

TLS

TCPOnly

`voice.qualityMonitoring.collector.server.x.port`

Set the port of a SIP server (report collector) that accepts voice quality reports contained in SIP PUBLISH messages.

Set x to 1 as only one report collector is supported at this time.

5060 (default)

1 to 65535

`voice.qualityMonitoring.failover.enable`

1 (default) - The phone performs a failover when voice quality SIP PUBLISH messages are unanswered by the collector server.

0 - No failover is performed; note, however, that a failover is still triggered for all other SIP messages.

This parameter is ignored if `voice.qualityMonitoring.collector.server.x.outboundProxy` is enabled.

`voice.qualityMonitoring.location`

Specify the device location or identifier for the phone, for the purposes of aggregation for the local endpoint.. If you do not configure a location value, you must use the default string 'Unknown'.

The phone sends this information in the SIP PUBLISH message, in the "LocalGroup" attribute of the voice quality session report. This parameter is only used when voice.qualityMonitoring.rfc6035.enable="1"

Unknown (default)

`voice.qualityMonitoring.rfc6035.enable`

0 (default) - The existing draft implementation is supported.

1 - Complies with RFC6035.

`voice.qualityMonitoring.rtcpxr.enable`

0 (default) - RTCP-XR packets are not generated.

1 - The packets are generated.

Change causes system to restart or reboot.

Call Control Parameters

This section provides information about the call control parameters for Poly Trio.

Microphone Mute Parameters

The following parameters configure microphone mute status alert tones.

`se.touchFeedback.enabled`

0 (default) - Does not play an alert tone when the mute status is changed on the Poly Trio system.

1 - An alert tone is played when the mute status is changed either from the Poly Trio or far-end system.

`call.mute.reminder.period`

The time interval in seconds to play an alert tone periodically when the Poly Trio system is in the mute state.

5 (default)

5 - 3600

Persistent Microphone Mute Parameter

Use the following parameter to enable persistent microphone mute.

`feature.persistentMute.enabled`

0 - The mute state ends when the active call ends or when the phone restarts.

1 (default)- When a user mutes the microphone during an active call, the microphone remains muted for all following calls until the user unmutes the microphone or the phone restarts.

Answer Incoming Calls with Mute Button Parameter

Use the following parameter to configure your Poly Trio system for answering calls with the Mute buttons.

`up.callAnswerWithMuteButton`

0 (default) – Disables using the Mute buttons to answer calls.

1 – Enables users to answer calls using the Mute buttons.

Calling Party Identification Parameters

Use the parameters in the following list to configure Calling Party Identification.

`up.useDirectoryNames`

1 (default) - The name field in the local contact directory is used as the caller ID for incoming calls from contacts in the local directory. Note: Outgoing calls and corporate directory entries are not matched.

0 - Names provided through network signaling are used for caller ID.

`call.callsPerLineKey`

Set the maximum number of concurrent calls per line key. This parameter applies to all registered lines.

Note: The per-registration parameter `reg.x.callsPerLineKey` overrides this parameter.

12 (default)

1 - 12

Remote Party Caller ID from SIP Messages Parameters

Use the following parameters to specify which SIP request and response messages to use to retrieve caller ID information.

`voIpProt.SIP.CID.request.sourceSipMessage`

Specify which header in the SIP request to retrieve remote party caller ID from. You can use:

- `voIpProt.SIP.callee.sourcePreference`
- `voIpProt.SIP.caller.sourcePreference`
- `voIpProt.SIP.CID.sourcePreference`

UPDATE takes precedence over the value of this parameter.

NULL (default) - Remote party caller ID information from INVITE is used.

INVITE

PRACK

ACK

0-6

This parameter does not apply to shared lines.

`voIpProt.SIP.CID.response.sourceSipMessage`

Specify which header in the SIP request to retrieve remote party caller ID from. You can use:

- `voIpProt.SIP.callee.sourcePreference`
- `voIpProt.SIP.caller.sourcePreference`
- `voIpProt.SIP.CID.sourcePreference`

NULL (default) - The remote party caller ID information from the last SIP response is used.

100, 180, 183, 200

0-3

This parameter does not apply to shared lines.

SIP Header Warning Parameters

You can use the parameters in the following list to enable the warning display or specify which warnings to display.

`voIpProt.SIP.header.warning.enable`

0 (default) - The warning header is not displayed.

1 - The warning header is displayed if received.

`voIpProt.SIP.header.warning.codes.accept`

Specify a list of accepted warning codes.

Null (default) - All codes are accepted. Only codes between 300 and 399 are supported.

For example, if you want to accept only codes 325 to 330:
`voIpProt.SIP.header.warning.codes.accept=325,326,327,328,329,330`

Distinctive Call Waiting Parameters

You can use the alert-info values and class fields in the SIP header to map calls to distinct call-waiting types when the phone is already on a call.

`voIpProt.SIP.alertInfo.x.class`

Specify a ringtone for single registered line using a string to match the Alert-Info header in the incoming INVITE.

NULL (default)

`voIpProt.SIP.alertInfo.x.value`

Alert-Info fields from INVITE requests are compared as many of these parameters as are specified (x=1, 2, ...,22) and if a match is found, the behavior described in the corresponding ring class is applied.

default (default)

Do Not Disturb Parameters

Use the parameters in the following list to configure the local DND feature.

`feature.doNotDisturb.enable`

1 (default) - Enable Do Not Disturb (DND).

0 - Disable Do Not Disturb (DND).

Change causes system to restart or reboot.

`feature.doNotDisturb.disableAfterEmergencyCall`

You can configure phones to disable local DND after a user places an emergency call. Emergency services personnel can call back in case the call disconnected or further information is needed without the phone's DND settings interfering. While local DND is disabled in this manner, a popup will display notifying the user that DND is disabled for a period of time.

0 (default)

1 - Disable DND after an emergency call is made.

Change causes a system to restart or reboot.

`feature.doNotDisturb.disableAfterEmergencyCall.timeout`

5 (default)

5-60 - Number of minutes that the phone automatically disables DND after an emergency call is made.

`feature.doNotDisturb.disableAfterEmergencyCall.title`

"911 Emergency Call Mode" (default)

0-127 - String character limit for the title of the popup window when DND is disabled after an emergency call.

`feature.doNotDisturb.disableAfterEmergencyCall.contentDisabled`

"DND is disabled as the phone is in the 911 emergency call back window for 5 mins." (default)

0-256 - String character limit for the content of the popup window when DND is disabled after an emergency call.

`feature.doNotDisturb.disableAfterEmergencyCall.contentEnabled`

"Exiting Emergency call back mode, DND Feature is now Enabled." (default)

0-256 - String character limit for the content of the popup window when DND is re-enabled after an emergency call.

`voIpProt.SIP.serverFeatureControl.dnd`

0 (default) - Disable server-based DND.

1 - Server-based DND is enabled. Server and local phone DND are synchronized.

`voIpProt.SIP.serverFeatureControl.localProcessing.dnd`

This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.dnd` .

If set to 1 (default) and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, the phone and the server perform DND.

If set to 0 and `voIpProt.SIP.serverFeatureControl.dnd` is set to 1, DND is performed on the server-side only, and the phone does not perform local DND.

If both `voIpProt.SIP.serverFeatureControl.localProcessing.dnd` and `voIpProt.SIP.serverFeatureControl.dnd` are set to 0, the phone performs local DND and the `localProcessing` parameter is not used.

1 (default) - Enabled

0 - Disabled

`call.rejectBusyOnDnd`

When enabled, the phone rejects incoming calls with a busy signal while Do Not Disturb is on. When disabled, the phone gives a visual alert of incoming calls, but no audible ring, when Do Not Disturb is on.

1 (default)- Enabled

0 - Disabled

Note: This parameter does not apply to shared lines since not all users may want DND enabled.

Change causes system to restart or reboot.

`call.donotdisturb.perReg`

This parameter determines if the do-not-disturb feature applies to all registrations on the phone or on a per-registration basis.

0 (default) - DND applies to all registrations on the phone.

1 - Users can activate DND on a per-registration basis.

Note: If `voIpProt.SIP.serverFeatureControl.dnd` is set to 1 (enabled), this parameter is ignored.

`call.shared.displayAlertWhenDnd`

When the phone is set to Do Not Disturb (DND) mode, users can disable visual call notifications for incoming intercom calls using this parameter.

0 - Disable call notifications.

1 (default) - Enable call notifications.

Remote Party Disconnect Alert Tone Parameter

You can configure this feature by using the parameter below.

`call.remoteDisconnect.toneType`

Choose an alert tone to play when the remote party disconnects call.

Silent (Default)

messageWaiting, instantMessage, remoteHoldNotification, localHoldNotification, positiveConfirm, negativeConfirm, welcome, misc1, misc2, misc3, misc4, misc5, misc6, misc7, custom1, custom2, custom3, custom4, custom5, custom6, custom7, custom8, custom9, custom10

Call Waiting Alert Parameters

Use the parameters in the following list to configure call waiting alerts.

`call.callWaiting.enable`

Enable or disable call waiting.

1 (default) - The phone alerts you to an incoming call while you are in an active call. If 1, and you end the active call during a second incoming call, you are alerted to the second incoming call.

0 - You are not alerted to incoming calls while in an active call. The incoming call is treated as if you did not answer it.

`call.callWaiting.ring`

Specifies the ringtone of incoming calls when another call is active. If no value is set, the default value is used.

beep (default) - A beep tone plays through the selected audio output mode on the active call.

ring - The configured ringtone plays on the speaker.

silent - No ringtone.

Incoming Call LED Indicator Parameter

Use the following parameter to flash the phone's LED indicator to flash when receiving an incoming call.

`call.offering.led`

0 (default) - The LED doesn't flash when receiving an incoming call from a call server.

1 - The LED flashes when receiving an incoming call from a call server.

Missed Call Notification Parameters

Use the following list to configure options for missed call notifications.

In the following parameters, replace x with the line registration index.

`call.missedCallTracking.x.enabled`

1 (default) - Missed call tracking for a specific registration is enabled.

0 - The missed call counter doesn't update regardless of how you configure `call.serverMissedCalls.x.enabled` or the server. The missed call list doesn't display in the phone menu.

If `call.missedCallTracking.x.enabled="1"` and `call.serverMissedCalls.x.enabled="0"`, then the number of missed calls increments regardless of how you configure the server.

If `call.missedCallTracking.x.enabled="1"` and `call.serverMissedCalls.x.enabled="1"`, then the handling of missed calls depends on how you configure the server.

Change causes system to restart or reboot.

`call.serverMissedCall.x.enabled`

0 (default) - All missed-call events increment the counter for a specific registration.

1 - Only missed-call events sent by the server increment the counter.

Note: This feature is supported only with the BroadSoft Synergy call server (previously known as Sylantro).

Change causes system to restart or reboot.

`call.serverMissedCall.led`

0 (default) - The LED doesn't flash if there is a missed call on the call server.

1 - The LED flashes when there is a missed call on the call server.

Call Hold Parameters

See the following list for the available parameters you can use to configure for Call Hold.

`voIpProt.SIP.useRFC2543hold`

0 (default) - SDP media direction parameters (such as a=sendonly) per RFC 3264 when initiating a call.

1 - the obsolete c=0.0.0.0 RFC2543 technique is used when initiating a call.

`voIpProt.SIP.useSendonlyHold`

1 (default) - The phone will send a reinvite with a stream mode parameter of "`sendonly`" when a call is put on hold.

0 - The phone will send a reinvite with a stream mode parameter of "`inactive`" when a call is put on hold

Note: The phone will ignore the value of this parameter if set to 1 when the parameter `voIpProt.SIP.useRFC2543hold` is also set to 1 (default is 0).

`call.hold.localReminder.enabled`

0 (default) - Users are not reminded of calls that have been on hold for an extended period of time.

1 - Users are reminded of calls that have been on hold for an extended period of time.

Change causes system to restart or reboot.

`call.hold.localReminder.period`

Specify the time in seconds between subsequent hold reminders.

60 (default)

Change causes system to restart or reboot.

`call.hold.localReminder.startDelay`

Specify a time in seconds to wait before the initial hold reminder.

90 (default)

Change causes system to restart or reboot.

`voIpProt.SIP.musicOnHold.uri`

A URI that provides the media stream to play for the remote party on hold. This parameter is used if `reg.x.musicOnHold.uri` is Null.

Null (default)

SIP URI

Call Transfer Parameters

Use the following list to specify call transfer behavior.

`call.defaultTransferType`

Set the transfer type the phone uses when transferring a call.

Generic Base Profile: Consultative (default) - Users can immediately transfer the call to another party.

Call Forwarding Parameters

Use the parameters in the following list to configure feature options for call forwarding.

`feature.forward.enable`

1 (default) - Enables call forwarding.

0 - Disables call forwarding. Users cannot use Call Forward and the option is removed from the phone's Features menu.

`voIpProt.SIP.serverFeatureControl.cf`

0 (default) - The server-based call forwarding is not enabled.

1 - The server-based call forwarding is enabled.

Change causes system to restart or reboot.

`voIpProt.SIP.serverFeatureControl.localProcessing.cf`

This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.cf` .

1 (default) - If set to 1 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, the phone and the server perform call forwarding.

0 - If set to 0 and `voIpProt.SIP.serverFeatureControl.cf` is set to 1, call forwarding is performed on the server side only, and the phone does not perform local call forwarding.

If both `voIpProt.SIP.serverFeatureControl.localProcessing.cf` and `voIpProt.SIP.serverFeatureControl.cf` are set to 0, the phone performs local call forwarding and the `localProcessing` parameter is not used.

`voIpProt.SIP.header.diversion.enable`

0 (default) - If set to 0, the diversion header is not displayed.

1 - If set to 1, the diversion header is displayed if received.

Change causes system to restart or reboot.

`voIpProt.SIP.header.diversion.list.useFirst`

1 (default) - If set to 1, the first diversion header is displayed.

0 - If set to 0, the last diversion header is displayed.

Change causes system to restart or reboot.

`divert.x.contact`

All automatic call diversion features uses this forward-to contact. All automatically forwarded calls are directed to this contact. The contact can be overridden by a busy contact, DND contact, or no-answer contact as specified by the `busy`, `dnd`, and `noAnswer` parameters that follow.

Null (default)

string - Contact address that includes ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com).

Change causes system to restart or reboot.

`divert.x.sharedDisabled`

1 (default) - Disables call diversion features on shared lines.

0 - Enables call diversion features on shared lines.

Change causes system to restart or reboot.

`divert.x.autoOnSpecificCaller`

1 (default) - Enables the auto divert feature of the contact directory for calls on registration x. You can specify to divert individual calls or divert all calls.

0 - Disables the auto divert feature of the contact directory for registration x.

Change causes system to restart or reboot.

`divert.busy.x.enabled`

1 (default) - Diverts calls registration x is busy.

0 - Does not divert calls if the line is busy.

Change causes system to restart or reboot.

`divert.busy.x.contact`

Calls are sent to the busy contact's address if it is specified; otherwise calls are sent to the default contact specified by `divert.x.contact .`

Null (default)string - contact address.

Change causes system to restart or reboot.

`divert.dnd.x.enabled`

0 (default) - Divert calls when DND is enabled on registration x.

1 - Does not divert calls when DND is enabled on registration x.

Change causes system to restart or reboot.

`divert.dnd.x.contact`

Calls are sent to the DND contact's address if it is specified; otherwise calls are sent to the default contact specified by `divert.x.contact` .

Null (default)

string - contact address.

Change causes system to restart or reboot.

`divert.fwd.x.enabled`

1 (default) - Users can forward calls on the phone's Home screen and use universal call forwarding.

0 - Users cannot enable universal call forwarding (automatic forwarding for all calls on registration x).

Change causes system to restart or reboot.

`divert.noanswer.x.enabled`

1 (default) - Unanswered calls after the number of seconds specified by timeout are sent to the no-answer `contact`.

0 - Unanswered calls are diverted if they are not answered.

Change causes system to restart or reboot.

`divert.noanswer.x.contact`

Null (default) - The call is sent to the default contact specified by `divert.x.contact`.

string - contact address

Change causes system to restart or reboot.

`divert.noanswer.x.timeout`

55 (default) - Number of seconds for timeout.

positive integer

Change causes system to restart or reboot.

`reg.x.fwd.busy.contact`

The forward-to contact for calls forwarded due to busy status.

Null (default) - The contact specified by `divert.x.contact` is used.

string - The contact specified by `divert.x.contact` is not used

`reg.x.fwd.busy.status`

0 (default) - Incoming calls that receive a busy signal is not forwarded

1 - Busy calls are forwarded to the contact specified by `reg.x.fwd.busy.contact`.

`reg.x.fwd.noanswer.contact`

Null (default) - The forward-to contact specified by `divert.x.contact` is used.

string - The forward to contact used for calls forwarded due to no answer.

`reg.x.fwd.noanswer.ringCount`

The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20.

0 - (default)

1 to 65535

`reg.x.fwd.noanswer.status`

0 (default) - The calls are not forwarded if there is no answer.

1 - The calls are forwarded to the contact specified by `reg.x.noanswer.contact` after ringing for the length of time specified by `reg.x.fwd.noanswer.ringCount` .

`reg.x.serverFeatureControl.cf`

This parameter overrides `voIpProt.SIP.serverFeatureControl.cf`.

0 (default) - The server-based call forwarding is disabled.

1 - server based call forwarding is enabled.

Change causes system to restart or reboot.

`voIpProt.SIP.serverFeatureControl.cf`

0 (default) - Disable server-based call forwarding.

1 - Enable server-based call forwarding.

This parameter overrides `reg.x.serverFeatureControl.cf`.

Change causes system to restart or reboot.

`voIpProt.SIP.serverFeatureControl.localProcessing.cf`

1 (default) - Allows to use the value for `voIpProt.SIP.serverFeatureControl.cf`.

0 - Does not use the value for

This parameter depends on the value of `voIpProt.SIP.serverFeatureControl.cf`.

`reg.x.serverFeatureControl.localProcessing.cf`

This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.cf`.

0 - If `reg.x.serverFeatureControl.cf` is set to 1 the phone does not perform local Call Forward behavior.

1 (default) - The phone performs local Call Forward behavior on all calls received.

`call.shared.disableDivert`

1 (default) - Enable the diversion feature for shared lines.

0 - Disable the diversion feature for shared lines. Note that this feature is disabled on most call servers.

Change causes system to restart or reboot.

Automatic Off-Hook Call Placement Parameters

As shown in the following list, you can specify an off-hook call contact, enable or disable the feature for each registration, and specify a protocol for the call.

You can specify only one line registration for the Poly Trio system.

In the following parameters, replace x with the line registration index.

`call.autoOffHook.x.contact`

Enter a SIP URL contact address. The contact must be an ASCII-encoded string containing digits, either the user part of a SIP URL (for example, 6416), or a full SIP URL (for example, `6416@polycom.com`).

NULL (default)

`call.autoOffHook.x.enabled`

0 (default) - No call is placed automatically when the phone goes off hook, and the other parameters are ignored.

1 - When the phone goes off hook, a call is automatically placed to the contact you specify in call.autoOffHook.x.contact and using the protocol you specify in call.autoOffHook.x.protocol.

`call.autoOffHook.x.protocol`

Specify the calling protocol. If no protocol is specified, the phone uses the protocol specified by `call.autoRouting.preferredProtocol`. If a line is configured for a single protocol, the configured protocol is used.

NULL (default)

SIP

H323

Multiple Line Keys Per Registration Parameter

Use the parameter below to configure this feature.

This feature is one of several features associated with Call Appearances.

`reg.x.lineKeys`

Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model.

1 (default)

1 to max

Multiple Call Appearance Parameters

Use the parameters in the following list to set the maximum number of concurrent calls per registered line and the default number of calls per line key.

Note that you can set the value for the reg.1.callsPerLineKey parameter to a value higher than 1, for example, 3. After you set the value to 3, for example, you can have three call appearances on line 1. By default, any additional incoming calls are automatically forwarded to voicemail. If you set more than two call appearances, a call appearance counter displays at the top-right corner on the phone.

`call.callsPerLineKey`

Set the maximum number of concurrent calls per line key. This parameter applies to all registered lines.

Note: The per-registration parameter `reg.x.callsPerLineKey` overrides this parameter.

12 (default)

1 - 12

`reg.x.callsPerLineKey`

Set the maximum number of concurrent calls for a single registration *x*. This parameter applies to all line keys using registration *x*. If registration *x* is a shared line, an active call counts as a call appearance on all phones sharing that registration.

This per-registration parameter overrides the `call.callsPerLineKey` parameter.

12 (default)

1 - 12

Bridged Line Appearance Parameters

To begin using Bridged Line Appearance, you must get a registered address dedicated for use with your call server provider.

This dedicated address must be assigned to a phone line in the `reg.x.address` parameter.

Use the parameters in the following list to configure this feature.

`reg.x.type`

private (default) - Use standard call signaling.

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

`reg.x.thirdPartyName`

Null (default) - In all other cases.

string address -This field must match the `reg.x.address` value of the registration which makes up the part of a bridged line appearance (BLA).

`voIpProt.SIP.blaGlareHonorRetryAfter`

Controls the Retry mechanism.

1 (default) - The phone honors the Retry-after header on glare and sends NOTIFY with the same state and line-id after the requested time interval.

0 - The phone ignores the Retry-after header on glare and immediately sends NOTIFY with the next available line-id.

Voicemail Parameters

Use the parameters in the following list to configure voicemail and voicemail settings.

`feature.voicemail.enabled`

1 (default) - Enable voicemail.

0 - Disable voicemail.

`msg.mwi.x.callBackMode`

The message retrieval mode and notification for registration x.

registration (default) - The registration places a call to itself (the phone calls itself).

contact - a call is placed to the contact specified by `msg.mwi.x.callback`.

disabled - Message retrieval and message notification are disabled.

`msg.mwi.x.callBack`

The contact to call when retrieving messages for this registration if `msg.mwi.x.callBackMode` is set to `contact .`

ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)

NULL (default)

`msg.mwi.x.subscribe`

Specify the URI of the message center server. ASCII encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL (6416 or 6416@polycom.com)

If non-Null, the phone sends a SUBSCRIBE request to this contact after boot up.

NULL (default)

`mwi.backLight.disable`

Specify if the phone screen backlight illuminates when you receive a new voicemail message.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`up.mwiVisible`

Specify if message waiting indicators (MWI) display or not.

0 (default) - If `msg.mwi.x.callBackMode=0`, MWI do not display in the message retrieval menus.

1 - MWI display.

Change causes system to restart or reboot.

`up.oneTouchVoiceMail`

0 (default) - The phone displays a summary page with message counts.

1 - You can call voicemail services directly from the phone, if available on the call server, without displaying the voicemail summary.

Change causes system to restart or reboot.

Local Call Recording Parameter

Use the following parameter to configure local call recording.

`feature.callRecording.enabled`

0 (default) - Disable audio call recording.

1 - Enable audio call recording.

Change causes system to restart or reboot.

Local and Centralized Conference Call Parameters

The following list includes available call management parameters.

You can specify whether, when the host of a three-party local conference leaves the conference, the other two parties remain connected or disconnected. If you want the other two parties remain connected, the phone performs a transfer to keep the remaining parties connected. If the host of four-party local conference leaves the conference, all parties are disconnected and the conference call ends. If the host of a centralized conference leaves the conference, each remaining party remains connected. For more ways to manage conference calls, see Conference Management.

`call.localConferenceCallHold`

0 (default) - The host cannot place parties on hold.

1 - During a conference call, the host can place all parties or only the host on hold.

`call.transferOnConferenceEnd`

1 (default) - After the conference host exits the conference, the remaining parties can continue.

0 - After the conference host exits the conference, all parties are exited and the conference ends.

`call.singleKeyPressConference`

Specify whether or not all parties hear sound effects while setting up a conference.

0 (default) - Phone sound effects are heard only by the conference initiator.

1 - A conference is initiated when a user presses Conference the first time. Also, all sound effects (dial tone, DTMF tone while dialing and ringing back) are heard by all participants in the conference.

`call.autoConfJoinDelayTimer`

Set the delay, in milliseconds, for joining an incoming and active call into a conference call. The delay helps mitigate participants being left on hold when they aren't properly joined to the conference call.

0 (default)

0-10000 ms

Note: This parameter is not supported on Poly Trio 8300.

`voIpProt.SIP.conference.address`

Null (default) - Conferences are set up on the phone locally.

String 128 max characters - Enter a conference address. Conferences are set up by the server using the conferencing agent specified by this address. Acceptable values depend on the conferencing server implementation policy.

`video.conf.addVideoWhenAvailable`

0 (default) - When Poly Trio system is added to a conference by another participant via digit dialing, the system does not add video.

1 - When Poly Trio system is added to a conference by another participant via digit dialing, the system adds video if video is available on the conference.

Conference Management Parameter

Use the parameter in the following list to configure the conference management feature.

`feature.nWayConference.enabled`

0 - Users can hold three-way conferences but conference management options are not available.

1 (default) - Users can hold conferences with the maximum number of parties, and the conference management options display to enable users to add, hold, mute, and remove participants.

Conference Meeting Dial-In Options Parameters

Use the following parameters to configure the dial-in information.

`exchange.meeting.join.promptWithList`

Specifies the behavior of the Join button on meeting reminder pop-ups.

0 (default) - A meeting reminder does not show a list of numbers to dial.

1 - Tapping Join on a meeting reminder should show a list of numbers to dial rather than immediately dialing the first one.

`exchange.meeting.parseWhen`

Specifies when to scan the meeting's subject, location, and description fields for dialable numbers.

NonSkypeMeeting (default)

Always

Never

Change causes system to restart or reboot.

`exchange.meeting.parseOption`

Select a meeting invite field to fetch a VMR or meeting number from.

All (default)

Location

LocationAndSubject

Description

Change causes a reboot.

`exchange.meeting.parseEmailsAsSipUris`

List instances of text like `user@domain` or `user@ipaddress` in the meeting description or subject under the More Actions pane as dialable SIP URIs.

0 (default) - it does not list the text as a dialable SIP URI

1 - it treats `user@domain` or user@ipaddress as a dialable SIP URI.

Change causes system to restart or reboot.

`exchange.meeting.parseAllowedSipUriDomains`

List of comma-separated domains that will be permitted to be interpreted as SIP URIs

Null (default)

String (maximum of 255 characters)

Change causes system to restart or reboot.

Hybrid Line Registration Parameters

Use the following parameters to configure dial plan and line switching for Hybrid Registration.

`dialplan.digitmap.lineSwitching.enable`

0 (default) - Disable the line switching in dial plan to switch the call to the dial plan matched line.

1 - Enable the line switching in dial plan to switch the call to the dial plan matched line.

This is not applicable for off-hook dialing.

`reg.limit`

Specify the maximum number of lines to use for registration.

1 (default)

- 3 is the maximum of registered lines.
- 12 is the maximum of unregistered lines. For all unregistered lines, make sure to set `reg.x.server.y.register` to **0**.
- Only one H.323 line, registered or unregistered, is supported

`reg.1.mergeServerDigitMapLocally`

1 (default) - Allow the digit map from the in-band provisioning parameter `dialplan.1.digitmap` to merge with the local digit map.

0 - The digit map is not merged.

Local Digit Maps Parameters

Use the following parameters to configure the local digit map.

`dialplan.applyToCallListDial`

Choose whether the dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dialplan.applyToDirectoryDial`

0 (default) - The dial plan is not applied to numbers dialed from the directory or speed dial, including auto-call contact numbers.

1 - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers.

Change causes system to restart or reboot.

`dialplan.applyToForward`

0 (default) - The dial plan does not apply to forwarded calls.

1 - The dial plan applies to forwarded calls.

Change causes system to restart or reboot.

`dialplan.applyToTelUriDial`

Choose whether the dial plan applies to URI dialing.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dialplan.applyToUserDial`

Choose whether the dial plan applies to calls placed when the user presses Dial.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dialplan.applyToUserSend`

Choose whether the dial plan applies to calls placed when the user presses Send.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dialplan.conflictMatchHandling`

Selects the dialplan based on more than one match with the least timeout.

0 (default) - Disabled

1 - Enabled

`dialplan.digitmap.timeOut`

Specify a timeout in seconds for each segment of the digit map using a string of positive integers separated by a vertical bar (|). After a user presses a key, the phone waits this many seconds before matching the digits to a dial plan and dialing the call.

(Default) `3 | 3 | 3 | 3 | 3| 3`

If there are more digit maps than timeout values, the default value 3 is used. If there are more timeout values than digit maps, the extra timeout values are ignored.

Change causes system to restart or reboot.

`dialplan.digitmap`

Specify the digit map used for the dial plan using a string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. This parameter enables the phone to automatically initiate calls to numbers that match a digit map pattern.

The string is limited to 2560 bytes and 100 segments of 64 bytes, and the following characters are allowed in the digit map.

- A comma (,), which turns dial tone back on.
- A plus sign (+) is allowed as a valid digit.
- The extension letter 'R' indicates replaced string.
- The extension letter 'Pn' indicates precedence, where 'n' range is 1-9.
  1 - Low precedence

  9 - High precedence

Change causes system to restart or reboot.

`dialplan.filterNonDigitUriUsers`

Determine whether to filter out (+) from the dial plan.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`dialplan.impossibleMatchHandling`

0 —The digits entered up to and including the point an impossible match occurred are sent to the server immediately.

1—The phone gives a reorder tone.

2 (default) —Users can accumulate digits and dispatch the call manually by pressing Send.

3 — No digits are sent to the call server until the timeout is configured by `dialplan.impossibleMatchHandling.timeout` parameter.

If a call orbit number begins with a pound (#) or asterisk (*), you need to set the value to 2 to retrieve the call using off-hook dialing.

Change causes system to restart or reboot.

`dialplan.removeEndOfDial`

Sets if the trailing # is stripped from the digits sent out.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dialplan.routing.emergency.outboundIdentity`

Choose how your phone is identified when you place an emergency call. The outbound identity is only used when dialing emergency numbers through one of the servers configured in dialplan.routing.server.x.address.

NULL (default)

10-25 digit number

SIP

TEL URI

If using a URI, the full URI is included verbatim in the P-A-I header. For example:

- `dialplan.routing.emergency.outboundIdentity = "5551238000"`
- `dialplan.routing.emergency.outboundIdentity= "sip:john@emergency.com"`
- `dialplan.routing.emergency.outboundIdentity = "tel:+16045558000"`

`dialplan.routing.emergency.preferredSource`

Set the precedence of the source of emergency outbound identities.

ELIN (default)— the outbound identity used in the SIP P-Asserted-Identity header is taken from the network using an LLDP-MED Emergency Location Identifier Number (ELIN).

Config— the parameter `dialplan.routing.emergency.outboundIdentity` has priority when enabled, and the LLDP-MED ELIN value is used if `dialplan.routing.emergency.outboundIdentity` is NULL.

`dialplan.routing.emergency.x.description`

Set the label or description for the emergency contact address.

x=1: Emergency, Others: NULL (default)

string

x is the index of the emergency entry description where x must use sequential numbering starting at 1.

Change causes system to restart or reboot.

`dialplan.routing.emergency.x.server.y`

Set the emergency server to use for emergency routing (`dialplan.routing.server.x.address` where x is the index).

x=1: 1, Others: Null (default)

positive integer

x is the index of the emergency entry and y is the index of the server associated with emergency entry x. For each emergency entry (x), one or more server entries (x,y) can be configured. x and y must both use sequential numbering starting at 1.

Change causes system to restart or reboot.

`dialplan.routing.emergency.x.value`

Set the emergency URL values that should be watched for. When the user dials one of the URLs, the call is directed to the emergency server defined by `dialplan.routing.server.x.address`.

x=1: 911, others: Null (default)

SIP URL (single entry)

x is the index of the emergency entry description where x must use sequential numbering starting at 15.

`dialplan.routing.server.x.address`

Set the IP address or hostname of a SIP server to use for routing calls. Multiple servers can be listed starting with x=1 to 3 for fault tolerance.

Null (default)

IP address

hostname

Blind transfer for 911 or other emergency calls may not work if registration and emergency servers are different entities.

Change causes system to restart or reboot.

`dialplan.routing.server.x.port`

Set the port of a SIP server to use for routing calls.

5060 (default)

1 to 65535

Change causes system to restart or reboot.

`dialplan.routing.server.x.transport`

Set the DNS lookup of the first server to use and dialed if there is a conflict with other servers.

DNSnaptr (default)

TCPpreferred

UDPOnly

TLS

TCPOnly

For example, if `dialplan.routing.server.1.transport` = "UDPOnly" and `dialplan.routing.server.2.transport` = "TLS", then UDPOnly is used.

Change causes system to restart or reboot.

`dialplan.userDial.timeOut`

Set the time, in seconds, the phone waits for digit input before placing a call when the phone is onhook.

0-99 seconds

You can apply `dialplan.userDial.timeOut` only when its value is lower than `up.IdleTimeOut`.

Enhanced 911 (E.911) Parameters

Use the following parameters to configure E.911.

Note: In E.911 configurations which use HELD to determine a phone's location, note that the phone defaults to a 24-hour HELD refresh interval if it can't calculate an expiration interval due to an error, if it doesn't have an SNTP connection, or if the calculated expiration interval is greater than 48 hours.

feature.E911.locationInfoSchema

HYBRID (default) - SIP invites use an XML schema as per the RFC4119 and RFC5139 standards.

RFC 4119 - SIP invites use an XML schema as per the RFC4119 standards.

RFC5139 - SIP invites use an XML schema as per the RFC5139 standards.

feature.E911.HELD.server

NULL (default)

Set the IP address or hostname of the Location Information Server (LIS) address. For example, host.domain.com or xxx.xxx.xxx.xxx.

0 - 255 characters

feature.E911.HELD.username

NULL (default)

Set the user name used to authenticate to the LIS.

0 - 255 characters

feature.E911.HELD.password

NULL (default)

Set the password used to authenticate to the Location Information Server.

0 - 255 characters

feature.E911.HELD.identity

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

NULL (default)

String 255 character max

feature.E911.HELD.identityValue

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

NULL (default)

String 255 character max

feature.E911.locationRetryTimer

Specify the retry timeout value in seconds for the location request sent to the Location Information Server (LIS).

The phone does not retry after receiving location information received through the LIS.

60 seconds (default)

60 - 86400 seconds

feature.E911.HELD.nai.enable

0 (default) – The NAI is omitted as a device identity in the location request sent to the LIS.

1 - The NAI is included as a device identity in the location request sent to the LIS.

`locInfo.source`

Specify the source of phone location information. This parameter is useful for locating a phone in environments that have multiple sources of location information.

LLDP (default for Generic Base Profile) – Use the network switch as the source of location information.

CONFIG - Use location information defined in the configuration.

LIS – Use the location information server as the source of location information. Generic Base Profile only.

DHCP – Use DHCP as the source of location information. Generic Base Profile only.

If location information is not available from a default or configured source, the fallback priority is as follows:

Generic Base Profile: No fallback supported for Generic Base Profile

`locInfo.x.label`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter a label for the location.

Null (default)

0 -255

`locInfo.x.country`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the country where the phone is located.

Null (default)

0 -255

`locInfo.x.A1`

To use this parameter, `set locInfo.source` to `CONFIG`.

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

Null (default)

0 -255

`locInfo.x.A3`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the city where the phone is located.

Null (default)

0 -255

`locInfo.x.PRD`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the leading direction of the street location.

Null (default)

0 -255

`locInfo.x.RD`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the name of road or street where the phone is located.

Null (default)

0 -255

`locInfo.x.STS`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the suffix of the name used in `locInfo.x.RD`. For example, street or avenue.

Null (default)

0 -255

`locInfo.x.POD`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the trailing street direction. For example, southwest.

Null (default)

0 -255

`locInfo.x.HNO`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the street address number of the phone's location.

Null (default)

0 -255

`locInfo.x.HNS`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter a suffix for the street address used in `locInfo.x.HNS`. For example, A or ½.

Null (default)

0 -255

`locInfo.x.LOC`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter any additional information that identifies the location.

Null (default)

0 -255

`locInfo.x.NAM`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter a proper name to associate with the location.

Null (default)

0 -255

`locInfo.x.PC`

To use this parameter, `set locInfo.source` to `CONFIG`.

Enter the ZIP or postal code of the phone's location.

Null (default)

0 -255

`feature.E911.enabled`

0 (default) - Disable the E.911 feature.

1 - Enable the E.911 feature.

The INVITE sent for emergency calls from the phone includes the geolocation header defined in RFC 6442 and PIDF presence element as specified in RFC3863 with a GEOPRIV location object specified in RFC4119 for in Open SIP environments.

`feature.E911.HELD.requestType`

Any (default) - Send a request to the Location Information Server (LIS) to return either 'Location by Reference' or 'Location by Value'. Note this is not the 'Any' value referred to in RFC 5985.

Civic - Send a request to the LIS to return a location by value in the form of a civic address for the device as defined in RFC 5985.

RefID - Send a request to the LIS to return a set of Location URIs for the device as defined in RFC 5985.

`voIpProt.SIP.header.priority.enable`

0 (default) – Do not include a priority header in the E.911 INVITE message.

1 - Include a priority header in the E.911 INVITE message.

`voIpProt.SIP.header.geolocation-routing.enable`

0 (default) – Do not include the geolocation-routing header in the E.911 INVITE message.

1 - Include the geolocation-routing header in the E.911 INVITE message.

`voIpProt.SIP.header.switchInfo.enable`

The phone gathers the MAC address and port information from LLDP and sends that data to the server, which determines phone location based on "Location" configurations.
0 (default) - The register message does not include the custom header `X-switch-info`.

1 - Register messages include the custom header `X-switch-info` that contains the MAC address and port information.

`feature.E911.HELD.secondary.server`

Set the IP address or hostname of the secondary Location Information Server (LIS) address. For example, host.domain.com or xxx.xxx.xxx.xxx.

NULL (default)

0-255

Dotted-decimal IP address

Hostname

Fully-qualified domain name (FQDN)

`feature.E911.HELD.secondary.username`

Set a user name to authenticate to the secondary Location information Server (LIS).

NULL (default)

String

0-255

`feature.E911.HELD.secondary.password`

Set a password to authenticate to the secondary LIS.

NULL (default)

String

`feature.E911.usagerule.retransmission`

0 (default) - The recipient of this location object is not permitted to share the enclosed location information, or the object as a whole, with other parties.

1 - Distributing this location is permitted.

MLPP with AS-SIP Parameters

The following parameters configure MLPP with AS-SIP.

`voIpProt.SIP.assuredService.defaultPriority`

Default priority assigned to an outgoing call.

1 (default)

1 to 10

This value is overridden if priority is assigned from the dial plan for that number.

`voIpProt.SIP.assuredService.enable`

0 (default) - Disables the AS-SIP feature.

1 - Enables the AS-SIP feature

`voIpProt.SIP.assuredService.namespace.custom.name`

The name for the custom namespace label.

Null (default)

String

`voIpProt.SIP.assuredService.namespace.custom.priority.x`

The namespace precedence values, lowest to highest.

Null (default)

String

`voIpProt.SIP.assuredService.precedenceThreshold`

The minimum call priority required for a call to be treated as a precedence call.

2 (default)

1 to 10

`voIpProt.SIP.assuredService.preemptionAutoTerminationDelay.local`

Set the duration after a callee preemption event that a call appearance is automatically cleared.

0 (default)

0- 3600

`voIpProt.SIP.assuredService.preemptionAutoTerminationDelay.remote`

Set the duration after a caller preemption event that a call appearance is automatically cleared.

3 (default)

0-3600

`voIpProt.SIP.assuredService.serverControlled`

1 (default) - The precedence level of outgoing calls is set by the server or non-EI equipment.

0 - The precedence level is set by the phone and must not change if it is an outgoing call.

International Dialing Prefix Parameters

The following parameters configure the international dialing prefixes.

`call.internationalDialing.enabled`

This parameter applies to all numeric dial pads on the phone, including for example, the contact directory.

1 (default) - Disable the key tap timer that converts a double tap of the asterisk "*" symbol to the "+" symbol to indicate an international call. By default, this parameter is enabled so that a quick double tap of "*" converts immediately to "+". To enter a double asterisk "**", tap "*" once and wait for the key tap timer to expire to enter a second "*".

0 - When you disable this parameter, you cannot dial "+" and you must enter the international exit code of the country you are calling from to make international calls.

Change causes system to restart or reboot.

`call.internationalPrefix.key`

The phone supports international call prefix (+) with both "0" and "*".

0 (default) - Set the international prefix with "*".

1 - Set the international prefix with "0".

Certificate Parameters

This section provides information on the certificate parameters for Poly Trio.

Custom URL Location for LDAP Server Certificates Parameter

Use the parameter below to configure a custom URL location for LDAP server certificates.

In addition to the parameter below, you must also configure the following Corporate Directory parameters:

- `sec.TLS.profileSelection.LDAP = ApplicationProfile1`

`sec.TLS.LDAP.customCaCertUrl`

Enter the URL location from where the phone can download LDAP server certificates.

String (default)

0 - Minimum

255 - Maximum

You must configure parameters `dir.corp.address` and `feature.corporateDirectory.enabled` as well to enable this parameter.

Online Certificate Status Protocol Parameter

OCSP is a more advanced protocol than the existing CRL. OCSP further offers a grace period for an expired certificate to access servers for a limited time before certificate renewal. OCSP is disabled by default.

`device.sec.TLS.OCSP.enabled`

Ensure that you set `device.set="1"`, and `device.sec.TLS.OCSP.enabled.set="1"` to enable OCSP.

0 (default) OCSP is disabled.

1 – OCSP is enabled

Change causes system to restart or reboot.

TLS Cipher Suite Parameters

You can use the parameters listed below to configure TLS Cipher Suites.

`sec.TLS.cipherList`

String (1 - 1024 characters)

RC4:@STRENGTH (default)

ALL:!aNULL:!eNULL:!DSS:!SEED

:!ECDSA:!IDEA:!MEDIUM:!LOW:!

EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!

RC4:@STRENGTH

The global cipher list parameter. The format for the cipher list uses OpenSSL syntax found at: https://www.openssl.org/docs/man1.0.2/apps/ciphers.html.

`sec.TLS.<application>.cipherList`

Specify the cipher list for a specific TLS Platform Profile or TLS Application Profile.

TLS Protocol Parameters

The following list includes the parameters for the TLS protocol supported applications.

`device.sec.TLS.protocol.dot1x`

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and 802.1x authentication. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`device.sec.TLS.protocol.prov`

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and provisioning. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`device.sec.TLS.protocol.syslog`

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and Syslog. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`sec.TLS.protocol.browser`

Configure the lowest TLS/SSL version to use for handshake negotiation between the phone and phone browser. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

The microbrowser restarts when there is a change in the browser TLS protocol or TLS cipher settings, and the last web page displayed is not restored.

`sec.TLS.protocol.exchangeServices`

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and Exchange services. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`sec.TLS.protocol.ldap`

Configure the lowest TLS/SSL version to use for handshake negotiation between phone and Lightweight Directory Access Protocol (LDAP). The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`sec.TLS.protocol.sip`

Configures the lowest TLS/SSL version to use for handshake negotiation between the phone and SIP signaling. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`sec.TLS.protocol.sopi`

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and SOPI. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

`sec.TLS.protocol.webServer`

Configures the lowest TLS/SSL version to use for handshake negotiation for the phone's web server.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

sec.TLS.protocol.xmpp

Configures the lowest TLS/SSL version to use for handshake negotiation between phone and XMPP. The phone handshake starts with the highest TLS version irrespective of the value you configure.

TLSv1_0 (default)

SSLv2v3

TLSv1_1

TLSv1_2

TLS Profile Selection Parameters

You can configure the parameters listed below to choose the platform profile or application profile to use for each TLS application.

sec.TLS.profileSelection.browser

Specifies to select a TLS platform profile or TLS application profile for the browser or a microbrowser.

PlatformProfile1 (default)

- PlatformProfile1
- PlatformProfile2
- ApplicationProfile1
- ApplicationProfile2
- ApplicationProfile3
- ApplicationProfile4
- ApplicationProfile5
- ApplicationProfile6
- ApplicationProfile7

sec.TLS.profileSelection.LDAP

Specifies to select a TLS platform profile or TLS application profile for the corporate directory.

PlatformProfile1 (default)

- PlatformProfile1
- PlatformProfile2
- ApplicationProfile1
- ApplicationProfile2
- ApplicationProfile3
- ApplicationProfile4
- ApplicationProfile5
- ApplicationProfile6
- ApplicationProfile7

sec.TLS.profileSelection.SIP

Specifies to select a TLS platform profile or TLS application profile for SIP operations.

PlatformProfile1 (default)

- PlatformProfile1
- PlatformProfile2
- ApplicationProfile1
- ApplicationProfile2

- ApplicationProfile3
- ApplicationProfile4
- ApplicationProfile5
- ApplicationProfile6
- ApplicationProfile7

`sec.TLS.profileSelection.syslog`

Specifies to select a TLS platform profile for the syslog operations.

PlatformProfile1 (default)

PlatformProfile1 or PlatformProfile2

TLS Platform Profile and Application Profile Parameters

By default, all preinstalled profiles are associated with the default cipher suite and use trusted and widely recognized CA certificates for authentication.

The following list shows parameters for TLS Platform Profile 1. To configure TLS Platform Profile 2, use a 2 at the end of the parameter instead of a 1. For example, set `device.sec.TLS.profile.caCertList2` instead of `device.sec.TLS.profile.caCertList1`.

You can use the parameters in the following list to configure the following TLS Profile feature options:

- Change the cipher suite, CA certificates, and device certificates for the two platform profiles and the six application profiles.
- Map profiles directly to the features that use certificates.

`device.sec.TLS.customCaCert1`

Specify a custom certificate.

Null (default)

String (maximum of 12288 characters)

`device.sec.TLS.profile.caCertList1`

Specify which CA certificates to use.

Null (default)

String (maximum of 1024 characters)

`device.sec.TLS.profile.cipherSuite1`

Specify the cipher suite.

Null (default)

String (maximum of 1024 characters)

`device.sec.TLS.profile.cipherSuiteDefault1`

Null (default)

0 - Use the custom cipher suite.

1 - Use the default cipher suite.

`device.sec.TLS.profile.deviceCert1`

Specify which device certificates to use.

Builtin (default)

Builtin, Platform1, Platform2

`sec.TLS.customCaCert.x`

The custom certificate for TLS Application Profile x (x= 1 to 6).

Null (default)

String

`sec.TLS.customDeviceKey.x`

The custom device certificate private key for TLS Application Profile x (x= 1 to 6).

Null (default)

String

`sec.TLS.profile.x.caCert.application1`

1 (default) - Enable a CA Certificate for TLS Application Profile 1.

0 - Disable a CA Certificate for TLS Application Profile 1.

`sec.TLS.profile.x.caCert.application2`

1 (default) - Enable a CA Certificate for TLS Application Profile 2.

0 - Disable a CA Certificate for TLS Application Profile 2.

`sec.TLS.profile.x.caCert.application3`

1 (default) - Enable a CA Certificate for TLS Application Profile 3.

0 - Disable a CA Certificate for TLS Application Profile 3.

`sec.TLS.profile.x.caCert.application4`

1 (default) - Enable a CA Certificate for TLS Application Profile 4.

0 - Disable a CA Certificate for TLS Application Profile 4.

`sec.TLS.profile.x.caCert.application5`

1 (default) - Enable a CA Certificate for TLS Application Profile 5.

0 - Disable a CA Certificate for TLS Application Profile 5.

`sec.TLS.profile.x.caCert.application6`

1 (default) - Enable a CA Certificate for TLS Application Profile 6.

0 - Disable a CA Certificate for TLS Application Profile 6.

`sec.TLS.profile.x.caCert.application7`

1 (default) - Enable a CA Certificate for TLS Application Profile 7.

0 - Disable a CA Certificate for TLS Application Profile 7.

`sec.TLS.profile.x.caCert.defaultList`

Specifies the list of default CA Certificate for TLS Application Profile x (x=1 to 7).

Null (default)

String

`sec.TLS.profile.x.caCert.platform1`

1 (default) - Enable a CA Certificate for TLS Platform Profile 1.

0 - Disable a CA Certificate for TLS Platform Profile 1.

`sec.TLS.profile.x.caCert.platform2`

1 (default) - Enable a CA Certificate for TLS Platform Profile 2.

0 - Disable a CA Certificate for TLS Platform Profile 2.

`sec.TLS.profile.x.cipherSuite`

Specifies the cipher suite for TLS Application Profile x (x=1 to 8).

Null (default)

String

`sec.TLS.profile.x.cipherSuiteDefault`

1 (default) - Use the default cipher suite for TLS Application Profile x (x= 1 to 8).

0 - Use the custom cipher suite for TLS Application Profile x (x= 1 to 8).

`sec.TLS.profile.x.deviceCert`

Specifies the device certificate to use for TLS Application Profile x (x = 1 to 7).

Polycom (default)

Platform1, Platform2, Application1, Application2, Application3, Application4, Application5, Application6,Application7

Configuration Parameters

This section provides information about the configuration parameters for Poly Trio.

Chord Parameters

Chord sets are the sound effect building blocks that use synthesized audio instead of sampled audio.

Poly phones support three chord sets. Each chord set has different chord names, represented by *x* in the following parameters.

- `callProg`, where *x* can be one of the following chord names:
  - `dialTone`
  - `busyTone`
  - `ringback`
  - `reorder`
  - `stutter_3`
  - `callWaiting`
  - `callWaitingLong`
  - `howler`
  - `recWarning`
  - `stutterLong`
  - `intercom`
  - `callWaitingLong`
  - `precedenceCallWaiting`
  - `preemption`
  - `precedenceRingback`
  - `spare1` to `spare6`
- `misc`, where *x* can be one of the following chord names:
  - `spare1` to `spare9`
  - `cs1` to `cs12`
- `ringer`, where *x* can be one of the following chord names:
  - `ringback`
  - `originalLow`
  - `originalHigh`
  - `spare1` to `spare19`

`tone.chord.callProg.x.freq.y` `tone.chord.misc.x.freq.y` `tone.chord.ringer.x.freq.y`

Frequency (in Hertz) for component y. Up to six chord-set components can be specified (y=1 to 6).

0-1600

`tone.chord.callProg.x.level.y` `tone.chord.misc.x.level.y` `tone.chord.ringer.x.level.y`

Level of component y in dBm0. Up to six chord-set components can be specified (y=1 to 6).

-57 to 3

`tone.chord.callProg.x.onDur tone.chord.misc.x.onDur tone.chord.ringer.x.onDur`

On duration (length of time to play each component) in milliseconds.

0=infinite

Positive integer

`tone.chord.callProg.x.offDur tone.chord.misc.x.offDur tone.chord.ringer.x.offDur`

Off duration (the length of silence between each chord component) in milliseconds

0=infinite

Positive integer

`tone.chord.callProg.x.repeat tone.chord.misc.x.repeat tone.chord.ringer.x.repeat`

Number of times each ON/OFF cadence is repeated.

0=infinite

Positive integer

Configuration Request Parameter

Use the following parameter to configure the phone's behavior when it receives a request for restart or reconfiguration.

`request.delay.type`

Specifies whether the phone should restart or reconfigure.

call (default) - The phone executes the request when there are no calls.

audio - The phone executes the request when there is no active audio.

Change causes system to restart or reboot.

Download Location Parameter for Language Files

The following parameter specifies the download location of the translated language files for the system web interface (Web Configuration Utility).

`webutility.language.plcmServerUrl`

Specifies the download location of the translated language files for the system web interface.

http://downloads.polycom.com/voice/software/languages/

(default)

URL

Ethernet Interface MTU Parameters

Use the following parameters to control the Ethernet interface maximum transmission unit (MTU).

`net.interface.mtu`

Configures the Ethernet or Wi-Fi interface maximum transmission unit (MTU).

1496 (default)

800 - 1500

This parameter affects the LAN port and the PC port.

`net.interface.mtu6`

Specifies the MTU range for IPv6.

1500 (default)

1280 - 1500

`net.lldp.extenedDiscovery`

Specifies the duration of time that LLDP discovery continues after sending the number of packets defined by the parameter `lldpFastStartCount` .

0 (default)

0 - 3600

The LLDP packets are sent every 5 seconds during this extended discovery period.

Feature Activation and Deactivation Parameters

Use the feature parameters to control the activation or deactivation of a feature at run time.

`feature.callCenterCallInformation.enable`

1 (default) - The phone displays a full-screen dialog showing call information details. The dialog closes after 40 seconds, or you can press **Exit** to close it and return to the active call screen. You can set how long the dialog displays using the parameter `up.idleTimeout`.

0 - The phone uses the active call screen, and ACD call information is not available.

`feature.callCenterStatus.enabled`

0 (default) - Disable the status event threshold capability.

1 - Enable the status event threshold capability to display at the top of the phone screen.

`feature.computeraudioconnector.enabled`

0 (default) - Disable the computer audio connector feature.

1 - Enable the computer audio connector feature.

`feature.flexibleLineKey.enable`

0 (default) - Disables the Flexible Line Key feature.

1 - Enables the Flexible Line Key feature.

`feature.lclConferenceDtmfRelay.enabled`

0 (default) - Relay DTMF receieved by the host on one leg to another.

1 - Do not relay DTMF receieved by the host on one leg to another.

`feature.nfc.enabled`

0 - Disable NFC.

1 (default) - Enable NFC.

`feature.photoIntegration.enable`

0 - Disable photo intetgration feature.

1 (default) - Enable photo intetgration feature.

`feature.ringDownload.enabled`

1 (default) - The phone downloads ringtones when starting up.

0 - The phone does not download ringtones when starting up.

Change causes system to restart or reboot.

`feature.uniqueCallLabeling.enabled`

0 (default) - Disable Unique Call Labeling.

1 - Enable Unique Call Labeling. Use `reg.x.line.y.label` to define unique labels.

Change causes system to restart or reboot.

`feature.urlDialing.enabled`

1 (default) - URL/name dialing is available from private lines, and unknown callers are identified on the display by their phone's IP address.

0 - URL/name dialing is not available.

`feature.usb.device.enabled`

The USB device port enables you to use a Poly Trio system as an audio device for your laptop.

1 (default) - Enable the USB device port.

0 - Disable the USB device port.

When you disable the Poly Trio system's USB device port using the parameter `feature.usb.device.enabled` , the USB Computer Connections option doesn't display on the phone menu at **Settings > Advanced > Administration Settings > USB Computer Connections**.

`feature.usb.host.enabled`

1 (default) Enable the USB host port on the Poly Trio system.

0 - Disable the USB host port on the Poly Trio system.

Use the host port for memory sticks, mouse, keyboards, and charging your devices.

`reg.x.urlDialing.enabled`

1 (default) - Enable dialing by URL for SIP registrations.

0 - Disable dialing by URL for SIP registrations.

Feature License Parameter

Use the following parameter to configure the feature licensing system.

Once you install a license on a phone, you can’t remove it.

`license.polling.time`

Specifies the time (using the 24-hour clock) to check if the license has expired.

02:00 (default)

00:00 - 23:59

Change causes system to restart or reboot.

General Security Parameters

Use the following parameters to configure security features of the phone.

`sec.tagSerialNo`

Enable to include the phone's serial number (MAC address) in application layer HTTP GET request headers and SIP contact headers.

0 (default) - The phone doesn't provide the serial number (MAC address).

1 - The phone provides the serial number (MAC address).

Change causes system to restart or reboot.

`sec.uploadDevice.privateKey`

0 (default) - While generating the Certificate Signing Request from the phone, the device private key isn’t uploaded to the provisioning server.

1 - The device private key is uploaded to the provisioning server along with the CSR.

DHCP Parameter

Use the following parameter to configure how the phone reacts to DHCP changes.

`tcpIpApp.dhcp.releaseOnLinkRecovery`

Specifies whether or not a DHCP release occurs.

1 (default) - Performs a DHCP release after the loss and recovery of the network.

0 - No DHCP release occurs.

DNS Parameters

Use the following parameters to set the DNS.

The values you set using DHCP have a higher priority, and the values you set using the <device/> parameter in a configuration file have a lower priority.

`tcpIpApp.dns.server`

Phone directs DNS queries to this primary server.

NULL (default)

IP address

Change causes system to restart or reboot.

`tcpIpApp.dns.altServer`

Phone directs DNS queries to this secondary server.

NULL (default)

IP address

Change causes system to restart or reboot.

`tcpIpApp.dns.domain`

Specifies the DNS domain for the phone.

NULL (default)

String

Change causes system to restart or reboot.

`tcpIpApp.dns.address.overrideDHCP`

Specifies how DNS addresses are set.

0 (default) - DNS address requested from the DHCP server.

1 - DNS primary and secondary address is set using the parameters `tcpIpApp.dns.server` and `tcpIpApp.dns.altServer`.

Change causes system to restart or reboot.

Note: If the DHCP server doesn't send the DNS server addresses to the phone, then the values set for the `device.dns.serverAddress` and `device.dns.altSrvAddress` parameters are used. Alternatively, the phone uses the DNS server addresses set using the `tcpIpApp*` parameters, which override `device.dns.*` parameters.

`tcpIpApp.dns.domain.overrideDHCP`

Specifies how the domain name is retrieved or set.

0 (default) - Domain name retrieved from the DHCP server, if one is available.

1 - DNS domain name is set using the parameter `tcpIpApp.dns.domain` parameter.

Change causes system to restart or reboot.

Note: If the DHCP server doesn't send the DNS domain to the phone, then the value set for `device.dns.domain` is used. Alternatively, the phone uses the DNS domain set using the `tcpIpApp*` parameter, which overrides `device.dns.*` parameter.

TCP Keep-Alive Parameters

Use the following parameters to configure TCP keep-alive on SIP TLS connections. The phone can detect a failure quickly (in minutes) and attempt to re-register with the SIP call server or its redundant pair.

`tcpIpApp.keepalive.tcp.idleTransmitInterval`

Specifies the amount of time to wait (in seconds) before sending the keep-alive message to the call server. Range is 10 to 7200.

30 (Default)

If this parameter is set to a value that is out of range, the default value is used.

Specifies the number of seconds TCP waits between transmission of the last data packet and the first keep-alive message.

`tcpIpApp.keepalive.tcp.noResponseTransmitInterval`

Specifies the amount of idle time between the transmission of the keep-alive packets the TCP stack waits on. This applies whether or not the last keep-alive was acknowledged.

If no response is received to a keep-alive message, subsequent keep-alive messages are sent to the call server at this interval (every *x* seconds). Range is 5 to 120.

`tcpIpApp.keepalive.tcp.sip.persistentConnection.enable1`

Specifies whether the TCP socket connection remains open or closes.

0 (Default) - The TCP socket opens a new connection when the phone tries to send any new SIP message and closes after one minute.

1 - The TCP socket connection remains open.

Change causes system to restart or reboot.

`tcpIpApp.keepalive.tcp.sip.tls.enable`

Specifies whether to disable or enable TCP keep-alive for SIP signaling connections.

0 (Default) - Disables TCP keep-alive for SIP signaling connections that use TLS transport.

1 - Enables TCP keep-alive for SIP signaling connections that use TLS transport.

File Transfer Parameter

Use the following parameter to configure file transfers from the phone to the provisioning server.

`tcpIpApp.fileTransfer.waitForLinkIfDown`

Specifies whether a file transfer from the FTP server is delayed or not attempted.

1 (Default) - File transfer from the FTP server is delayed until Ethernet comes back up.

0 - File transfer from the FTP server is not attempted.

HTTPD Web Server Parameters

The phone contains a local system web interface server for user and administrator features.

The web server supports both basic and digest authentication. You can’t configure the authentication user name and password.

`httpd.enabled`

Base Profile = Generic

1 (default) - The web server is enabled.

0 - The web server is disabled.

Change causes system to restart or reboot.

`httpd.cfg.enabled`

Base Profile = Generic

1 (default) - The system web interface is enabled.

0 - The system web interface is disabled.

Change causes system to restart or reboot.

`httpd.cfg.port`

Port is 80 for HTTP servers. Take care when choosing an alternate port.

80 (default)

1 to 65535

Change causes system to restart or reboot.

`httpd.cfg.secureTunnelPort`

The port to use for communications when the secure tunnel is used.

443 (default)

1 to 65535

Change causes system to restart or reboot.

`httpd.cfg.secureTunnelRequired`

1 (default) - Access to the system web interface is allowed only over a secure tunnel (HTTPS) and non-secure (HTTP) is not allowed.

0 - Access to the system web interface is allowed over both a secure tunnel (HTTPS) and non-secure (HTTP).

Change causes system to restart or reboot.

Message Waiting Parameters

Use the following parameters to configure the message-waiting feature, supported on a per-registration basis.

The maximum number of registrations (*x*) for each phone model is listed in the Flexible Call Appearances section under the column "Registrations."

`msg.bypassInstantMessage`

0 (default) - Displays the **Message Center** and **Instant Messages** menus when a user presses the **Messages** or **MSG** key.

1 - Bypasses the menus and goes to voicemail.

`msg.mwi.x.led`

1 (default) - The LED flashes as long as the phone has new unread voicemail messages for any line.

0 - Red MWI LED doesn't flash when there are new unread messages for the selected line.

x is an integer referring to the registration indexed by `reg.x`.

`mwi.sharedLineIcon.enable`

1 (default) – Shows that the message waiting indicator appears for all the registered lines.

0 – The message waiting indicator shows only for the first line appearance if there are multiple lines registered on the phone.

Per-Registration Call Parameters

Poly phones support an optional per-registration feature that enables automatic call placement when the phone is off-hook.

The phones also support a per-registration configuration that determines which events cause the missed-calls counter to increment. You can enable/disable missed call tracking on a per-line basis.

`call.advancedMissedCalls.addToReceivedList`

Applies to calls on that are answered remotely.

0 (default) - Calls answered from the remote phone are not added to the local receive call list.

1 - Calls answered from the remote phone are added to the local

receive call list.

`call.advancedMissedCalls.enabled`

Use this parameter to improve call handling.

1 (default) - Shared lines can correctly count missed calls.

0 - Shared lines may not correctly count missed calls.

`call.advancedMissedCalls.reasonCodes`

Enter a comma-separated list of reason code indexes interpreted to mean that a call should not be considered as a missed call.

200 (default)

`call.autoAnswer.micMute`

1 (default) - The microphone is initially muted after a call is auto-answered.

0 - The microphone is active immediately after a call is auto-answered.

`call.autoAnswer.ringClass`

The ring class to use when a call is to be automatically answered using the auto-answer feature. If you set to a ring class with a type other than `answer` or `ring-answer`, the settings are overridden such that a ringtone of `visual` (no ringer) applies.

ringAutoAnswer (default)

`call.autoAnswer.ringTone`

Intercom (default) – Auto answer plays the intercom tone.

doubleBeep – Auto answer plays the double-beep tone.

`call.autoAnswer.SIP`

0 (default) - Disable auto-answer for SIP calls.

1 - Enable auto-answer for SIP calls.

`call.autoAnswer.ringTone`

intercom (default) – While auto answering a call, phone plays an intercom tone.

doubleBeep – Phone plays the double beep tone.

`call.autoAnswerMenu.enable`

1 (default) - The **Autoanswer** menu displays and is available to the user.

0 - The **Autoanswer** menu is disabled and is not available to the user.

`call.BlindTransferSpecialInterop`

0 (default) - Do not wait for an acknowledgment from the transferee before ending the call.

1 - Wait for an acknowledgment from the transferee before ending the call.

`call.dialtoneTimeOut`

The time is seconds that a dial tone plays before a call is dropped.

60 (default)

0 - The call is not dropped.

Change causes system to restart or reboot.

`call.internationalDialing.enabled`

Use this parameter to enable or disable the key tap timer that converts a double tap of the asterisk (*) symbol to the plus (+) symbol used to indicate an international call.

1 (default) - A quick double tap of * converts immediately to +. To enter a double asterisk (**), tap the asterisk (*) once and wait for the key tap timer to expire to enter a second asterisk (*).

0 - You cannot dial plus (+) symbol and you must enter the international exit code of the country you are calling from to make international calls.

This parameter applies to all numeric dial pads on the phone including for example, the contact directory.

Change causes system to restart or reboot.

`call.internationalPrefix.key`

0 (default)

1

`call.localConferenceEnabled`

1 (default) - The feature to join a conference during an active call is enabled and the Conference soft key displays.

0 - The feature to join a conference during an active call is disabled and the Conference soft key doesn't display. When you try to join the Conference, an "Unavailable" message displays.

Change causes system to restart or reboot.

`call.offeringTimeOut`

Specify a time in seconds that an incoming call rings before the call is dropped.

60 (default)

0 - No limit.

Note that the call diversion, no answer feature takes precedence over this feature when enabled.

Change causes system to restart or reboot.

`call.playLocalRingBackBeforeEarlyMediaArrival`

Determines whether the phone plays a local ring-back after receiving a first provisional response from the far end.

1 (default) - The phone plays a local ringback after receiving the first provisional response from the far end. If early media is received later, the phone stops the local ringback and plays the early media.

0 - No local ringback plays, and the phone plays only the early media received.

`call.playLocalRingBackBeforeEarlyMediaArrival`

0 (default) - URL mode is used for URL calls.

1 - Number mode is used for URL calls.

`call.ringBackTimeOut`

Specify a time in seconds to allow an outgoing call to remain in the ringback state before dropping the call.

60 (default)

0 - No limit.

Change causes system to restart or reboot.

`call.showDialpadOnProceeding`

0 (default) – The phone doesn't show the dialpad button while a placed call is outgoing.

1 – The phone displays the dialpad button while a placed call is outgoing.

`call.stickyAutoLineSeize`

0 (default) - Dialing through the call list uses the line index for the previous call. Dialing through the contact directory uses a random line index.

1 - The phone uses sticky line seize behavior. This helps with features that need a second call object to work with. The phone attempts to initiate a new outgoing call on the same SIP line that is currently in focus on the LCD. Dialing through the call list when there is no active call uses the line index for the previous call. Dialing through the call list when there is an active call uses the current active call line index. Dialing through the contact directory uses the current active call line index.

Change causes system to restart or reboot.

`call.stickyAutoLineSeize.onHookDialing`

0 (default)

If you set `call.stickyAutoLineSeize` to `1`, this parameter has no effect. The regular `stickyAutoLineSeize` behavior is followed.

If you set `call.stickyAutoLineSeize` to `0` and set this parameter to `1`, this overrides the `stickyAutoLineSeize` behavior for hot dial only. (Any new call scenario seizes the next available line.)

If you set `call.stickyAutoLineSeize` to `0` and set this parameter to `0`, there is no difference between hot dial and new call scenarios.

A hot dial occurs on the line which is currently in the call appearance. Any new call scenario seizes the next available line.

Change causes system to restart or reboot.

`call.switchToLocalRingbackWithoutRTP`

Determines whether local ringback plays in the event that early media stops.

0 (default) – No ringback plays when early media stops.

1 – The local ringback plays if no early media is received.

`call.teluri.showPrompt`

1 (default) - Phone displays a pop-up box to either call or cancel the number when tel URI is executed.

0 - Phone doesn't display the pop-up box.

Per-Registration Dial Plan Parameters

All the following parameters are per-registration parameters that you can configure instead of the general equivalent dial plan parameters.

Per-registration parameters override the general parameters where *x* is the registration number; for example, dialplan.x.applyToTelUriDial overrides dialplan.applyToTelUriDial for registration x.

`dialplan.userDial.timeOut`

Set the time, in seconds, the phone waits for digit input before placing a call when the phone is onhook.

Note: The default value varies depending on the setting for the `device.baseProfile` setting.

`Generic` (default) – 0

`Lync` (default) – 4

0-99 seconds

You can apply `dialplan.userDial.timeOut` only when its value is lower than `up.IdleTimeOut`.

`dialplan.x.applyToCallListDial`

Note: The default value varies depending on the setting for the `device.baseProfile` setting.

`Generic` (default) – 0

`Lync` (default) – 0

0 - The dial plan does not apply to numbers dialed from the received call list or missed call list, including sub-menus for this line.

1 - The dial plan applies to numbers dialed from the received call list or missed call list, including sub-menus for this line.

Change causes system to restart or reboot.

`dialplan.x.applyToDirectoryDial`

0 (default) - The dial plan is not applied to numbers dialed from the directory or speed dial, including auto-call contact numbers for this line.

1 - The dial plan is applied to numbers dialed from the directory or speed dial, including auto-call contact numbers for this line.

Change causes system to restart or reboot.

`dialplan.x.applyToForward`

0 (default) - The dial plan applies to forwarded calls for this line.

1 - The dial plan applies to forwarded calls for this line.

`dialplan.x.applyToTelUriDial`

0

1 (default)

Change causes system to restart or reboot.

`dialplan.x.applyToUserDial`

0

1 (default)

Change causes system to restart or reboot.

`dialplan.x.applyToUserSend`

0

1 (default)

Change causes system to restart or reboot.

`dialplan.x.conflictMatchHandling`

Note: The default value varies depending on the setting for the `device.baseProfile` setting.

Selects the dialplan based on more than one match with the least timeout.

`Generic` (default) - 0

`Lync` (default) - 1

0 - Conflict match handling is disabled.

1 - Conflict match handling is enabled.

`dialplan.x.digitmap.timeOut`

Set the time, in seconds, the phone waits for digit input before placing a call when the phone is offhook.

Note: The default value varies depending on the setting for the `device.baseProfile` setting.

`Generic` (default) – Null

`Lync` (default) – 4

0-100 seconds

Change causes system to restart or reboot.

`dialplan.x.digitmap`

Note: The default value varies depending on the setting for the `device.baseProfile` setting.

`Generic` (default) - Null

`Lync` (default) - 4

string - max number of characters 100

Change causes system to restart or reboot.

`dialplan.x.digitmap.mode`

Specify whether a dial plan uses Poly digit map rules or regular expressions (regex) to manage line switching.

digitmap (default) - The dial plan uses a digit map defined in `dialplan.x.digitmap` to manage line switching.

regex - The dial plan uses regular expression rules defined in `dialplan.x.digitmap` to manage line switching.

Note: Poly Trio systems don't support dial-string manipulation using regex at this time.

`dialplan.x.e911dialmask`

Null (default)

string - max number of characters 256

`dialplan.x.e911dialstring`

Null (default)

string - max number of characters 256

`dialplan.x.impossibleMatchHandling`

0 (default) - Digits are sent to the call server immediately.

1 - A reorder tone is played and the call is canceled.

2 - No digits are sent to the call server until the Send or Dial key is pressed.

3 - No digits are sent to the call server until the timeout is configured by `dialplan.X.impossibleMatchHandling.timeOut` parameter.

Change causes system to restart or reboot.

`dialplan.x.originaldigitmap`

Null (default)

string - max number of characters 2560

`dialplan.x.removeEndOfDial`

0

1 (default)

Change causes system to restart or reboot.

`dialplan.x.routing.emergency.y.server.z`

0 (default)

1

2

3

x, y, and z = 1 to 3

Change causes system to restart or reboot.

`dialplan.x.routing.emergency.y.value`

Null (default)

string - max number of characters 64

Change causes system to restart or reboot.

`dialplan.x.routing.server.y.address`

Null (default)

string - max number of characters 256

Change causes system to restart or reboot.

`dialplan.x.routing.server.y.port`

5060 (default)

1 to 65535

Change causes system to restart or reboot.

`dialplan.x.routing.server.y.transport`

DNSnaptr (default)

TCPpreferred

UDPOnly

TLS

TCPOnly

Change causes system to restart or reboot.

Presence Parameters

Use the following parameters to configure the presence feature.

Note that the parameter pres.reg is the line number used to send SUBSCRIBE. If this parameter is missing, the phone uses the primary line to send SUBSCRIBE.

`pres.idleTimeoutoffHours.enabled`

1 (default) - Enables the off hours idle timeout feature.

0 - Disables the off hours idle timeout feature.

`pres.idleTimeoutoffHours.period`

The number of minutes to wait while the phone is idle during off hours before showing the Away presence status.

15 (default)

1 - 600

`pres.idleTimeout.officeHours.enabled`

1 (default) - Enables the office hours idle timeout feature

0 - Disables the office hours idle timeout feature

`pres.idleTimeout.officeHours.periods`

The number of minutes to wait while the phone is idle during office hours before showing the Away presence status

15 (default)

1 - 600

Provisioning Parameters

Use the following parameters to control the provisioning server system for your phones.

`prov.autoConfigUpload.enabled`

1 (default) - Enables the automatic upload of configuration files from the phone or Web configuration utility to the provisioning server.

0 - Disabled the automatic upload of configuration files from the phone or Web configuration utility to the provisioning server.

`prov.configUploadPath`

Specifies the directory path where the phone uploads the current configuration file.

Null (default)

String

`prov.eula.accepted`

0 (default) - Accept manually the product EULA agreement on the Poly Trio system at the initial start-up.

1 - The EULA agreement is automatically accepted on the Poly Trio system at the initial start-up.

`prov.login.lcCache.domain`

The user's domain name to sign in.

Null (default)

String

`prov.login.lcCache.user`

The user's sign-in name to log in.

Null (default)

String

`prov.login.password.encodingMode`

The default encoding mode for the text in the **Password** field on the **User Login** screen.

123 (default)

Alphanumeric

`prov.login.userId.encodingMode`

The default encoding mode for the text in the **User ID** field on **User Login** screen.

Abc (default)

Alphanumeric

`prov.loginCredPwdFlushed.enabled`

1 (default) - Resets the password field when the user logs in or logs out.

0 - Does not reset the password field when the user logs in or logs out.

`prov.startupCheck.enabled`

1 (default) - The phone is provisioned on startup.

0 - The phone is not provisioned on startup.

`prov.quickSetup.limitServerDetails`

0 (default) - Provide all the necessary details for the given fields.

1 - Enter only the user name and password fields. Other details are taken from `ztp/dhcp` (option66).

Quick Setup Soft Key Parameter

Use the following parameter to configure the **Quick Setup** soft key.

`prov.quickSetup.enabled`

0 (default) - Disables the quick setup feature.

1 - Enables the quick setup feature.

Remote Packet Capture Parameters

Use these parameters to enable and set up the remote packet capture feature.

`diags.dumpcore.enabled`

Determine whether the phone generates a core file if it crashes.

1 (default) - The phone generates a core file.

0 - The phone doesn't generate a core file.

Change causes system to restart or reboot.

`diags.pcap.enabled`

Enable or disable all on-board packet capture features.

0 (default) - Disable on-board packet capture features.

1 - Enable on-board packet capture features.

`diags.pcap.remote.enabled`

Enable or disable the remote packet capture server.

0 (default) - Disable the remote packet capture server.

1 - Enable the remote packet capture server.

`diags.pcap.remote.password`

Enter the remote packet capture password.

<MAC Address>(default)

alphanumeric value

`diags.pcap.remote.port`

Specify the TLS profile to use for each application.

2002 (default)

Valid TCP Port

SDP Parameters

Use the following parameters to configure the Session Description Protocol (SPD).

`voIpProt.SDP.answer.useLocalPreferences`

0 (default) - Attempt to match the negotiated voice and video codecs using the order in the SDP offer from the far end.

1- Answer SDP offers using the phone's local preferences for codec ordering instead of the preference order from the offer.

`voIpProt.SDP.answer.useLocalPreferences.video`

Allows you to reset the parameter `voIpProt.SDP.answer.useLocalPreferences` to the default 0 for audio only.

1(default) - The phone uses its own preference list instead of the preference list in the offer when deciding which video codec to use.

0 - The phone’s use of its own preference list is disabled.

`voIpProt.SDP.early.answerOrOffer`

0 (default) - SDP offer or answer is not generated.

1 - SDP offer or answer is generated in a provisional reliable response and PRACK request and response.

Note: An SDP offer or answer is not generated if `reg.x.musicOnHold.uri` is set.

`voIpProt.SDP.offer.iLBC.13_33kbps.includeMode`

1(default) - The phone should include the mode=30 FMTP parameter in SDP offers:

- If you set `voice.codecPref.iLBC.13_33kbps`, and `voice.codecPref.iLBC.15_2kbps` is `Null`.
- If you set both `voice.codecPref.iLBC.13_33kbps` and `voice.codecPref.iLBC.15_2kbps`, the iLBC 13.33 Kbps codec is set to a higher preference.

0 - the phone should not include the mode=30 FTMP parameter in SDP offers even if iLBC 13.33 Kbps codec is being advertised.

`voIpProt.SDP.offer.rtcpVideoCodecControl`

This parameter determines whether or not RTCP-FB-based controls are offered in Session Description Protocol (SDP) when the phone negotiates video I-frame request methods. Even when RTCP-FB-based controls aren't offered in SDP, the phone may still send and receive RTCP-FB I-frame requests during calls depending on other parameter settings. For more information about video I-frame request behavior, see `video.forceRtcpVideoCodecControl`. For an account of all parameter dependencies refer to "I-Frames."

section.

0 - The phone doesn't include the SDP attribute "a=rtcp-fb".

1 (default) - The phone includes the SDP attribute "a=rtcp-fb" into offers during outbound SIP calls.

Session Header Parameters

You can enable session header parameters to convey information between phones about the SIP message.

Use the following parameters to configure session header components.

`voIpProt.SIP.supportFor100rel`

Poly recommends setting this parameter to `1` when using Poly Trio systems with Polycom RealPresence DMA systems.

1 (default) - The phone advertises support for reliable provisional responses in its offers and responses.

0 - The phone doesn't offer 100rel and rejects offers requiring 100rel.

`voIpProt.SIP.keepalive.sessionTimers`

0 (default) – The phone doesn't declare a timer in the Support header in an INVITE. The call doesn't disconnect when the phone doesn't receive UPDATE packet. The phone still responds to a re-INVITE or UPDATE and follows the session timer to send re-INVITE or UPDATE if the remote endpoint asks for it.

1 – The session timer is enabled and the call disconnects when the phone doesn't receive an UPDATE packet within the specified session timer.

`reg.x.keepalive.sessionTimers`

1 (default) – The session timer is enabled and the call received on the registered line disconnects when the phone doesn't receive an UPDATE packet within the specified timer.

0 – The session timer is disabled and the call received on the registered line doesn't disconnect when the phone doesn't receive an UPDATE packet.

Upgrade Parameters

Specify the URL of a custom download server and the PVOS download server when you want the phone to check when to search for software upgrades.

`upgrade.custom.server.url`

The URL of a custom download server.

URL (default) - NULL

`upgrade.plcm.server.url`

The URL of the PVOS software download.

URL - http://downloads.polycom.com/voice/software/

User Preferences Parameters

Use the following parameters to set phone user preferences.

`up.backlight.idleIntensity`

Brightness of the LCD backlight when the phone is idle. Range is 0 to 3.

1 (Default) - Low

0

2 - Medium

3 - High

If this setting is higher than active backlight brightness (`onIntensity`), the active backlight brightness is used.

`up.backlight.onIntensity`

Brightness of the LCD backlight when the phone is active (in use). Range is 0 to 3.

3 (Default) - High

1 - Low

2 - Medium

`up.backlight.timeout`

Number of seconds to wait before the backlight dims from the active intensity to the idle intensity. Range is 5 to 60.

40 (default)

`up.basicSettings.networkConfigEnabled`

Specifies whether **Network Configuration** is shown or not shown under the **Basic Settings** menu.

0 (default) - **Network Configuration** is not shown under **Basic Settings**.

1 - **Basic Settings** menu shows **Network Configuration** with configurable network options for the user without administrator rights.

`up.DIDFormat`

NumberAndExtension (default) – Display the DID number and extension.

NumberOnly – Display the DID number on the phone screen.

`up.cfgWarningsEnabled`

Specifies whether a warning displays on a phone or not.

0 (Default) - Warning does not display.

1 - Warning is displayed on the phone if it is configured with pre-UC Software 3.3.0 parameters.

`up.formatPhoneNumbers`

Enable or disable automatic number formatting.

1 (Default)

0

`up.hearingAidCompatibility.enabled`

Specifies whether audio Rx equalization is enabled or disabled.

0 (Default) - Audio Rx equalization is enabled.

1 - Phone audio Rx (receive) equalization is disabled for hearing aid compatibility.

`up.idleRestingState`

menu (default) – The idle screen displays the **Home** screen menu.

calendar – The idle screen displays a top-level calendar.

dialpad – The idle screen displays a dial pad

`up.idleStateView`

Sets the phone default view.

0 (Default) - Call/line view is the default view.

1 - Home screen is the default view.

Change causes system to restart or reboot.

`up.idleTimeout`

Set the number of seconds that the phone is idle for before automatically leaving a menu and showing the idle display.

During a call, the phone returns to the Call screen after the idle timeout.

40 seconds (default)

0 to 65535 seconds

Change causes system to restart or reboot.

`up.lineKeyCallTerminate`

Specifies whether or not you can press the line key to end an active call.

0 (Default) - User cannot end an active call by pressing the line key.

1 - User can press a line key to end an active call.

`up.numberFirstCID`

Specifies what is displayed first on the **Caller ID** display.

0 (Default) - **Caller ID** display shows the caller's name first.

1 - Caller's phone number is shown first.

Change causes system to restart or reboot.

`up.numOfDisplayColumns`

Sets the maximum number of columns on the display. Set the maximum number of columns that phones display. Range is 0 to 4.

3 (Default)

0 - Phones display one column.

Change causes system to restart or reboot.

`up.osdIncomingCall.Enabled`

Specifies whether or not to display full screen popup or OSD for incoming calls.

1 (Default) - Full screen popup or OSD for incoming calls displays.

0 - Full screen popup or OSD for incoming calls does not display.

`up.rebootSoundEnabled`

1 (default) – Enable a sound effect alert when the phone reboots.

0 – Disable a sound effect alert when the phone reboots.

`up.ringer.minimumVolume`

Configure the minimum ringer volume. This parameter defines how many volume steps are accessible below the maximum level by the user.

16 (Default) - Full 16 steps of volume range are accessible.

0 - Ring volume is not adjustable by the user and the phone uses maximum ring volume.

Example: Upon bootup, the volume is set to ½ the number of configured steps below the maximum (16). If the parameter is set to 8 on bootup, the ringer volume is set to 4 steps below maximum.

`up.screenSaver.enabled`

0 (Default) - Screen saver feature is disabled.

1 - Screen saver feature is enabled. If a USB flash drive containing images is connected to the phone a slide show cycles through the images from the USB flash drive when the screen saver feature is enabled.

The images must be stored in the directory on the flash drive specified by `up.pictureFrame.folder`. The screen saver displays when the phone has been in the idle state for the amount of time specified by `up.screenSaver.waitTime`.

`up.screenSaver.waitTime`

Number of minutes that the phone waits in the idle state before the screen saver starts. Range is 1 to 9999 minutes.

15 (Default)

`up.simplifiedSipCallInfo`

1 (Default) - This displayed host name is trimmed for both incoming and outgoing calls and the protocol tag/information is not displayed for incoming and outgoing calls.

0 - The full host name displays and the protocol tag/information displays for incoming and outgoing calls.

`up.softkey.transferTypeOption.enabled`

1 - The user can change the transfer type from consultative to blind and vice versa using a soft key after the user has initiated a transfer, but before completing the call to the far end.

0 (default) - There is no option to change from consultative to blind and blind to consultative when the user is in dial prompt after pressing the **Transfer** soft key.

`up.status.message.flash.rate`

Controls the scroll rate of the status bar. Range is 2 to 8 seconds.

2 seconds (Default)

`up.showDID`

AllScreens (default) – Display the DID number on all the screens.

None – Disable DID number on phone.

LockedScreen – Display the DID number on the lock screen.

StatusScreen – Display the DID number on the Status screen/Idle screen.

IncomingOSD – Display the DID number on the incoming On Screen Display (OSD) screen.

LockedScreenIncomingOSD – Display the DID number on the lock and incoming OSD screen.

LockedAndStatusScreen – Display the DID number on the lock and Status/Idle screen.

StatusScreenIncomingOSD – Display the DID number on the incoming OSD and Status/Idle screen.

`up.volumeChangeTone.enabled`

1 (default) – The phone plays a tone when the user adjusts the ringer or call volume.

0 – The phone does not play a tone.

Zoom Rooms Base Profile: 0 (default)

`up.warningLevel`

Line keys block display of the background image. All warnings are listed in the **Warnings** menu.

0 (Default) - The phone's warning icon and a pop-up message display on the phone for all warnings.

1 - Warning icon and pop-up messages are only shown for critical warnings.

2 - Phone displays a warning icon and no warning messages. For all the values, all warnings are listed in the **Warnings** menu.

Access to the **Warnings** menu varies by phone model.

Change causes system to restart or reboot.

`up.welcomeSoundEnabled`

1 (Default) - Welcome sound is enabled and played each time the phone reboots.

0 - Welcome sound is disabled.

To use a welcome sound you must enable the parameter `up.welcomeSoundEnabled` and specify a file in `saf.x`. The default UC Software welcome sound file is Welcome.wav.

Change causes system to restart or reboot.

`up.welcomeSoundOnWarmBootEnabled`

0 (Default) - Welcome sound is played when the phone powers on (cold boot), but not after it restarts or reboots (warm boot).

1 - Welcome sound plays each time the phone powers on, reboots, or restarts.

Change causes system to restart or reboot.

`up.display.showFullCallerID`

Phone displays the caller ID.

0 (default) – Phone displays the caller ID on the first line.

1 – Phone displays the caller ID on the second line.

`up.answerCall.listOrder`

Defines the order to answer a call upon pressing speaker button on the phone.

LIFO (default) - Last-In, First-Out.

FIFO - First-In, First-Out.

Voice Parameters

Use the following parameters to configure phone audio.

`voice.rxPacketFilter`

Define a high-pass filter to improve sound intelligibility when the phone receives narrow band signals. Narrow band signals occur when a narrow band codec is in use, such as G.711mu, G.711A, G.729AB, iLBC, and some Opus and SILK variants.

0 (default) - Pass through.

1 - 300 Hz high-pass.

2 - 300 Hz high-pass with pre-emphasis. Use this value with G.729.

`voice.txPacketDelay`

Null (default)

normal, Null - Audio parameters are not changed.

low - If there are no precedence conflicts, the following changes are made:

```
voice.codecPref.G722="1"
voice.codecPref.G711Mu="2"
voice.codecPref.G711A="3"
voice.codecPref.<OtherCodecs>=""
voice.audioProfile.G722.payloadSize="10"
voice.audioProfile.G711Mu.payloadSize= "10"
voice.audioProfile.G711A.payloadSize= "10"
voice.aec.hs.enable="0"
voice.ns.hs.enable="0"
```

Change causes system to restart or reboot.

`voice.txPacketFilter`

Null (default)

0 - Tx filtering is not performed.

1 - Enables Narrowband Tx high pass filter.

Change causes system to restart or reboot.

Acoustic Echo Suppression (AES) Parameter

Use the following parameter to enable speakerphone acoustic echo suppression (AES).

This feature removes residual echo after AEC processing. Because AES depends on AEC, enable AES only when you also enable AEC using voice.aec.hd.enable.

`voice.aes.hs.enable`

1 (default) - Enables the handset AES function.

0 - Disables the handset AES function.

Comfort Noise Parameters

Use the following parameters to configure the addition and volume of comfort noise during conferences.

`voice.cn.hf.enable`

0 (default) - Comfort noise not added.

1 - Adds comfort noise added into the Tx path for hands-free operation.

Far end users should use this feature when they find the phone to be 'dead', as the near end user stops talking.

`voice.cn.hf.attn`

35 (default) - quite loud

0 - 90

Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hf.enabled` is `1`.

`voice.cn.hd.attn`

30 (default) - quite loud

0 - 90

Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hd.enabled` is `1`.

`voice.cn.hs.enable`

0 (default) - Comfort noise is not added into the Tx path for the handset.

1 - Adds comfort noise is added into the Tx path for the headset.

Far end users should use this feature when they find the phone to be 'dead', as the near end user stops talking.

`voice.cn.hs.attn`

35 (default) - quite loud

0 - 90

Attenuation of the inserted comfort noise from full scale in decibels; smaller values insert louder noise. Use this parameter only when `voice.cn.hs.enabled` is `1`.

`voice.vadRxGain`

Tunes VAD or CNG interoperability in a multi-vendor environment.

0 (default)

-20 to +20 dB

The specified gain value in dB is added to the noise level of an incoming VAD or CNG packet, when in a narrow band call.

When tuning in multi-vendor environments, the existing Poly to Poly phone behavior can be retained by setting `voice.vadTxGain = -voice.vadRxGain`.

This parameter is ignored for HD calls.

`voice.vadTxGain`

Tunes VAD or CNG interoperability in a multi-vendor environment.

0 (default)

-20 to +20 dB

The specified gain value in dB is added to the noise level of an incoming VAD or CNG packet, when in a narrow band call.

This causes the noise level to synthesize at the local phone to change by the specified amount.

When tuning in multi-vendor environments, the existing Poly to Poly phone behavior can be retained by setting `voice.vadTxGain = -voice.vadRxGain`.

This parameter is ignored for HD calls.

Voice Jitter Buffer Parameters

Use the following parameters to configure wired network interface voice traffic and push-to-talk interface voice traffic.

`voice.rxQoS.avgJitter`

The average jitter in milliseconds for wired network interface voice traffic.

20 (default)

0 to 80

`avgJitter`: The wired interface minimum depth will be automatically configured to adaptively handle this level of continuous jitter without packet loss.

Change causes system to restart or reboot.

`voice.rxQoS.maxJitter`

The average jitter in milliseconds for wired network interface voice traffic.

240 (default)

0 to 320

`maxJitter`: The wired interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss.

Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets are lost. Actual jitter above the maximum value always results in packet loss. If legacy `voice.audioProfile.x.jitterBuffer.*` parameters are explicitly specified, they are used to configure the jitter buffer and these `voice.rxQoS` parameters are ignored.

Change causes system to restart or reboot.

`voice.rxQoS.ptt.avgJitter`

The average jitter in milliseconds for IP multicast voice traffic.

150 (default)

0 - 200

`avgJitter`: The PTT/Paging interface minimum depth is automatically configured to adaptively handle this level of continuous jitter without packet loss.

Change causes system to restart or reboot.

`voice.rxQoS.ptt.maxJitter`

The maximum jitter in milliseconds for IP multicast voice traffic.

480 (default)

20 - 500

`maxJitter`: The PTT/Paging interface jitter buffer maximum depth will be automatically configured to handle this level of intermittent jitter without packet loss.

Actual jitter above the average but below the maximum may result in delayed audio play out while the jitter buffer adapts, but no packets will be lost. Actual jitter above the maximum value will always result in packet loss.

If legacy `voice.audioProfile.x.jitterBuffer.*` parameters are explicitly specified, they will be used to configure the jitter buffer and these `voice.rxQoS` parameters are ignored.

Change causes system to restart or reboot.

`voice.handsfreePtt.rxdg.offset`

This parameter allows a digital Rx boost for Push To Talk.

0 (default)

9 to -12 - Offsets the RxDg range of the hands-free and hands-free Push-to-Talk (PTT) by the specified number of decibels.

`voice.ringerPage.rxdg.offset`

This parameter allows a digital Rx boost for Push To Talk. Use this parameter for handsfree paging in high noise environments.

0 (default)

9 to -12 - Raise or lower the volume of the ringer and hands-free page by the specified number of decibels.

Digital Gain Parameters

Use the following parameters configure the gain applied to microphones.

`voice.handset.txdg`

Digital gain applied to the wired handset mic.

0 (Default)

-90 to 90

`voice.handsfree.txdg`

Digital gain applied to the built-in hands free mic.

0 (Default)

-90 to 90

`voice.headset.txdg`

Digital gain applied to the wired headset mic.

0 (Default)

-90 to 90

`voice.usb.headset.txdg`

Digital gain applied to the USB headset mic.

0 (Default)

-90 to 90

`voice.bt.headset.txdg`

Digital gain applied to the Bluetooth headset mic.

0 (Default)

-90 to 90

XML Streaming Protocol Parameters

Use the following parameters to set the XML streaming protocols for instant messaging, presence, and contact lists for BroadSoft features.

`xmpp.1.auth.domain`

Specify the domain name of the XMPP server.

Null (Default)

Other values - UTF-8 encoded string

`xmpp.1.auth.useLoginCredentials`

Specifies whether or not to use the login credentials provided in the phone's **Login Credentials** menu for XMPP authentication.

0 (Default)

1

`xmpp.1.enable`

Specifies to enable or disable XMPP presence.

0 (Default)

1

Content Sharing Parameters

This section provides information on the content sharing parameters for Poly Trio.

AirPlay Debugging Log Parameters

If you experience further issues using AirPlay-certified devices with the Poly Trio 8800 system, you can enable the following logging parameters on your Poly Trio to get extended debugging data.

`airp`

Session management and communication specifically for AirPlay certified devices.

`airpl`

Protocol library for AirPlay-certified devices

`airps`

Android service AirPlay-certified devices

`lc`

Local Content (including for AirPlay-certified devices and PPCIP) session management

Bluetooth Discovery Parameters

Use the following parameter to enable Bluetooth Discovery between Poly Trio and the Polycom Content App.

`bluetooth.beacon.ipAddress.enabled`

Set to send the IP address of the system over Bluetooth.

1 (default) - Enables sending the system IP address over Bluetooth. Turns Bluetooth radio on when `feature.bluetooth.enabled = 1`.

0 - Disables sending the system IP address over Bluetooth

Note: Enable the parameter `feature.bluetooth.enabled` to use this feature.

Display Content While Idle Parameters

Set up the Poly Trio system to display content when idle using the following parameters.

`mr.localContent.autoDisplay.idle`

When the phone is not in a call, content shared via HDMI is displayed on a connected monitor.

0 (Default) – Content does not display when the phone is idle.

1 – Content displays when the phone is idle.

`mr.localContent.autoDisplay.idle.autoResumeTimeout`

Sets the amount of idle time, in minutes, allowed after content sharing is stopped. Once this time limit is reached, the content sharing resumes.

60 (default)

0 - Infinite

Content Sharing Parameters

Use the following parameters to configure content sharing options.

To enable device pairing with the Poly Trio solution, use the `smartpairing*` parameters.

`content.autoAccept.rdp`

1 (default) - Content sent via Skype for Business Remote Desktop Protocol (RDP) by far-end users is automatically accepted and displayed on a near-end Poly Trio solution.

0 - Near-end users are prompted to accept Skype for Business RDP content sent to Poly Trio solution from a far-end user.

`content.bfcp.enabled`

1 (default) - Enable content sharing by offering or accepting the Binary Floor Control Protocol (BFCP) in Session Description Protocol (SDP) negotiation during SIP calls. Does not apply to Skype for Business calls.

0 - Disable content sharing using BFCP.

`content.bfcp.port`

15000 (default)

0 - 65535

`content.bfcp.transport`

UDP (default)

TCP

`content.local.bitRate`

Set the default maximum bit rate for local content, content not shared during an active call.

8000000 (8Mbps) (default)

Range is 1000000 (1Mbps) to 20000000 (20Mbps).

This parameter does not apply to content shared using Screen Mirroring with Airplay or the Wireless Display feature.

`content.vbssPush.bitRateMinimum`

Sets the minimum allowable VbSS (video-based screen sharing) bit rate. VbSS clients that request a bit rate lower than the set minimum don't receive a content stream.

200000 (200kbps) (default)

100000 (100kbps) - 2000000 (2Mbps)

`mr.content.rdp.tls.enabled`

Enable or disable TLS encryption of Skype for Business RDP content between hubs and devices.

1 (default)

0

`mr.content.rdp.tls.enabled`

1 (default) - Enable TLS encryption of Skype for Business RDP content between hubs and devices.

Disable TLS encryption of Skype for Business RDP content.

`mr.contentStreamPortEnd`

The IP port range end port used for content input streams received from the Poly Trio Visual+ system.

4320 (default)

1024 – 65436

`mr.contentStreamPortStart`

The IP port range start port used for content input streams received from the Poly Trio Visual+ system.

4300 (default)

1024 – 65436

`mr.localContent.autoDisplay.enable`

1 (Default) – Content automatically displays when an HDMI cable is plugged in.

0 – Content does not automatically start if HDMI is plugged in. Use the Poly Trio system user interface to begin sharing content.

`reg.x.content.bfcp.enabled`

1 (default) - Enable Binary Floor Control Protocol content to be shared during calls for the line you specify.

0 - Disable Binary Floor Control Protocol content.

`smartPairing.mode`

Enables users with People+Content IP or Desktop on a computer or Mobile on a tablet to pair with the Poly Trio conference phone using SmartPairing.

Disabled (default) - Users cannot use SmartPairing to pair with the conference phone.

Manual - Users must enter the IP address of the conference phone to pair with it.

`smartPairing.volume`

The relative volume to use for the SmartPairing ultrasonic beacon.

6 (default)

0 - 10

HDMI and VGA Content Sharing Parameters

Use the following parameters to configure HDMI and VGA content sharing with the paired Poly Trio VisualPro or RealPresence Group Series system.

`mr.contentStreamPortStart`

Sets where the IP port range begins for content input streams from a network device.

4300 (default)

1024 - 65436

Change causes system to restart or reboot.

`mr.contentStreamPortEnd`

Sets where the IP port range ends for content input streams from a network device.

4320 (default)

1024 - 65436

Change causes system to restart or reboot.

Polycom People+Content IP Parameters

The following table lists parameters that configure content sharing with the Poly Trio solution.

`content.ppcipServer.authType`

The authentication type used for PPCIP content sessions.

none (default) - No security code for Polycom People+Content IP-enabled clients is required.

passcode - Use a security code to authenticate Polycom People+Content IP-enabled clients.

`content.ppcipServer.enabled`

1 (default) - Enable Polycom People+Content IP content server for sharing.

0 - Disable Polycom People+Content IP content server.

`content.ppcipServer.enabled.Trio8500`

1 (default) - Enable Polycom People+Content IP content server for content sharing with Poly Trio 8500.

0 - Disable Polycom People+Content IP content server for Poly Trio 8500.

`content.ppcipServer.enabled.Trio8800`

1 (default) - Enable Polycom People+Content IP content server for content sharing with Poly Trio 8800.

0 - Disable Polycom People+Content IP content server for Poly Trio 8800.

Polycom People+Content IP over USB Parameter

The following parameter configures the People+Content over USB feature.

`feature.usb.device.content`

1 (default) - Enable content sharing using the People+Content IP application on a computer connected by USB to Poly Trio solution.

0 - Disable content sharing using the People+Content IP application on a computer connected by USB to Poly Trio solution.

Poly Trio 8800 for AirPlay Parameters

Use the following parameters to configure the Poly Trio 8800 system for AirPlay-certified devices.

`bluetooth.beacon.ipAddress.enabled`

Set to send the IP address of the system over Bluetooth.

1 (default) - Enables sending the system IP address over Bluetooth. Turns Bluetooth radio on when `feature.bluetooth.enabled = 1`.

0 - Disables sending the system IP address over Bluetooth

Note: Enable the parameter `feature.bluetooth.enabled` to use this feature.

`content.airplayServer.authType`

none (default) - No security code for AirPlay certified devices is required.

passcode - Use a security code to authenticate AirPlay-certified devices.

`content.airplayServer.enabled`

0 (default) - Disable the content sink for AirPlay-certified devices.

1 - Enable the content sink for AirPlay-certified devices.

`content.airplayServer.maxResolution`

Set the content resolution.

720p (default)

1080p

1024x1024

960x960

480x480

`content.airplayServer.name`

Specify a system name for the local content sink for AirPlay certified devices. If left blank the previously configured or default system name is used.

NULL (default)

`content.local.authChangeInterval`

Set the interval in minutes between changes to the local content authentication credentials.

1440 (default)

0 - 65535

0 - Do not change

`content.local.authChangeMode`

Specify when the security code for content sharing with AirPlay-certified devices changes.

session (default) - Code changes at the end of each content sharing session.

relativeTime - Code changes at an interval specified by the `content.local.authChangeInterval` parameter.

Poly Trio 8800 for Miracast-Certified Devices Parameters

Use the following parameters to configure Wireless Display on the Poly Trio 8800 system.

`content.wirelessDisplay.sink.authorizationType`

Auto (Default) - Content is automatically accepted and displays on the Poly Trio 8800 system.

Button - Users must confirm content acceptance on a popup message.

`content.wirelessDisplay.sink.bitrate`

Set the content maximum bitrate in Mbps

30 (default)

0 - 60

0 allows auto-negotiation.

`content.wirelessDisplay.sink.enabled`

0 (default) - Disable Wireless Display.

1 - Enable Wireless Display.

`content.wirelessDisplay.sink.fps`

Set the content frame rate in frames per second.

30 (default)

0 - 60

0 allows auto-negotiation

`content.wirelessDisplay.sink.height`

Set the maximum content height in pixels.

1080 (default)

0 - 1200

0 allows auto-negotiation

`content.wirelessDisplay.sink.name`

NULL - default

Specify a system name for the local content sink for Android or Windows devices. If left blank the previously configured or default system name is used.

`content.wirelessDisplay.sink.width`

Set the maximum content width in pixels.

1920 (Default and Maximum)

0 allows auto-negotiation

Wireless Display Debugging Log Parameters

If you experience further issues using Wireless Display on the Poly Trio 8800 system, you can enable the following logging parameters on your Poly Trio 8800 system to get extended debugging data.

`wdisp`

Wireless Display session management and communication with the Wireless Display source

`apps`

Wireless Display support for Android

`lc`

Local Content (including Wireless Display and PPCIP) session management

Device Parameters

The < `device/` > parameters—also known as device settings—contain default values that you can use to configure basic settings for multiple phones within your network.

Poly provides a global `device.set` parameter that you must enable to install software and change device parameters. In addition, each `<device/>` parameter has a corresponding `.set` parameter that enables or disables the value for that device parameter. You need to enable the corresponding `.set` parameter for each parameter you want to apply.

After you complete the software installation or configuration changes to device parameters, remove `device.set` to prevent the phones from rebooting and triggering a reset of device parameters that phone users might have changed after the initial installation.

If you configure any parameter values using the `<device/>` parameters, any subsequent configuration changes you make from the system web interface or phone local interface do not take effect after a phone reboot or restart.

The `<device/>` parameters are designed to be stored in flash memory and for this reason, the phone does not upload `<device/>` parameters to the `<MAC>-web.cfg` or `<MAC>-phone.cfg` override files if you make configuration changes through the system web interface or phone interface. This design protects your ability to manage and access the phones using the standard set of parameters on a provisioning server after the initial software installation.

Changing Device Parameters

Keep the following in mind when modifying device parameters:

- Note that some parameters may be ignored. For example, if DHCP is enabled, it will still override the value set with `device.net.ipAddress`.
- Though individual parameters are checked to see whether they are in range, the interaction between parameters is not checked. If a parameter is out of range, an error message displays in the log file and the parameter is not be used.
- Incorrect configuration can put the phones into a reboot loop. For example, server A has a configuration file that specifies that server B should be used, and server B has a configuration file that specifies that server A should be used.

To detect errors, including IP address conflicts, Polycom recommends that you test the new configuration files on two phones before initializing all phones.

Device Parameters

Use the following <device/> parameters to configure some device settings.

Note: The default values for the <device/> parameters are set at the factory when the phones are shipped. For a list of the default values, see the latest Product Shipping Configuration Change Notice at [Poly Engineering Advisories and Technical Notifications](#).

`device.auth.localAdminPassword`

Set the phone's local administrative password. The minimum length is defined by `sec.pwd.length.admin`.

String (32 character max)

`device.auth.localUserPassword`

Set the phone user's local password. The minimum length is defined by `sec.pwd.length.user`.

String (32 character max)

`device.auxPort.enable`

Enable or disable the phone auxiliary port.

0 - Disable the phone auxiliary port.

1 (default) - Enable the phone auxiliary port.

Change causes system to restart or reboot.

`device.baseProfile`

NULL (default)

Generic - Sets the base profile to Generic for OpenSIP environments.

Lync - Sets the base profile for Skype for Business deployments.

USBOptimized - Sets the base profile for connecting the Poly Trio solution to a Microsoft Room System or a Microsoft Surface Hub.

MSTeams - Sets the base profile for Microsoft Teams deployments.

ZoomRooms - Sets the base profile for Zoom Rooms deployments.

Change causes system to restart or reboot.

`device.dhcp.bootSrvOpt`

When the boot server is set to Custom or Custom+Option66, specify the numeric DHCP option that the phone looks for.

Null

128 to 254

Change causes system to restart or reboot.

`device.dhcp.bootSrvOptType`

Set the type of DHCP option the phone looks for to find its provisioning server if `device.dhcp.bootSrvUseOpt="Custom"`.

IP address - The IP address provided must specify the format of the provisioning server.

String - The string provided must match one of the formats specified by `device.prov.serverName`.

Change causes system to restart or reboot.

`device.dhcp.bootSrvUseOpt`

Default - The phone looks for option number 66 (string type) in the response received from the DHCP server. The DHCP server sends address information in option 66 that matches one of the formats described for `device.prov.serverName`.

Custom - The phone looks for the option number specified by `device.dhcp.bootSrvOpt` and the type specified by `device.dhcp.bootSrvOptType` in the response received from the DHCP server.

Static - The phone uses the boot server configured through the provisioning server `device.prov.*` parameters.

Custom and Default - The phone uses the custom option first or use option 66 if the custom option is not present.

Change causes system to restart or reboot.

`device.dhcp.dhcpVlanDiscOpt`

Set the DHCP private option to use when `device.dhcp.dhcpVlanDiscUseOpt="Custom"`.

128 to 254

Change causes system to restart or reboot.

`device.dhcp.dhcpVlanDiscUseOpt`

Set how VLAN Discovery occurs.

Disabled - No VLAN discovery through DHCP.

Fixed - Use predefined DHCP vendor-specific option values of 128, 144, 157 and 191 (`device.dhcp.dhcpVlanDiscOpt` is ignored).

Custom - Use the number specified by `device.dhcp.dhcpVlanDiscOpt`.

Change causes system to restart or reboot.

`device.dhcp.enabled`

Enable or disable DHCP.

0 - DHCP is disabled.

1 (default) - DHCP is enabled.

Change causes system to restart or reboot.

`device.dhcp.option60Type`

Set the DHCP option 60 type.

Binary - Vendor-identifying information is in the format defined in RFC 3925.

ASCII - Vendor-identifying information is in ASCII format.

Change causes system to restart or reboot.

`device.dns.altSrvAddress`

Sets the secondary server where the phone directs DNS queries.

Server Address

Change causes system to restart or reboot.

`device.dns.domain`

Set the phone’s DNS domain.

String

Change causes system to restart or reboot.

`device.dns.serverAddress`

Sets the primary server where the phone directs DNS queries.

Server Address

Change causes system to restart or reboot.

`device.hostname`

Specify a hostname for the phone when using DHCP by adding a hostname string to the phone's configuration.

If `device.host.hostname.set="1"` and `device.host.hostname="Null"`, the DHCP client uses option 12 to send a predefined host name to the DHCP registration server using `Polycom_`*<MACaddress>*.

String —The maximum length of the host name string is ≤ 255 bytes, and the valid character set is defined in RFC 1035.

Change causes system to restart or reboot.

device.net.cdpEnabled

Determine if the phone attempts to determine its VLAN ID and negotiate power through CDP.

0 - Disabled

1 - Enabled

Change causes system to restart or reboot.

device.net.dot1x.anonid

EAP-TTLS and EAP-FAST only. Set the anonymous identity (user name) for 802.1X authentication.

String

Change causes system to restart or reboot.

device.net.dot1x.enabled

Enable or disable 802.1X authentication.

0 - Disabled

1 - Enabled

Change causes system to restart or reboot.

device.net.dot1x.identity

Set the identity (user name) for 802.1X authentication.

String

Change causes system to restart or reboot.

device.net.dot1x.method

Specify the 802.1X authentication method, where EAP-NONE means no authentication.

EAP-None

EAP-TLS

EAP-PEAPv0-MSCHAPv2

EAP-PEAPv0-GTC

EAP-TTLS-MSCHAPv2

EAP-TTLS-GTC

EAP-FAST

EAP-MD5

device.net.dot1x.password

Set the password for 802.1X authentication. This parameter is required for all methods except EAP-TLS.

String

Change causes system to restart or reboot.

`device.net.etherModeLAN`

Set the LAN port mode that sets the network speed over Ethernet.

Poly recommends that you don't change this setting.

0 - Auto (default)

1 - 10HD

2 - 10FD

3 - 100HD

4 - 100FD

5 - 1000FD

HD means half-duplex and FD means full duplex.

Change causes system to restart or reboot.

`device.net.etherModePC`

Set the PC port mode that sets the network speed over Ethernet.

-1 - Disables the PC port

0 - Auto (default)

1 - 10HD

2 - 10FD

3 - 100HD

4 - 100FD

5 - 1000FD

HD means half-duplex and FD means full duplex.

Change causes system to restart or reboot.

`device.net.etherStormFilter`

1 - DoS storm prevention is enabled and received Ethernet packets are filtered to prevent TCP/IP stack overflow caused by bad data or too much data.

0 - DoS storm prevention is disabled.

Change causes system to restart or reboot.

`device.net.etherStormFilterPpsValue`

Set the corresponding packets per second (pps) for storm filter and to control the incoming network traffic.

17 to 40

38 (default)

`device.net.etherStormFilterPpsValue.set`

0 (default) - You can't configure the `device.net.etherStormFilterPpsValue` parameter.

1 - You can configure the `device.net.etherStormFilterPpsValue` parameter.

`device.net.ipAddress`

Set the phone's IP address.

This parameter is disabled when `device.dhcp.enabled="1"`.

String

Change causes system to restart or reboot.

`device.net.IPgateway`

Set the phone's default router.

IP address

Change causes system to restart or reboot.

`device.net.lldpEnabled`

0 - The phone doesn't attempt to determine its VLAN ID.

1 - The phone attempts to determine its VLAN ID and negotiate power through LLDP.

Change causes system to restart or reboot.

`device.net.lldp.extendedDiscover`

0 to 3600 - Duration (in seconds) of LLDP extended discovery duration applied in both the application and updater

0 (default)

Change causes system to restart or reboot.

This parameter overrides `net.lldp.extendedDiscovery`.

`device.net.lldpFastStartCount`

Specify the number of consecutive LLDP packets the phone sends at the time of LLDP discovery, which are sent every one second.

5 (default)

3 to 10

`device.net.subnetMask`

Set the phone's subnet mask.

This parameter is disabled when `device.dhcp.enabled="1"`.

Subnet mask

Change causes system to restart or reboot.

`device.net.vlanId`

Set the phone's 802.1Q VLAN identifier.

Null - No VLAN tagging.

0 to 4094

Change causes system to restart or reboot.

`device.prov.maxRedunServers`

Set the maximum number of IP addresses to use from the DNS.

1 to 8

Change causes system to restart or reboot.

`device.prov.password`

Set the password for the phone to log in to the provisioning server, which may not be required.

If you modify this parameter, the phone reprovisions. The phone may also reboot if the configuration on the provisioning server has changed.

String

Change causes system to restart or reboot.

`device.prov.redunAttemptLimit`

Set the maximum number of attempts to attempt a file transfer before the transfer fails. When multiple IP addresses are provided by DNS, one attempt is considered to be a request sent to each server.

1 to 10

Change causes system to restart or reboot.

`device.prov.redunInterAttemptDelay`

Set the number of seconds to wait after a file transfer fails before retrying the transfer. When multiple IP addresses are returned by DNS, this delay only occurs after each IP has been tried.

0 to 300

Change causes system to restart or reboot.

`device.prov.serverName`

IP address

Domain name string

URL

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

`device.prov.serverType`

Set the protocol the phone uses to connect to the provisioning server. Active FTP is not supported for BootROM version 3.0 or later, and only implicit FTPS is supported.

FTP (default)

TFTP

HTTP

HTTPS

FTPS

Change causes system to restart or reboot.

`device.prov.tagSerialNo`

0 - The phone's serial number (MAC address) isn't included in the User-Agent header of HTTPS/HTTPS transfers and communications to the microbrowser and web browser.

1 - The phone's serial number is included.

`device.prov.upgradeServer`

Specify the URL or path for a software version to download to the device.

On the system web interface, the path to the software version you specify displays in the drop-down menu on the **Software Upgrade** page.

NULL (default)

String

0 to 255 characters

`device.prov.user`

The username required for the phone to log in to the provisioning server (if required).

If you modify this parameter, the phone reprovisions, and it may reboot if the configuration on the provisioning server has changed.

String

`device.prov.ztpEnabled`

Enable or disable Zero Touch Provisioning (ZTP).

0 - Disabled

1 - Enabled

For information, see Zero-Touch Provisioning.

`device.sec.configEncryption.key`

Set the configuration encryption key used to encrypt configuration files.

String

For more information, see the section on Configuration File Encryption.

Change causes system to restart or reboot.

`device.sec.coreDumpEncryption.enabled`

Determine whether to encrypt the core dump or bypass the encryption of the core dump.

0 - Encryption of the core dump is bypassed.

1 (default) - the core dump is encrypted.

`device.sec.TLS.customCaCert1( TLS Platform Profile 1 )`

Set the custom certificate to use for TLS Platform Profile 1 and TLS Platform Profile 2 and TLS Application Profile 1 and TLS Application Profile 2. The parameter `device.sec.TLS.profile.caCertList` must be configured to use a custom certificate. Custom CA certificates can't exceed 4096 bytes total size.

String

PEM format

device.sec.TLS.customDeviceCert1.privateKey
device.sec.TLS.customDeviceCert2.privateKey

Enter the corresponding signed private key in PEM format (X.509).

Size constraint is 4096 bytes for the private key.

device.sec.TLS.customDeviceCert1.publicCert
device.sec.TLS.customDeviceCert2.publicCert

Enter the signed custom device certificate in PEM format (X.509).

Size constraint is 8192 bytes for the device certificate.

device.sec.TLS.customDeviceCert1.set
device.sec.TLS.customDeviceCert2.set

Use to set the values for parameters `device.sec.TLS.customDeviceCertX.publicCert` and `device.sec.TLS.customDeviceCertX.privateKey`.

Size constraints are 4096 bytes for the private key and 8192 bytes for the device certificate.

0 (default) - Disabled

1 - Enabled

device.sec.TLS.profile.caCertList1 (TLS Platform Profile 1)

Choose the CA certificate(s) to use for TLS Platform Profile 1 and TLS Platform Profile 2 authentication:

Builtin - The built-in default certificate

BuiltinAndPlatform - The built-in and Custom #1 certificates

BuiltinAndPlatform2 - The built-in and Custom #2 certificates

All - Any certificate (built in, Custom #1 or Custom #2)

Platform1 - Only the Custom #1 certificate

Platform2 - Only the Custom #2 certificate

Platform1AndPlatform2 - Either the Custom #1 or Custom #2 certificate

device.sec.TLS.profile.cipherSuite1 (TLS Platform Profile 1)

Enter the cipher suites to use for TLS Platform Profile 1 and TLS Platform Profile 2

String

device.sec.TLS.profile.cipherSuiteDefault1 (TLS Platform Profile 1)

Determine the cipher suite to use for TLS Platform Profile 1 and TLS Platform profile 2.

0 - The custom cipher suite is used.

1 - The default cipher suite is used.

device.sec.TLS.profile.deviceCert1 (TLS Platform Profile 1)

Choose the device certificate(s) for TLS Platform Profile 1 and TLS Platform Profile 2 to use for authentication.

Builtin

Platform1

Platform2

`device.sec.TLS.profileSelection.dot1x`

Choose the TLS Platform Profile to use for 802.1X.

PlatformProfile1

PlatformProfile2

`device.sec.TLS.profileSelection.provisioning`

Set the TLS Platform Profile to use for provisioning.

PlatformProfile1

PlatformProfile2

Change causes system to restart or reboot.

`device.sec.TLS.profileSelection.syslog`

Set the TLS Platform Profile to use for syslog.

PlatformProfile1

PlatformProfile2

Change causes system to restart or reboot.

`device.sec.TLS.prov.strictCertCommonNameValidation`

0 - Disables common name validation.

1 (default) - Provisioning server always verifies the server certificate for the `commonName/SubjectAltName` match with the server hostname that the phone is trying to connect.

`device.sec.TLS.syslog.strictCertCommonNameValidation`

0

1 - Syslog always verifies the server certificate for the `commonName/SubjectAltName` match with the server hostname that the phone is trying to connect.

`device.showOOB`

Set to display the setup wizard when the system powers on as if the system is powering on for the first time.

1 (default) – The system displays a setup wizard when it powers on.

0 – The system doesn’t display a setup wizard when it powers on.

`device.sntp.gmtOffset`

Set the GMT offset, in seconds, to use for daylight saving time, corresponding to -12 to +13 hours.

-43200 to 46800

`device.sntp.gmtOffsetcityID`

Sets the correct time zone location description that displays on the phone menu and in the system web interface.

NULL (default)

0 to 127

For descriptions of all values, refer to the Time Zone Location Description.

`device.sntp.serverName`

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

IP address

Domain name string

`device.syslog.facility`

Determine a description of what generated the log message.

0 to 23

For more information, see RFC 3164.

`device.syslog.prependMac`

0

1 - The phone's MAC address is prepended to the log message sent to the syslog server.

Change causes system to restart or reboot.

`device.syslog.renderLevel`

Specify the logging level for the lowest severity of events to log in the syslog. When you choose a log level, the log includes all events of an equal or greater severity level, but it excludes events of a lower severity level.

0 or 1 - SeverityDebug(7).

2 or 3 - SeverityInformational(6).

4 - SeverityError(3).

5 - SeverityCritical(2).

6 - SeverityEmergency(0).

Change causes system to restart or reboot.

`device.syslog.serverName`

Set the syslog server IP address or domain name string.

IP address

Domain name string

`device.syslog.transport`

Set the transport protocol that the phone uses to write to the syslog server.

None - Transmission is turned off but the server address is preserved.

UDP

TCP

TLS

Diagnostics and Status Parameters

This section provides information about the diagnostics and status parameters for Poly Trio.

Reset to Factory Configuration Parameters

By default, only administrators can initiate a factory reset. However, you can make the **Reset to Factory** setting available to users.

`up.basicSettings.factoryResetEnabled`

0 (default) - Doesn't display the **Reset to Factory** option under **Basic** settings.

1 - Displays the **Reset to Factory** option under **Basic** settings.

`feature.restrictPerDataUploadMenu.enabled`

1 (default) - Displays the **Restrict Personal Data Upload** menu under **Basic** settings.

0 - Doesn't display the **Reset to Factory** menu under **Basic** settings.

`feature.clearPerInfoMenu.enabled`

1 (default) - Displays the **Clear Personal Information** menu under **Basic** settings.

0 - Doesn't display the **Clear Personal Information** menu under **Basic** settings.

`device.system.recoveryType`

Defines what settings the phone resets via MKC updater boot-up when a user tries a factory reset.
FullRecovery (default) - All settings are returned to factory default.

PreserveAdmin - All settings are returned to factory default except the administrator password.

CloudProv - All settings are returned to factory default except the administrator password and provisioning. Provisioning is changed to ZTP.

Scheduled Reboot Parameters

Use the following parameters to configure scheduled reboot times for Poly phones.

`prov.scheduledReboot.enabled`

0 (default)— Disables scheduled reboot.

1—Enables scheduled reboot.

`prov.scheduledReboot.periodDays`

Specify the time in days between scheduled reboots.

1 day (default)

1–365 days

`prov.scheduledReboot.time`

Specify a time to reboot the Poly phone. Use the 24 hour time format (hh:mm).

03:00 (default)

`prov.scheduledReboot.timeRandomEnd`

If this parameter is set to a specific time, the scheduled reboot occurs at a random time between the time set for `prov.scheduledReboot.time` and `prov.scheduledReboot.timeRandomEnd`. The time is in 24-hour format.

0–5 hours

hh:mm

Directories and Contacts Parameters

This section provides information about the directories and contacts parameters for Poly Trio.

Call List Parameters

Use the following parameters to configure call lists.

`callLists.collapseDuplicates`

Generic Base Profile - 1 (default)

1 - Consecutive incomplete calls to/from the same party and in the same direction are collapsed into one record in the calls list. The collapsed entry displays the number of consecutive calls.

0 - Each call is listed individually in the calls list.

`callLists.logConsultationCalls`

Generic Base Profile - 1 (default)

0 - Consultation calls not joined into a conference call aren't listed as separate calls in the calls list.

1 - Each consultation call is listed individually in the calls list.

`feature.callList.enabled`

1 (default) - Allows you to enable the missed, placed, and received call lists on all phone menus including the Home screen and dialpad.

0 - Disables all call lists.

`feature.callListMissed.enabled`

0 (Default) - The missed call list is disabled.

1 - The missed call list is enabled.

To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.

`feature.callListPlaced.enabled`

0 (Default) - The placed call list is disabled.

1 - The placed call list is enabled.

To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.

`feature.callListReceived.enabled`

0 (Default) - The received call list is disabled.

1 - The received call list is enabled.

To enable the missed, placed, or received call lists, `feature.callList.enabled` must be enabled.

`feature.exchangeCallLog.enabled`

If Base Profile is:

Generic - 0 (default)

1 - The Exchange call log feature is enabled, user call logs are synchronized with the server, and the user call history of Missed, Received, and outgoing calls can be retrieved on the phone.

You must also enable the parameter `feature.callList.enabled` to use the Exchange call log feature.

0 - The Exchange call log feature is disabled, the user call log history can't be retrieved from the Exchange server, and the phone generates call logs locally.

Corporate Directory Parameters

Use the parameters in the following list to configure the corporate directory.

Note that the exact configuration of a corporate directory depends on the LDAP server you use.

Note: For detailed explanations and examples of all currently supported LDAP directories, see *Technical Bulletin 41137: Best Practices When Using Corporate Directory* on Poly phones at [Polycom Engineering Advisories and Technical Notifications].

`dir.corp.address`

Set the IP address or hostname of the LDAP server interface to the corporate directory.

Null (default)

IP address

Hostname

FQDN

Change causes system to restart or reboot.

`dir.corp.allowCredentialsFromUI.enabled`

Enable or disable prompting users to enter LDAP credentials on the phone when accessing the Corporate Directory.

Note: Users are only prompted to enter their credentials when credentials are not added through configuration or after a login failure.

0 (default) - Disabled

1 - Enabled

`dir.corp.alt.transport`

Choose a transport protocol used to communicate to the corporate directory.

TCP (default)

TLS

`dir.corp.attribute.x.addstar`

Determine if the wild-card character, asterisk(*), is appended to the LDAP query field.

0 - Wild-card character is not appended.

1 (default) - Wild-card character is appended.

Change causes system to restart or reboot.

`dir.corp.attribute.x.filter`

Set the filter string for this parameter, which is edited when searching.

Null (default)

UTF-8 encoding string

Change causes system to restart or reboot.

`dir.corp.attribute.x.label`

Enter the label that shows when data is displayed.

Null (default)

UTF-8 encoding string

Change causes system to restart or reboot.

`dir.corp.attribute.x.name`

Enter the name of the parameter to match on the server. Each name must be unique; however, a global address book entry can have multiple parameters with the same name. You can configure up to eight parameters (x = 1 to 8).

Null (default)

UTF-8 encoding string

Change causes system to restart or reboot.

`dir.corp.attribute.x.searchable`

Determine whether quick search on parameter x (if x is 2 or more) is enabled or disabled.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`dir.corp.attribute.x.sticky`

Sets whether the filter string criteria for attribute x is reset or retained after a phone reboot. If you set an attribute to be sticky (set this parameter to 1), a '*' displays before the label of the attribute on the phone.

0 (default) – Reset after a phone reboot.

1 – Retain after a phone reboot.

Change causes system to restart or reboot.

`dir.corp.attribute.x.type`

Define how x is interpreted by the phone. Entries can have multiple parameters of the same type. If the user saves the entry to the local contact directory on the phone, first_name, last_name, and phone_number are copied. The user can place a call to the phone_number and SIP_address from the global address book directory.

first_name

last_name (default)

phone_number

SIP_address

H323_address URL

other

Change causes system to restart or reboot.

`dir.corp.auth.useLoginCredentials`

0 (default) - Disabled

1 - Enabled

`dir.corp.autoQuerySubmitTimeout`

Set the timeout in seconds between when the user stops entering characters in the quick search and when the search query is automatically submitted.

0 (default)

0 - 60

Change causes system to restart or reboot.

`dir.corp.backGroundSync`

Determine if background downloading from the LDAP server is enabled or disabled.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`dir.corp.backGroundSync.period`

Set the time in seconds the corporate directory cache is refreshed after the corporate directory feature has not been used for the specified period of time.

86400 (default)

3600 to 604800

Change causes system to restart or reboot.

`dir.corp.baseDN`

Enter the base domain name, which is the starting point for making queries on the LDAP server.

Null (default)

UTF-8 encoding string

Change causes system to restart or reboot.

`dir.corp.bindOnInit`

Enable or disabled use of bind authentication on initialization.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`dir.corp.cacheSize`

Set the maximum number of entries that can be cached locally on the phone.

128 (default)

32 to 256

Change causes system to restart or reboot.

dir.corp.customError

Enter the error message to display on the phone when the LDAP server finds an error.

Null (default)

UTF-8 encoding string

dir.corp.domain

Enter the port that connects to the server if a full URL is not provided.

0 to 255

dir.corp.filterPrefix

Enter the predefined filter string for search queries.

(objectclass=person) (default)

UTF-8 encoding string

Change causes system to restart or reboot.

dir.corp.pageSize

Set the maximum number of entries requested from the corporate directory server with each query.

64 (default)

32 (default for Trio 8300)

8 to 64

Change causes system to restart or reboot.

dir.corp.password

Enter the password used to authenticate to the LDAP server.

Null (default)

UTF-8 encoding string

dir.corp.persistentCredentials

Enable to securely store and encrypt LDAP directory user credentials on the phone. Enable `dir.corp.allowCredentialsFromUI.enabled` to allow users to enter credentials on the phone.

Note: If you disable the feature after enabling it, then all the saved user credentials are deleted.

0 (default) - Disabled

1 - Enabled

dir.corp.port

Enter the port that connects to the server if a full URL is not provided.

389 (default for TCP)

636 (default for TLS)

0

Null

1 to 65535

Change causes system to restart or reboot.

`dir.corp.querySupportedControlOnInit`

Enable the phone to make an initial query to check the status of the server when booting up.

0 - Disabled

1 (default) - Enabled

`dir.corp.scope`

sub (default) - a recursive search of all levels below the base domain name is performed.

one - a search of one level below the base domain name is performed.

base - a search at the base domain name level is performed.

Change causes system to restart or reboot.

`dir.corp.serverSortNotSupported`

0 (default) - The server supports server-side sorting.

1 - The server does not support server-side sorting, so the phone handles the sorting.

`dir.corp.sortControl`

Determine how a client can make queries and sort entries.

0 (default) - Leave sorting as negotiated between the client and server.

1 - Force sorting of queries, which causes excessive LDAP queries and should only be used to diagnose LDAP servers with sorting problems.

Change causes system to restart or reboot.

`dir.corp.transport`

Specify whether a TCP or TLS connection is made with the server if a full URL is not provided.

TCP (default)

TLS

Null

Change causes system to restart or reboot.

`dir.corp.user`

Enter the user name used to authenticate to the LDAP server.

Null (default)

UTF-8 encoding string

`dir.corp.viewPersistence`

0 (default) - The corporate directory search filters and browsing position are reset each time the user accesses the corporate directory.

1 - The search filters and browsing position from the previous session are displayed each time the user accesses the corporate directory.

Change causes system to restart or reboot.

`dir.corp.vlv.allow`

Determine whether virtual view list (VLV) queries are enabled and can be made if the LDAP server supports VLV.

0 (default)

1

Change causes system to restart or reboot.

`dir.corp.vlv.sortOrder`

Enter the list of parameters, in exact order, for the LDAP server to use when indexing. For example: `sn, givenName, telephoneNumber` .

Null (default)

list of parameters

Change causes system to restart or reboot.

`feature.contacts.enabled`

1 (default) - The Contacts icon displays on the Home screen, the global menu, and in the dialer.

0 - Disable display of the Contacts icon.

`feature.corporateDirectory.enabled`

0 (default) - The corporate directory feature is disabled and the icon is hidden.

1 - The corporate directory is enabled and the icon shows.

Local Contact Directory File Size Parameters

Use the following parameters to set the size of the local contact directory.

The maximum local directory size is limited based on the amount of flash memory in the phone and varies by phone model. Configure a provisioning server that allows uploads to ensure a back-up copy of the directory when the phone reboots or loses power.

`dir.local.nonVolatile.maxSize`

Set the maximum file size of the local contact directory stored on the phone's non-volatile memory.

1 - 100 KB

`dir.local.volatile`

0 (default) - The phone uses non-volatile memory for the local contact directory.

1 - Enables the use of volatile memory for the local contact directory.

`dir.local.volatile.maxSize`

Sets the maximum file size of the local contact directory stored on the phone's volatile memory.

1 - 200 KB

Parameter Elements for the Local Contact Directory

The following table describes each of the parameter elements and permitted values that you can use in the local contact directory.

Local Contact Directory Parameter Elements

Element	Definition	Permitted Values
fn	The contact's first name	UTF-8 encoded string of up to 40 bytes1
ln	The contact's last name	UTF-8 encoded string of up to 40 bytes1
ct	Contact Used by the phone to address a remote party in the same way that a user manually dials a string of digits or a SIP URL . Also used to associate incoming callers with a particular directory entry. The maximum field length is 128 characters. **Note:** You can't duplicate this field or leave it `Null`.	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL
sd	Speed Dial Index Associates a particular entry with a speed dial key for one-touch dialing or dialing.	Null, 1 to 20
lb	The label for the contact The label of a contact directory item is by default the label attribute of the item. If the label attribute does not exist or is `Null`, then the first and last names form the label. A space is added between first and last names.	UTF-8 encoded string of up to 40 bytes1
pt	Protocol The protocol to use when placing a call to this contact.	SIP, H323, or Unspecified

Element	Definition	Permitted Values
rt	Ring Tone When incoming calls match a directory entry, this field specifies the ringtone to use.	Null, 1 to 21
dc	Divert Contact The address to forward calls to if the Auto Divert feature is enabled.	UTF-8 encoded string containing digits (the user part of a SIP URL) or a string that constitutes a valid SIP URL
ad	Auto Divert If set to `1`, callers that match the directory entry are diverted to the address specified for the divert contact element. **Note:** If auto-divert is enabled, it has precedence over auto-reject.	0 or 1
ar	Auto Reject If set to `1`, callers that match the directory entry specified for the auto reject element are rejected. **Note:** If auto divert is also enabled, it has precedence over auto reject.	0 or 1
bw	Buddy Watching If set to `1`, this contact is added to the list of watched phones.	0 or 1
bb	Buddy Block If set to `1`, this contact is blocked from watching this phone.	0 or 1
up	User Photo The contact's photo icon set by the icons.x parameter.	1-24

Local Contact Directory Parameters

The following parameters configure the local contact directory.

`contactPhotoIntegration.hideMyPhoto`

Don't show the signed-in user's photo on the line key but still show other users' photos.

0 (default) - Disable the Hide My Photo feature.

1 - Enable the Hide My Photo feature.

`dir.local.contacts.maxNum`

Set the maximum number of contacts that can be stored in the Local Contact Directory. The maximum number varies by phone model, refer to section 'Maximum Capacity of the Local Contact Directory'.

- 2000 (default)
- Maximum 3000 contacts

Change causes system to restart or reboot.

`dir.local.readonly`

0 (default) - Disable read-only protection of the local Contact Directory.

1 - Enable read-only protection of the local Contact Directory.

`feature.directory.enabled`

0 - The local contact directory is disabled.

1 (default) - The local contact directory is enabled.

`dir.search.field`

Specify whether to sort contact directory searches by first name or last name.

0 (default) - Last name.

1 - First name.

`voIpProt.SIP.specialEvent.checkSync.downloadDirectory`

0 (default) - The phone downloads updated directory files after receiving a checksync NOTIFY message.

1 - The phone downloads the updated directory files along with any software and configuration updates after receiving a checksync NOTIFY message. The files are downloaded when the phone restarts, reboots, or when the phone downloads any software or configuration updates.

Note: The parameter `hotelingMode.type` set to `2` or `3` overrides this parameter.

`dir.local.passwordProtected`

Specify whether you are prompted for an Admin/User password when adding, editing, or deleting contacts in the Contact Directory.

0 - Disabled (default)

1 - Enabled

`feature.pauseAndWaitDigitEntryControl.enabled`

1 (default) - Enable processing of control characters in the contact phone number field. When enabled, "," or "p" control characters cause a one-second pause.

For example, a "," or "p" control character causes a one-second pause. A ";" or "w" control character causes a user prompt that allows a user-controlled wait. Subsequent digits entered to the contact field are dialed automatically.

0 - Disable processing of control characters.

`up.regOnPhone`

0 (default) - Contacts you assign to a line key display on the phone in the position assigned.

1 - Contacts you assign to a line key are pushed to the attached expansion module.

Change causes system to restart or reboot.

Speed Dial Contacts Parameters

After setting up your per-phone directory file (**<MACaddress>-directory.xml**), enter a number in the speed dial `<sd>`field to display a contact directory entry as a speed dial contact on the phone. Speed dial entries automatically display on unused line keys on the phone and are assigned in numerical order.

On some call servers, enabling presence for an active speed dial contact displays that contact's status on the speed dial's line key label.

Use the parameter below, which identifies the directory XML file and the parameters you need to set up your speed dial contacts.

`dir.local.contacts.maxFavIx`

Configure the maximum number of speed dial contacts that can display on the Home screen.

Enter a speed dial index number in the <sd>x</sd> element in the <MAC address>-directory.xml file to display a contact directory entry as a speed dial key on the phone. Speed dial contacts are assigned to unused line keys and to entries in the phone's speed dial list in numerical order.

Network Parameters

This section provides information about the network parameters for Poly Trio.

Bluetooth and NFC-Assisted Bluetooth Parameters

Use the following parameters to configure Bluetooth on Poly Trio systems and NFC on the Poly Trio 8800 systems.

`bluetooth.beacon.ipAddress.enabled`

Set to send the IP address of the system over Bluetooth.

1 (default) - Enables sending the system IP address over Bluetooth. Turns Bluetooth radio on when `feature.bluetooth.enabled = 1`.

0 - Disables sending the system IP address over Bluetooth

Note: Enable the parameter `feature.bluetooth.enabled` to use this feature.

`bluetooth.device.audio.enabled`

Set to enable or disable audio playback through a Bluetooth connection.

0 - Bluetooth device audio is disabled.

1 (default) - Bluetooth device audio is enabled.

Note: This feature can also be controlled from the Poly Trio system user interface.

`bluetooth.device.discoverable`

1 (default) - This device is discoverable for Bluetooth pairing.

0 - This device is not discoverable for Bluetooth pairing.

`bluetooth.device.maxPaired`

Set the limit for the maximum number of Bluetooth devices that can be paired with a Poly Trio system.
10 (default)

0 - 10

`bluetooth.device.name`

NULL (default)

UTF-8 string

Enter the name of the device that broadcasts over Bluetooth to other devices.

`bluetooth.device.pairedTimeout`

Set the timeout limit for automatically unpairing Bluetooth devices when `bluetooth.device.maxPaired` is set to 0.

30 (default)

30 - 1800 (in minutes)

`bluetooth.discoverableTimeout`

0 (default) - Other devices can always discover this device over Bluetooth.

0 - 3600 seconds

Set the time in seconds after which other devices can discover this device over Bluetooth.

`bluetooth.pairedDeviceMemorySize`

10 (default)

0 - 10

`bluetooth.radioOn`

0 - The Bluetooth radio (transmitter/receiver) is off.

1 (default) - The Bluetooth radio is on. The Bluetooth radio must be turned on before other devices can connect to this device over Bluetooth.

`feature.bluetooth.enabled`

For high security environments.

1 (default) - Bluetooth connection is enabled and the Bluetooth menu displays.

0 - Bluetooth connection is disabled and the Bluetooth icon does not display.

`feature.nfc.enabled`

0 - Disable the NFC pairing feature for the Trio 8800 system.

1 - Enables the NFC pairing feature, and users can pair NFC-capable devices to the Trio 8800 system.

ICE Parameters

Enter a short description of your reference here (optional).

Enter the actual information in this section (optional).

`feature.nat.ice.enabled`

Default - 0 Disable ICE

1 - Enable ICE

`reg.x.ice.answerIceOfferInSessionProgress`

Default - 0 Disable.

1 - Enable.

When enabled, incoming calls will trigger ICE candidate collection so that a 183 with SDP can be sent. When disabled, the phone will reply to incoming calls with a 180 response and the ICE candidates are provided only when the call is answered via the 200 OK.

`reg.x.ice.enabled`

Default - 0 Disable ICE.

1 - Enable ICE. This parameter is only applicable when reg.x.nat.traversal.mode is not set to "Auto". When set to 1, the ICE protocol will be enabled and available.

`reg.x.ice.holdWithNewSession`

Default - 0 Disable.

1 - Enable.

When enabled, every time a call is put on hold, the existing ICE negotiation is discarded and the phone will collect new ICE candidates from the STUN and/or TURN server to send in the Hold request.

`reg.x.ice.resumeWithNewSession`

Default - 0 Disable.

1 - Enable.

When enabled, every time a call is resumed, the existing ICE negotiation is discarded and the phone will collect new ICE candidates from the STUN and/or TURN server to send in the resume request.

`tcpIpApp.ice.NetworkMode`

Default - TCPUDP.

1 - Enable.

Based on the mode, gathers all possible ice candidates.

`tcpIpApp.ice.password`

<password>

The password required for authentication to your TURN server.

`tcpIpApp.ice.reflexiveChecksRequired`

0 - Disable

1 - Default

When enabled, reflexive ICE candidates will be collected as part of the ice collection process.

`tcplpApp.ice.stun.udpPort`

3478 - Default

1 to 65535 - Values

The UDP port on the TURN server the phone will use for relaying media.

`tcpIpApp.ice.turn.udpPort`

443 - Default

<port number>

The UDP port on the TURN server the phone will use for relaying media.

`tcplpApp.tcp.enabled`

1 - Default.

When enabled, the phone will make both a TCP and UDP connection to the TURN server to collect ICE candidates. By default (disabled) only UDP will be used for relaying media.

Note: This default has changed from previous SW release.

`tcpIpApp.ice.turn.tcpPort`

The TCP port on the TURN server the phone will use for relaying media.

`tcpIpApp.ice.username`

<username>

The username required for authentication to your TURN server.

`voIpProt.SIP.optionSupport.enabled`

0 - Default.

1 - Enable.

When enabled, the phone identifies itself as ICE capable by adding the "sip.ice" attribute into the contact header of a registration (per RFC 5768 section 3.1). The "ice" option will not be used within the requires header.

`voIpProt.SDP.rtcpMultiplexing`

0 - Default.

1 - Enable.

When enabled, RTCP will be sent and received on the same port that RTP is sent from, as described in RFC 5761. The default behaviour (disabled) uses RTP port + 1 for RTCP.

IPv4 Parameters

You can configure IPv4 using the following parameters.

`device.icmp.ipv4IcmpIgnoreRedirect.set`

0 (default) - The phone doesn't allow you to use the `device.icmp.ipv4IcmpIgnoreRedirect` parameter to configure Enhanced IPv4 ICMP Management feature.

1 - The phone allows you to use the `device.icmp.ipv4IcmpIgnoreRedirect` parameter to configure Enhanced IPv4 ICMP Management feature.

`device.icmp.ipv4IcmpIgnoreRedirect`

1 (default) - The phone ignores ICMP redirect requests for an alternate path from the router or gateway.

0 - The phone allows ICMP redirects.

IPv6 Parameters

Use the parameters in the following list to enable and configure IPv6.

`device.dhcp.bootSrvUseOpt`

Specifies the source for the boot server address for the phone. It can take values from 0 to 9.

In IPv4 mode, the following values are applicable:

- 0 (Default) - The phone gets the boot server address from option 66.
- 1 - The phone gets the boot server details from the custom option number provided through DHCP.
- 2 - The phone uses the boot server configured through the **Server** Menu.
- 3 - The phone uses the custom option first or uses option 66 if the custom option is not present

In IPv6 mode, the following values are applicable:

- 4 - The phone uses the boot server configured through the **Server** menu.
- 5 - The phone uses the boot server option provided through DHCPv6.

In Dual Stack Mode (IPv4/IPv6 mode), the following values are applicable:

- 6 - The phone uses the boot server configured through the **Server** menu.
- 7 - The phone gets the boot server details from DHCPv6 option or the option 66 on DHCP server.
- 8 - The phone gets the boot server details through DHCPv6 or through the custom option configured on DHCP server for the provisioning.
- 9 - The phone gets the boot server from DHCPv6 option or custom option or option 66 configured on DHCP server for the provisioning.

`device.ipv6.icmp.echoReplies`

NULL (default)

0

1

`device.ipv6.icmp.echoReplies.set`

0 (default)

1

`device.ipv6.icmp.genDestUnreachable`

0

1

`device.ipv6.icmp.genDestUnreachable.set`

0

1

`device.icmp.ipv6IcmpIgnoreRedirect`

1 (default) - The phone ignores ICMP redirect requests for an alternate path from the router or gateway.

0 - The phone allows ICMP redirects.

`device.icmp.ipv6IcmpIgnoreRedirect.set`

0 (default) - The phone doesn’t allow you to use `device.icmp.ipv6IcmpIgnoreRedirect` parameter to configure Enhanced IPv6 ICMP Management.

1 - The phone allows you to use `device.icmp.ipv6IcmpIgnoreRedirect` parameter to configure Enhanced IPv6 ICMP Management.

`device.ipv6.icmp.txRateLimiting`

0

1

`device.ipv6.icmp.txRateLimiting.set`

0 to 6000

`device.net.ipStack`

Configures the IPv4, IPv6, or dual stack mode for the phone.

0 (Default) - IPv4 is enabled and IPv6 is disabled.

1 - IPv6 is enabled and IPv4 is disabled.

2 - Dual stack is enabled and phone operates on both IPv4 and IPv6 networks at the same time.

`device.net.ipv6AddrDisc`

Specify whether the IPv6 address and related parameters for the phone are obtained from DHCPv6 or SLAAC or statically configured for the phone.

1 (Default) - IPv6 global address and options are configured from DHCPv6.

2 - IPv6 global address is configured using prefixes received from Router Advertisements (RA) and options are configured from stateless DHCPv6.

0 - You must configure IPv6 global address and options manually.

`device.net.ipv6Address`

Specify a valid global IPv6 unicast address for the phone.

Null (default)

`device.net.ipv6Gateway`

Specify theIPv6 address of the default gateway for the phone.

Null (default)

`device.net.ipv6LinkAddress`

Specifies a valid Link Local IPv6 address for the phone.

Null (default)

`device.net.ipv6PrivacyExtension`

Configure whether or not the IPv6 global and link local addresses are in 64-bit Extended Unique Identifier (EUI-64) format.

0 (Default) - IPv6 global and link local addresses are in EUI-64 format.

1 - Global and link local IPv6 addresses are not in EUI-64 format. Instead, the last 48 bits for the IPv6 address are generated randomly.

`device.net.ipv6ULAAddress`

Specifies a valid Unique Local IPv6 address (ULA) for the phone.

Null (default)

`device.net.preferredNetwork`

Specify IPv4 or IPv6 as the preferred network in a Dual Stack mode.

1 (default) - Specifies IPv6 as a preferred network.

0 - Specifies IPv4 as a preferred network.

ipv6.mldVersion

2 (default)

1

voipProt.SIP.anat.enabled

Enables or disables Alternative Network Address Types (ANAT).

0 (default) - ANAT is disabled.

1 - ANAT is enabled.

IP Type-of-Service Parameters

You can configure the IP TOS feature specifically for RTP and call control packets, such as SIP signaling packets.

Type of Service (ToS) and the Differentiated Services Code Point (DSCP) allows specification of a datagrams desired priority and routing through low-delay, high-throughput, or highly-reliable networks.

The IP ToS header consists of four ToS bits and a 3-bit precedence field. DSCP replaces the older ToS specification and uses a 6-bit DSCP in the 8-bit differentiated services field (DS field) in the IP header.

The parameters listed below configure the type of service field RTP and call control packets for Quality of Service (QoS).

qos.ethernet.tcpQosEnabled

0 (default) - The phone does not send configured QoS priorities for SIP over TCP transport.

1 - The phone sends configured QoS priorities for SIP over TCP transport.

Change causes system to restart or reboot.

qos.ip.callControl.dscp

Specify the DSCP of packets.

If the value is set to the default NULL the phone uses `qos.ip.callControl.*` parameters.

If the value is not NULL, this parameter overrides `qos.ip.callControl.*` parameters.

Change causes system to restart or reboot.

qos.ip.callControl.max_reliability

Set the max reliability bit in the IP ToS field of the IP header used for call control.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

qos.ip.callControl.max_throughput

Set the throughput bit in the IP ToS field of the IP header used for call control.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.callControl.min_cost`

Set the min cost bit in the IP ToS field of the IP header used for call control.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.callControl.min_delay`

Set the min delay bit in the IP ToS field of the IP header used for call control.

1 (default) - The bit is set.

0 - The bit in the IP ToS field of the IP header is not set.

Change causes system to restart or reboot.

`qos.ip.callControl.precedence`

Set the min delay bit in the IP ToS field of the IP header used for call control.

5 (default)

0 - 7

Change causes system to restart or reboot.

`qos.ip.rtp.dscp`

Specify the DSCP of packets.

If the value is set to the default NULL, the phone uses `quality.ip.rtp.*` parameters.

If the value is not NULL, this parameter overrides `quality.ip.rtp.*` parameters.

- Null (default)
- 0 to 63
- EF
- Any of AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42, AF43

Change causes system to restart or reboot.

`qos.ip.rtp.max_reliability`

Set the max reliability bit in the IP ToS field of the IP header used for RTP.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.max_throughput`

Set the throughput bit in the IP ToS field of the IP header used for RTP.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.min_cost`

Set the min cost bit in the IP ToS field of the IP header used for RTP.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.min_delay`

Set the min delay bit in the IP ToS field of the IP header used for RTP.

1 (default) - The bit is set.

0 - The bit in the IP ToS field of the IP header is not set.

Change causes system to restart or reboot.

`qos.ip.rtp.precedence`

Set the precedence bit in the IP ToS field of the IP header used for RTP.

5 (default)

0 - 7

Change causes system to restart or reboot.

`qos.ip.rtp.video.dscp`

Allows you to specify the DSCP of packets.

If the value is set to the default NULL, the phone uses `qos.ip.rtp.video.*` parameters.

If the value is not NULL, this parameter overrides `qos.ip.rtp.video.*` parameters.

- NULL (default)
- 0 to 63
- EF
- Any of AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42, AF43

Change causes system to restart or reboot.

`qos.ip.rtp.video.max_reliability`

Set the reliability bits in the IP ToS field of the IP header used for RTP video.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.video.max_throughput`

Set the throughput bits in the IP ToS field of the IP header used for RTP video.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.video.min_cost`

Set the min cost bits in the IP ToS field of the IP header used for RTP video.

0 (default) - The bit in the IP ToS field of the IP header is not set.

1 - The bit is set.

Change causes system to restart or reboot.

`qos.ip.rtp.video.min_delay`

Set the min delay bits in the IP ToS field of the IP header used for RTP video.

1 (default) - The bit is set.

0 - The bit in the IP ToS field of the IP header is not set.

Change causes system to restart or reboot.

`qos.ip.rtp.video.precedence`

Set the precedence bits in the IP ToS field of the IP header used for RTP video.

5 (default)

0 - 7

Change causes system to restart or reboot.

Network Address Translation Parameters

You can configure the external IP addresses and ports used by the NAT on the phone's behalf on a per-phone basis.

Use the parameters in the following list to configure NAT.

`nat.ip`

Specifies the IP address to advertise within SIP signaling. This should match the external IP address used by the NAT device.

Null (default)

IP address

Change causes system to restart or reboot.

`nat.keepalive.interval`

The keep-alive interval in seconds. Sets the interval at which phones sends a keep-alive packet to the gateway/NAT device to keep the communication port open so that NAT can continue to function. If Null or 0, the phone does not send out keep-alive messages.

0 (default)

0 - 3600

`nat.mediaPortStart`

The initially allocated RTP port. Overrides the value set for `tcpIpApp.port.rtp.mediaPortRangeStart` parameter.

0 (default)

0 - 65440

Change causes system to restart or reboot.

`nat.signalPort`

The port used for SIP signaling. Overrides the `voIpProt.local.port` parameter.

0 (default)

1024 - 65535

Network Signaling Validation Parameters

The following list includes the parameters you can use to specify the validation type, method, and the events for validating incoming network signaling.

`voIpProt.SIP.requestValidation.x.method`

Null (default) - No validation is made.

Source - Ensure request is received from an IP address of a server belonging to the set of target registration servers.

digest - Challenge requests with digest authentication using the local credentials for the associated registration (line).

both or all - Apply both of the above methods.

Change causes system to restart or reboot.

`voIpProt.SIP.requestValidation.x.request`

Sets the name of the method for which validation will be applied.

Null (default)

INVITE, ACK , BYE, REGISTER, CANCEL, OPTIONS, INFO, MESSAGE, SUBSCRIBE, NOTIFY, REFER, PRACK, UPDATE

Note: Intensive request validation may have a negative performance impact due to the additional signaling required in some cases.

Change causes system to restart or reboot.

`voIpProt.SIP.requestValidation.x.request.y.event`

Determines which events specified with the Event header should be validated; only applicable when `voIpProt.SIP.requestValidation.x.request` is set to `SUBSCRIBE` or `NOTIFY`.

Null (default) - all events will be validated.

A valid string - specified event will be validated.

Change causes system to restart or reboot.

Provisional Polling Parameters

Use the parameters in the following list to configure provisional polling.

Note: If `prov.startupCheck.enabled` is set to 0, then the phones do not look for the sip.ld or the configuration files when they reboot, lose power, or restart. Instead, they look only when receiving a checksync message, a polling trigger, or a manually started update from the menu or web UI.

Some files such as bitmaps, .wav, the local directory, and any custom ringtones are downloaded each time as they are stored in RAM and lost with every reboot.

`prov.polling`

To enable polling and set the mode, period, time, and time end parameters.

`prov.polling.enabled`

0 (default) - Disables the automatic polling for upgrades.

1 - Initiates the automatic polling for upgrades.

`prov.polling.mode`

The polling modes for the provisioning server.

`abs` (default) – The phone polls every day at the time specified by `prov.polling.time` .

`rel` – The phone polls after the number of seconds specified by `prov.polling.period`.

`random` – The phone polls at random between a starting time set in `prov.polling.time` and an end time set in `prov.polling.timeRandomEnd`.

If you set the polling period in `prov.polling.period` to a time greater than 86400 seconds (one day) polling occurs on a random day within that polling period and only between the start and end times. The day within the period is decided based upon the phones MAC address and does not change with a reboot whereas the time within the start and end is calculated again with every reboot

`prov.polling.period`

The polling period is calculated in seconds and is rounded up to the nearest number of days in an absolute and random mode. If this is set to a time greater than 86400 (one day) polling occurs on a random day based on the phone's MAC address.

86400 (default) - Number of seconds in a day.

Integer - An integer value greater than 3600 seconds.

`prov.polling.time`

The start time for polling on the provisioning server.

03:00 (default)

hh:mm

`prov.polling.timeRandomEnd`

The stop time for polling on the provisioning server.

Null (default)

hh:mm

Example Provisional Polling Configuration

The following are examples of polling configurations.

- If `prov.polling.mode` is set to rel and `prov.polling.period` is set to **7200**, the phone polls every two hours.
- If `prov.polling.mode` is set to abs and `prov.polling.timeRandomEnd` is set to **04:00**, the phone polls at 4am every day.

- If `prov.polling.mode` is set to random, `prov.polling.period` is set to **604800 (7 days)**, `prov.polling.time`is set to `01:00` ,`prov.polling.timeRandomEnd` is set to **05:00**, and you have 25 phones, a random subset of those 25 phones, as determined by the MAC address, polls randomly between 1am and 5am every day.
- If `prov.polling.mode` is set to **abs** and `prov.polling.period` is set to **2328000**, the phone polls every 20 days.

Server Redundancy Parameters

Use the parameters in the following list to set up server redundancy for your environment.

reg.x.auth.optimizedInFailover

Set the destination for the first new SIP request when failover occurs.

0 (default) - The SIP request is sent to the server with the highest priority in the server list.

1 - The SIP request is sent to the server that sent the proxy authentication request.

reg.x.outboundProxy.failOver.failBack.mode

The mode for failover failback (overrides `reg.x.server.y.failOver.failBack.mode`).

duration (default) - The phone tries the primary server again after the time specified by `reg.x.outboundProxy.failOver.failBack.timeout` expires.

newRequests - All new requests are forwarded first to the primary server regardless of the last used server.

DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL you configured for the server the phone is registered to.

reg.x.outboundProxy.failOver.failBack.timeout

3600 (default) - The time to wait (in seconds) before failback occurs (overrides `reg.x.server.y.failOver.failBack.timeout`).

0, 60 to 65535 - The phone does not fail back until a failover event occurs with the current server.

reg.x.outboundProxy.failOver.failRegistrationOn

1 (default) - The global and per-line `reRegisterOn` parameter is enabled and the phone silently invalidates an existing registration.

0 - The global and per-line `reRegisterOn` parameter is enabled and existing registrations remain active.

reg.x.outboundProxy.failOver.onlySignalWithRegistered

1 (default) - The global and per-line `reRegisterOn` and `failRegistrationOn` parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs.

0 - The global and per-line `reRegisterOn` and `failRegistrationOn` parameters are enabled, signaling is accepted from and sent to a server that has failed.

reg.x.outboundProxy.failOver.reRegisterOn

This parameter overrides `reg.x.server.y.failOver.reRegisterOn`.

0 (default) - The phone won't attempt to register with the secondary server.

1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server.

`reg.x.outboundProxy.port`

The port of the SIP server to which the phone sends all requests.

0 - (default)

1 to 65535

`reg.x.outboundProxy.transport`

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

DNSnaptr (default)

DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly

`reg.x.server.y.failOver.concurrentRegistration`

0 (default) - If 0 and `failOver.reRegisterOn` is set to 1, add this server to the set of redundant failover servers.

1 - This server registers concurrently with other servers for this registration.

Note that the default value of the new parameter `reg.x.server.y.failOver.concurrentRegistration=0` effective UC Software 5.5.2 for Poly Trio systems changes default behavior in previous releases. Prior to UC Software 5.5.2, the server you specify in `y` concurrently registers with other configured servers. As of UC Software 5.5.2, server `y` is added to the set of redundant failover servers. If you want to register the server concurrently with other servers set `reg.x.server.y.failOver.concurrentRegistration=1`.

`voIpProt.server.y.failOver.concurrentRegistration`

0 (default) - If 0 and `failOver.reRegisterOn` is set to 1, add this server to the set of redundant failover servers.

1 - This server registers concurrently with other servers.

The default value of the new parameter `voIpProt.server.y.failOver.concurrentRegistration=0` effective UC Software 5.5.2 for Poly Trio systems changes default behavior in previous releases. Prior to UC Software 5.5.2, the server you specify in `y` concurrently registers with other configured servers. As of UC Software 5.5.2, server `y` is added to the set of redundant failover servers. If you want to register the server concurrently with other servers set `voIpProt.server.y.failOver.concurrentRegistration=1`.

`voIpProt.server.x.failOver.failBack.mode`

Specify the failover failback mode.

duration (default) - The phone tries the primary server again after the time specified by `voIpProt.server.x.failOver.failBack.timeout`

newRequests - All new requests are forwarded first to the primary server regardless of the last used server.

DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.

registration - The phone tries the primary server again when the registration renewal signaling begins.

`voIpProt.server.x.failOver.failBack.timeout`

If `voIpProt.server.x.failOver.failBack.mode` is set to duration, this is the time in seconds after failing over to the current working server before the primary server is again selected as the first server to forward new requests. Values between 1 and 59 result in a timeout of 60. 0 means do not fail-back until a fail-over event occurs with the current server.

3600 (default)

0, 60 to 65535

`voIpProt.server.x.failOver.failRegistrationOn`

1 (default) - When set to 1, and the global or per-line `reRegisterOn` parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over.

0 - When set to 0, and the global or per-line `reRegisterOn` parameter is enabled, existing registrations remain active. This means that the phone attempts failback without first attempting to register with the primary server to determine if it has recovered.

`voIpProt.server.x.failOver.onlySignalWithRegistered`

1 (default) - When set to 1, and the global or per-line `reRegisterOn` and `failRegistrationOn` parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server.

0 - When set to 0, and the global or per-line `reRegisterOn` and `failRegistrationOn` parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).

`voIpProt.server.x.failOver.reRegisterOn`

0 (default) - When set to 0, the phone won't attempt to register with the second.

1 - When set to 1, the phone attempts to register with (or by, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server.

SIP Instance Parameter

Use the following parameter to enable a SIP instance on a registered line.

`reg.x.gruu`

The parameter `reg.x.gruu` provides a contact address to a specific user agent (UA) instance, which helps to route the request to the UA instance and is required in cases in which the REFER request must be routed to the correct UA instance. Refer to the following list for the parameters to configure this feature.

1 - The phone sends sip.instance in the REGISTER request.

0 (default) - The phone does not send sip.instance in the REGISTER request.

SIP Subscription Timers Parameters

Use the parameters in the following list to configure when a SIP subscription expires and when expiration dates overlap.

`voIpProt.server.x.subscribe.expires`

The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period.

3600 - (default)

10 - 2147483647

`voIpProt.server.x.subscribe.expires.overlap`

The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server.

60 - (default)

5 - 65535 seconds

`reg.x.server.y.subscribe.expires`

The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period.

3600 seconds - (default)

10 - 2147483647 (seconds)

You can use this parameter in conjunction with `reg.x.server.y.subscribe.expires.overlap` .

`reg.x.server.y.subscribe.expires.overlap`

The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server.

60 seconds (default)

5 - 65535 seconds

Static DNS Parameters

Use the following parameters to configure static DNS settings.

`reg.x.address`

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

Null (default)

String address

`reg.x.server.y`

Specify the call server used for this registration.

`reg.x.server.y.specialInterop`

Specify the server-specific feature set for the line registration.

All other phones:

Standard (default), GENBAND, ALU-CTS, ocs2007r2, lcs2005

`reg.x.server.y.address`

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

Null (default) - SIP server doesn’t accept registrations.

IP address or hostname - SIP server that accepts registrations. If not Null, all of the parameters in this list override the parameters specified in `voIpProt.server.*`.

`reg.x.server.y.expires`

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

3600 (default)

Positive integer, minimum 10

`reg.x.server.y.expires.lineSeize`

Requested line-seize subscription period.

30 - (default)

0 to 65535

`reg.x.server.y.expires.overlap`

The number of seconds before the expiration time returned by server x at which the phone should try to re-register. The phone tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value.

60 (default)

5 to 65535

`reg.x.server.y.failOver.failBack.mode`

duration (default) - The phone tries the primary server again after the time specified by `reg.x.server.y.failOver.failBack.timeout`.

newRequests - All new requests are forwarded first to the primary server regardless of the last used server.

DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.

registration - The phone tries the primary server again when the registration renewal signaling begins.

Note: This parameter overrides `voIpProt.server.x.failOver.failBack.mode`.

`reg.x.server.y.failOver.failBack.timeout`

3600 (default) - The time to wait (in seconds) before failback occurs.

0 - The phone does not fail back until a failover event occurs with the current server.

60 to 65535 - If set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.

`reg.x.server.y.failOver.failRegistrationOn`

1 (default) - The `reRegisterOn` parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over.

0 - The `reRegisterOn` parameter is disabled, existing registrations remain active.

`reg.x.server.y.failOver.onlySignalWithRegistered`

1 (default) - Set to this value and `reRegisterOn` and `failRegistrationOn` parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server.

0 - Set to this value and `reRegisterOn` and `failRegistrationOn` parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).

`reg.x.server.y.failOver.reRegisterOn`

0 (default) - The phone does not attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information.

1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server.

Note: This parameter overrides `voIpProt.server.x.failOver.reRegisterOn`.

`reg.x.server.y.port`

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

0 - The port used depends on `reg.x.server.y.transport`.

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

`reg.x.server.y.register`

1 (default) - Calls can't be routed to an outbound proxy without registration.

0 - Calls can be routed to an outbound proxy without registration.

See voIpProt.server.x.register for more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Poly Engineering Advisories and Technical Notifications.

`reg.x.server.y.registerRetry.baseTimeOut`

For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Used in conjunction with `reg.x.server.y.registerRetry.maxTimeOut` to determine how long to wait.

60 (default)

10 - 120 seconds

`reg.x.server.y.registerRetry.maxTimeout`

For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Use in conjunction with r `eg.x.server.y.registerRetry.baseTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.

180 - (default)

60 - 1800 seconds

`reg.x.server.y.retryMaxCount`

The number of retries attempted before moving to the next available server.

3 - (default)

0 to 20 - 3 is used when the value is set to 0.

`reg.x.server.y.retryTimeOut`

0 (default) - Use standard RFC 3261 signaling retry behavior.

0 to 65535 - The amount of time (in milliseconds) to wait between retries.

`reg.x.server.y.subscribe.expires`

The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period.

3600 seconds - (default)

10 - 2147483647 (seconds)

You can use this parameter in conjunction with `reg.x.server.y.subscribe.expires.overlap`.

`reg.x.server.y.subscribe.expires.overlap`

The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server.

60 seconds (default)

5 - 65535 seconds

`reg.x.server.y.transport`

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

DNSnaptr (default) - If `reg.x.server.y.address` is a *<hostname>* and `reg.x.server.y.port` is 0 or Null, perform NAPTR then SRV lookups to try to discover the transport, ports, and servers, as per RFC 3263.

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

TCPpreferred

UDPOnly - Only UDP is used.

TLS - If TLS fails, transport fails. Leave port field empty (defaults to 5061) or set to 5061.

TCPOnly - Only TCP is used.

`reg.x.server.y.useOutboundProxy`

1 (default) - Enables to use the outbound proxy specified in `reg.x.outboundProxy.address` for server x.

0 - Disable to use the outbound proxy specified in `reg.x.outboundProxy.address` for server x.

`divert.x.sharedDisabled`

1 (default) - Disables call diversion features on shared lines.

0 - Enables call diversion features on shared lines.

Change causes system to restart or reboot.

`dns.cache.A.x.`

Specify the DNS A address, hostname, and cache time interval.

`dns.cache.A.x.address`

Null (default)

IP version 4 address

`dns.cache.A.x.name`

Null (default)

valid hostname

`dns.cache.A.x.ttl`

The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again.

300 (default)

300 to 536870912 (2^29), seconds

`dns.cache.NAPTR.x.`

Specify the DNS NAPTR parameters, including: name, order, preference, regexp, replacement, service, and ttl.

`dns.cache.NAPTR.x.flags`

The flags to control aspects of the rewriting and interpretation of the fields in the record. Characters are case-sensitive. At this time, only 'S', 'A', 'U', and 'P' are defined as flags. See RFC 2915 for details of the permitted flags.

Null (default)

A single character from [A-Z, 0-9]

`dns.cache.NAPTR.x.name`

Null (default)

domain name string - The domain name to which this resource record refers.

`dns.cache.NAPTR.x.order`

0 (default)

0 to 65535 - An integer that specifies the order in which the NAPTR records must be processed to ensure the correct ordering of rules.

`dns.cache.NAPTR.x.preference`

0 (default)

0 to 65535 - A 16-bit unsigned integer that specifies the order in which NAPTR records with equal "order" values should be processed. Low numbers are processed before high numbers.

`dns.cache.NAPTR.x.regexp`

This parameter is currently unused. Applied to the original string held by the client. The substitution expression is applied in order to construct the next domain name to lookup. The grammar of the substitution expression is given in RFC 2915.

Null (default)string containing a substitution expression

`dns.cache.NAPTR.x.replacement`

The next name to query for NAPTR records depending on the value of the flags field. It must be a fully qualified domain-name.

Null (default)

domain name string with SRV prefix

dns.cache.NAPTR.x.service

Specifies the service(s) available down this rewrite path. For more information, see RFC 2915.

Null (default)

string

dns.cache.NAPTR.x.ttl

The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again.300 (default)

300 to 536870912 (2^29), seconds

dns.cache.A.networkOverride

0 (default) - Does not allow the static DNS A record entry to take priority over dynamic network DNS.

1 - Allows the static DNS cached A record entry to take priority over dynamic network DNS. Moreover, the DNS TTL value is ignored.

dns.cache.SRV.x.

Specify DNS SRV parameters, including: name, port, priority, target, ttl, and weight.

dns.cache.SRV.x.name

Null (default)

Domain name string with SRV prefix

dns.cache.SRV.x.port

The port on this target host of this service. For more information, see RFC 2782.

0 (default)

0 to 65535

dns.cache.SRV.x.priority

The priority of this target host. For more information, see RFC 2782.

0 (default)

0 to 65535

dns.cache.SRV.x.target

Null (default)

domain name string - The domain name of the target host. For more information, see RFC 2782.

`dns.cache.SRV.x.ttl`

The TTL describes the time period the phone uses the configured static cache record. If a dynamic network request receives no response, this timer begins on first access of the static record and once the timer expires, the next lookup for that record retries a dynamic network request before falling back on the static entry and it resets TTL timer again.

300 (default)

300 to 536870912 (2^29), seconds

`dns.cache.SRV.x.weight`

A server selection mechanism. For more information, see [RFC 2782](#).

0 (default)

0 to 65535

`tcpIpApp.dns.address.overrideDHCP`

Specifies how DNS addresses are set.

0 (default) - DNS address requested from the DHCP server.

1 - DNS primary and secondary address is set using the parameters `tcpIpApp.dns.server` and `tcpIpApp.dns.altServer`.

Change causes system to restart or reboot.

Note: If the DHCP server doesn't send the DNS server addresses to the phone, then the values set for the `device.dns.serverAddress` and `device.dns.altSrvAddress` parameters are used. Alternatively, the phone uses the DNS server addresses set using the `tcpIpApp*` parameters, which override `device.dns.*` parameters.

`tcpIpApp.dns.domain.overrideDHCP`

Specifies how the domain name is retrieved or set.

0 (default) - Domain name retrieved from the DHCP server, if one is available.

1 - DNS domain name is set using the parameter `tcpIpApp.dns.domain` parameter.

Change causes system to restart or reboot.

Note: If the DHCP server doesn't send the DNS domain to the phone, then the value set for `device.dns.domain` is used. Alternatively, the phone uses the DNS domain set using the `tcpIpApp*` parameter, which overrides `device.dns.*` parameter.

`dns.cache.dynamicRestore.enable`

1 - Allows the phone to restore the expired cache entries to a specified TTL when the DNS server isn't reachable.

0 (default) - Doesn't allow the phone to restore the expired cache entries to a specified TTL when the DNS server isn't reachable.

`dns.queryRetryCount`

Defines the number of retries the phone attempts before it restores the cache using the `dns.queryRetryCount` parameter.

0 to 48 - The number of retries that the phone attempts before the cache is restored.

0 - Disable.

4 (default)

Note: Requires `dns.cache.dynamicRestore.enable` to be enabled.

dns.cache.dynamicRestore.ttl

Specify a TTL value to restore the expired cache entries when the DNS server isn't reachable.

120 (default)

90 to 600 seconds

reg.x.secureTransportRequiresSrtp

0 (default) - Doesn't allow the phone to dynamically overwrite the configured values of `reg.x.srtp.offer` parameter and `reg.x.srtp.require` parameter based on the NAPTR response for per line registration.

1 - Allows the phone to dynamically overwrite the configured values of `reg.x.srtp.offer` parameter and `reg.x.srtp.require` parameter based on the NAPTR response for per line registration to enable SRTP only.

voIpProt.SIP.naptrAllowDuplicateTransport.enable

0 (Default) - The system ignores NAPTR records with duplicate protocols.

1 - The system considers all NAPTR records, regardless of transport, up to a maximum of 16 records.

STUN Parameters

This section lists parameters that configure Simple Traversal of UDP though NAT (STUN).

feature.nat.stun.enabled

0 (default) - Disable STUN.

1 - Enable STUN. SIP responses are sent to the source IP address and source port where the request originated. If you also enable the `voIpProt.SIP.rport` parameter, then the phone adds the received IP address and port in the VIA header while generating the response.

Change causes system to restart or reboot.

nat.stun.server

Enter a STUN server IP address or domain name.

Null (default)

Change causes system to restart or reboot.

nat.stun.port

Set the STUN server port number.

3478 (default)

1 to 65535

Change causes system to restart or reboot.

reg.x.nat.traversal.mode

Enable or disable NAT traversal mode with STUN for signaling and media on the basis of the phone-level STUN feature.

Auto - Apply NAT configuration to both media and signaling per registration.

Disabled (default) - The phone doesn't use STUN for NAT traversal for this registration.

For example, if `feature.nat.stun.enabled="1"` and `reg.x.nat.traversal.mode="Auto"`, the STUN feature is enabled for signaling and media for the registered line.

RTP Ports Parameters

Use the parameters in the following list to configure RTP packets and ports.

tcpIpApp.port.rtp.feccPortRange.enable

0 (default) – Use the Open SIP far-end camera control media port range.

1 - Use the far-end camera control port range configuration for Open SIP-registered lines.

tcpIpApp.port.rtp.feccPortRangeEnd

Specify the far-end camera control port range end port for Open SIP registrations.

2419 (default)

1024 - 65486

tcpIpApp.port.rtp.feccPortRangeStart

Specify the far-end camera control port range start port for Open SIP registrations.

2372 (default)

1024 – 65486

tcpIpApp.port.rtp.filterByIp1

IP addresses can be negotiated through the SDP or H.323 protocols.

1 (Default) - Phone rejects RTP packets that arrive from non-negotiated IP addresses.

Change causes system to restart or reboot.

tcpIpApp.port.rtp.filterByPort1

Ports can be negotiated through the SDP protocol.

0 (Default)

1 - Phone rejects RTP packets arriving from (sent from) a non-negotiated port.

Change causes system to restart or reboot.

tcpIpApp.port.rtp.forceSend1

Send all RTP packets to, and expect all RTP packets to arrive on, this port. Range is 0 to 65535.

0 (Default) - RTP traffic is not forced to one port.

Both `tcpIpApp.port.rtp.filterByIp` and `tcpIpApp.port.rtp.filterByPort` must be set to 1.

Change causes system to restart or reboot.

tcpIpApp.port.rtp.mediaPortRangeEnd

Determines the maximum supported end range of audio ports. Range is 1024 to 65485.

2269 (Default)

Change causes system to restart or reboot.

tcpIpApp.port.rtp.mediaPortRangeStart1

Set the starting port for RTP port range packets. Use an even integer ranging from 1024 to 65440.

2222 (Default)

Each call increments the port number +2 to a maximum of 24 calls after the value resets to the starting point. Because port 5060 is used for SIP signaling, ensure that port 5060 is not within this range when you set this parameter. A call that attempts to use port 5060 has no audio.

Change causes system to restart or reboot.

tcpIpApp.port.rtp.videoPortRange.enable

Specifies the range of video ports.

0 - Video ports are chosen within the range specified by `tcpIpApp.port.rtp.mediaPortRangeStart` and `tcpIpApp.port.rtp.mediaPortRangeEnd`.

1 - Video ports are chosen from the range specified by `tcpIpApp.port.rtp.videoPortRangeStart` and `tcpIpApp.port.rtp.videoPortRangeEnd`.

Generic = 0 (Default)

tcpIpApp.port.rtp.videoPortRangeEnd

Determines the maximum supported end range of video ports. Range is 1024 to 65535.

2319 (Default)

Change causes system to restart or reboot.

tcpIpApp.port.rtp.videoPortRangeStart

Determines the start range for video ports. Range is 1024 to 65486.

2272 (Default)

Used only if value of `tcpIpApp.port.rtp.videoPortRange.enable` is 1.

Change causes system to restart or reboot.

Two-Way Active Measurement Protocol Configuration Parameters

The following list includes the new or modified parameters for the two-way active measurement protocol feature.

feature.twamp.enabled

0 (default) - Disable TWAMP protocol support.

1 - Enable TWAMP protocol support.

twamp.port.udp.PortRangeEnd

Set the TWAMP UDP session max port range value.

60000 (default)

1024 - 65486

`twamp.port.udp.PortRangeStart`

Set the TWAMP UDP session start port range value.

40000 (default)

1024 - 65485

`twamp.udp.maxSession`

Set the maximum UDP session supported by TWAMP.

1 (default)

1 - 10

Wi-Fi Parameters

Poly Trio 8800 Poly Trio C60 systems are shipped with a security-restrictive worldwide safe Wi-Fi country code setting.

Use the following parameters to configure wireless network settings for your organization, which is dependent on the security mode for your organization and whether or not you enable DHCP. Poly Trio systems supports the following Wi-Fi security modes:

- WEP
- WPA PSK
- WPA2 PSK
- WPA2 Enterprise

Note: The Poly Trio 8500 solution does not support Wi-Fi connectivity.

`device.wifi.country`

Enter the two-letter code for the country where you are operating the Poly Trio system with Wi-Fi enabled.

NULL (default)

Two-letter country code

`device.wifi.dhcpBootServer`

0 (default)

1

2

V4

V6

Static

`device.wifi.dhcpEnabled`

Enable or disable DHCP for Wi-Fi.

0 (default) - Disable

1 - Enable

`device.wifi.enabled`

Enable or disable Wi-Fi.

0 (default) - Disable

1 - Enable

`device.wifi.ipAddress`

Enter the IP address of the wireless device if you are not using DHCP.

0.0.0.0 (default)

String

`device.wifi.ipGateway`

Enter the IP gateway address for the wireless interface if not using DHCP.

0.0.0.0 (default)

String

`device.wifi.psk.key`

Enter the hexadecimal key or ASCII passphrase.

0xFF (default)

String

`device.wifi.securityMode`

Specify the wireless security mode.

NULL (default)

None

WEP

WPA-PSK

WPA2-PSK

WPA2-Enterprise

`device.wifi.ssid`

Set the Service Set Identifier (SSID) of the wireless network.

SSID1 (default)

SSID

`device.wifi.subnetMask`

Set the network mask address of the wireless device if not using DHCP.

255.0.0.0 (default)

String

`device.wifi.wep.key`

Set the length of the hexadecimal WEP key.

0 = 40-bits (default)

1 = 104-bits

device.wifi.wpa2Ent.method

Set the Extensible Authentication Protocol (EAP) to use for 802.1X authentication.

NULL (default)

EAP-PEAPv0/MSCHAPv2

EAP-FAST

EAP-TLS

EAP-PEAPv0-GTC

EAP-TTLS-MSCHAPv2

EAP-TTLS-GTC

EAP-PEAPv0-NONE

EAP-TTLS-NONE

EAP-PWD

device.wifi.wpa2Ent.password

The WPA2-Enterprise password.

device.wifi.wpa2Ent.user

The WPA2-Enterprise user name.

Wi-Fi Settings in Basic Menu Parameter

Use the parameter below to allow access to Wi-Fi settings in the Basic menu on Poly Trio 8800 systems.

feature.basicWifiMenu.enabled

Enable to allow access to Wi-fi Menu in Basic settings.

0 (default) - Disabled

1 - Enabled

Related Links

Wi-Fi Settings in Basic Menu Parameter on page 146

Web Proxy Configuration Parameters

The following parameters configure web proxy settings.

feature.wpad.enabled

Set to enable web proxy.
0 (default) - Disables web proxy.

1 - Enables web proxy. Default for the Skype Base Profile.

Change causes system to restart or reboot.

feature.wpad.curl

Enter the PAC file location.

Change causes system to restart or reboot.

`feature.wpad.proxy`

Configure the web proxy server address. If you configure this parameter with a proxy address, the phones don't discover DHCP or DNS-A or fetch the PAC file even if you configure a PAC file location using `feature.wpad.curl`.

0 to 255 characters

Change causes system to restart or reboot.

`feature.wpad.proxy.username`

Enter the user name to authenticate with the proxy server.

0 to 255 characters

Change causes system to restart or reboot.

`feature.wpad.proxy.password`

Enter the password to authenticate with the proxy server.

The credentials you can use depend on how authentication is enabled on the proxy server. You can use administrator or user credentials. If Skype for Business Active Directory is integrated with the proxy server, you don't need to configure user name or password credentials.

0 to 255 characters

Change causes system to restart or reboot.

VoIP Server Parameters

The list below describes VoIP server configuration parameters.

`voIpProt.server.dhcp.available`

0 (default) - Do not check with the DHCP server for the SIP server IP address.

1 - Check with the server for the IP address.

Change causes system to restart or reboot.

`voIpProt.server.dhcp.option`

The option to request from the DHCP server if `voIpProt.server.dhcp.available` = 1.

128 (default) to 254

If `reg.x.server.y.address` is non-Null, it takes precedence even if the DHCP server is available.

Change causes system to restart or reboot.

`voIpProt.server.dhcp.type`

Type to request from the DHCP server if `voIpProt.server.dhcp.available` is set to 1.

0 (default) - Request IP address

1 - Request string

Change causes system to restart or reboot.

`voIpProt.OBP.dhcpv4.type`

Define the type of Outbound Proxy address.

0 (default) - IP address

1 - String

Change causes system to restart or reboot.

`voIpProt.OBP.dhcpv4.option`

The phone requests for DHCP option 120 and applies the outbound proxy obtained in DHCP to

120 (default)

Change causes system to restart or reboot.

`voIpProt.OBP.dhcpv6.option`

Define the type of Outbound Proxy address from DHCPv6.

21 (default) - list of domain name

22 - list of IP address

Change causes system to restart or reboot.

Pairing Parameters

This section provides information about the paring parameters for Poly Trio.

Daisy-Chaining Parameters

Use the following parameters to configure daisy-chaining options for Poly Trio systems.

`up.daisyChain.device.style`

Choose how to visually indicate which Poly Trio system is set to **Device** in a daisy-chaining scenario.

LineAtTopOfScreen (default) - A line displays at the top of the Poly Trio screen.

GlobalMenuIcon - An icon displays on the Poly Trio system Home screen.

Change causes system to restart or reboot.

`mr.pair.maxDevices`

Set the maximum number of paired devices to **2** for daisy-chaining Trio systems.

5 (default)

0 - 15

Poly Trio Visual+ Pairing Parameters

To pair using configuration files, enter the MAC address of your Poly Trio Visual+ device as the value for the mr.pair.uid.1 parameter.

The MAC address can be in either of the following formats:

- `00e0d::B09128D`
- `00E0DB09128D`

Use the following parameters to configure this feature and additional feature options:

`mr.PairButton.notification`

1 (default) - The Poly Trio system displays notifications of devices available to pair with after you press the **Pair** button on the Poly Trio Visual+.

0 - The Poly Trio system doesn't display pairing notifications.

`mr.audio.srtp.require`

1 (default) - SRTP is used to encrypt and authenticate modular room audio signals sent between Poly Trio and Poly Trio Visual+.

0

`mr.pair.uid.1`

Enter the MAC address of the Poly Trio Visual+ you want to pair with.

Null (default)

String (maximum of 64 characters)

`mr.pair.tls.enabled`

1 (default) - Enable TLS to encrypt communication between the Poly Trio and Poly Trio Visual+ systems.

0 - Disable TLS for communication between Poly Trio systems and Poly Trio Visual+ systems.

Change causes system to restart or reboot.

`mr.video.camera.focus.auto`

NULL (default)

0 - Disable the camera's automatic focus.

1 - Enable the camera's automatic focus.

Change causes system to restart or reboot.

`mr.video.camera.focus.range`

Specify the distance to the camera's optimally focused target.

NULL (default)

0

0 to 255

`mr.video.iFrame.minPeriod`

Choose the minimum time in seconds between transmitted video I-Frames or transmitted I-Frame requests.

2 (default)

1 to 60

`smartPairing.mode`

Enables users with RealPresence Desktop on a laptop or RealPresence Mobile on a tablet to pair with the Poly Trio system using SmartPairing.

Disabled (default)

Manual

`smartPairing.volume`

The relative volume to use for the SmartPairing ultrasonic beacon.

6 (default)

0 to 10

Poly Trio VisualPro Pairing Parameters

You can pair your peripheral system with Poly Trio using the mr.pair.uid.1 parameter. Once paired, you can configure some Trio VisualPro system settings (including software updates) using the following parameters.

`mr.pair.uid.1`

Enter the MAC address of the peripheral you want to pair with.

Null (default)

String (maximum of 64 characters)

`mr.pair.tls.enabled`

1 (default) - Enable TLS to encrypt communication between the Poly Trio and peripheral systems.

0 - Disable TLS for communication between Poly Trio systems and peripheral systems.

Change causes system to restart or reboot.

`mr.deviceMgmt.vc2.param.softwareUpdateUri`

Identifies the URI where the paired peripheral gets its software updates.

String

`mr.deviceMgmt.vc2.param.softwareUpdateProxyServer`

Identifies the proxy server where the paired peripheral gets its software updates.

String

`mr.deviceMgmt.vc2.param.softwareUpdateMaintenceWindowEnabled`

1 (default) - Enables the maintenance window for updating the paired peripheral's software.

0 - Disables the maintenance window for updating the paired peripheral's software.

`mr.deviceMgmt.vc2.param.softwareUpdateMaintenceWindowStart`

Specifies when the paired peripheral's maintenance window begins in 24-hour clock format (for example, `10:30` or `15:00`).

String

`mr.deviceMgmt.vc2.param.softwareUpdateMaintenceWindowDuration`

Specifies how long the paired peripheral's maintenance window lasts.

Range is 1-6 hours; default is 3.

`mr.deviceMgmt.vc2.param.displayName`

Sets the paired peripheral's system name.

String

`mr.deviceMgmt.vc2.param.hostName`

Sets the paired peripheral's host name.

String

USB Device Pass-through Parameters

Use the following parameters to configure the Poly Trio system to work with a Windows 10 computer and additional feature options.

`feature.usb.device.audio`

Set to allow the microphones and speakers on Poly Trio systems to be used as separate USB devices when connected to a computer.
1 (default) – Enables the ability to use Poly Trio as a USB speakerphone.

0 – Disables the ability to use Trio as a USB speakerphone.

Requires a restart.

`mr.devicePassThrough.usb.enabled`

Set to allow the connected computer to use the camera connected to a paired Poly Trio Visual+ or VisualPro system as a USB camera device when the Poly Trio system is connected to a computer with the Poly Trio Pass-through application installed.
1 (default) - Enables the USB Device Pass-through feature on the Poly Trio system.

0 - Disables the USB Device Pass-through feature.

`mr.devicePassThrough.camera.resolution`

Set to determine the camera resolution for connected cameras when USB Device Pass-through is enabled.
Auto (default)

1080p

720p

540p

360p

`mr.pair.maxDevices`

Set the maximum number of paired devices.
5 (default)

0 - 15

Phone Display Parameters

This section provides information about the phone display parameters for Poly Trio.

Administrator Menu Parameters

Use the following parameters to enable the **Administrator** or **Advanced** menu.

`device.auth.localAdvancedPassword.set`

Set a password for the **Advanced** menu.

0 (default) - You cannot set a password for the **Advanced** menu.

1 - You can set a password for the **Administrator** menu.

`device.auth.localAdvancedPassword`

Enter a password for the **Administrator** menu.

Null (default)

String (0 to 64 characters)

`feature.advancedUser.enabled`

0 (default) - The password-protected **Advanced** menu displays.

1 - Renames the **Advanced** menu item to **Admin** and adds a menu item **Advanced** that contains a subset of administrator features.

`feature.advancedUser.web.enabled`

Display the **Advanced** menu in the system web interface.

0 (default) – The system web interface provides login options for **Admin** or **User** only.

1 - Enable the **Advanced** user login option on the system web interface.

`ui.menu.advancedUser.networkConfiguration.`

Set whether to display the **Network** option under **Settings** for advanced users.

1 – (default) Displays the **Network** option.

0 – The **Network** option doesn't display.

`ui.menu.advancedUser.networkConfiguration.tls`

Set whether to display the **TLS** option under **Settings > Network** for advanced users.

1 – (default) Displays the **TLS** option.

0 – The **TLS** option doesn't display.

This parameter requires **ui.menu.advancedUser.networkConfiguration** to be set to *1*.

Capture Current Phone Screen Parameters

User the following parameters to get a screen capture of the current screen on your phone.

`up.screenCapture.enabled`

0 (Default) - The Screen Capture menu is hidden on the phone.

1 - The Screen Capture menu displays on the phone.

When the phone reboots, screen captures are disabled from the Screen Capture menu on the phone.

Change causes system to restart or reboot.

`up.screenCapture.allowed`

0 (Default) - The Screen Capture feature is disabled.

1 - The Screen Capture feature is enabled.

Custom Icon Parameters

Use the following parameters to configure custom icons for favorite contacts and line registrations.

`icons.x`

Specify the icon filename or URL location associated with the registered line (x), where x equals 1-24. The icons display on the phone for lines configured using parameter `reg.y.icon` or favorite contacts set for <up></up> in the `Mac-directory.xml` file.

Null (default)

icon file name or URL location

For example `icons.1="filename1"` or `icons.x="ftp://icons:icons@10.233.234.18/icon1.png"`

Change causes system to restart or reboot.

`reg.y.icon`

Assign an icon specified in `icons.x` to this registered line (y), where y equals 1-24.

Null

iconX, where x is 1-24

For example, if `icons.1="filename1"` then `reg.1.icon="icon1"`.

Change causes system to restart or reboot.

Custom Call Control Options Parameters

Use the following parameters to customize the call control menu.

`up.callStateView.controlRelegation.transfer`

0 (Default) – The **Transfer** option is available in the call control menu.

1 – The **Transfer** option is not available in the call control menu.

Change causes system to restart or reboot.

`up.callStateView.controlRelegation.mute`

0 (Default) – The **Mute** option is available in the call control menu.

1 – The **Mute** option is not available in the call control menu.

Change causes system to restart or reboot.

Default In-Call Screen Parameters

Use the following parameters to configure the default screens while the Poly Trio system is in a call.

`up.callStateView`

Set the default screen while in a call.

Drawer (default) – Displays the in-call controls.

Dialpad – Displays the dial pad.

Note that `reg.X.callStateView` can override this setting.

`up.callStateView.conference`

Set the default screen while in a conference call.

Roster (default) – Displays the roster view.

Drawer – Displays the in-call controls.

Note that `reg.X.callStateView.conference` can override this setting.

`reg.X.callStateView`

Set the default screen while in a call with registered line x. During the call, this parameter overrides the setting for `up.callStateView` if it's set to anything other than `Default`.

Drawer – Displays the in-call controls.

Dialpad – Displays the dial pad.

Default (default) – Uses the `up.callStateView` setting.

`reg.X.callStateView.conference`

Set the default screen while in a conference call with registered line x. During the call, this parameter overrides the setting for `up.callStateView.conference` if it's set to anything other than `Default`.

Poly Trio systems can bridge multiple ecosystems for conference calls; in these cases, the lowest involved line determines which screen displays.

Roster – Displays the roster view.

Dialpad – Displays the dial pad.

Default (default) – Uses the `up.callStateView.conference` setting.

Display Name Parameters

Set the phone name using one or more of following parameters.

`content.airplayServer.name`

Specify a system name for the local content sink for AirPlay-certified devices. If left blank, the previously configured or default system name is used.

NULL (default)

UTF-8 encoded string

`content.wirelessDisplay.sink.name`

Specify a system name for the local content sink for Android or Windows devices. If left blank the previously configured or default system name is used

NULL (default)

UTF-8 encoded string

`bluetooth.device.name`

Enter the name of the system that broadcasts over Bluetooth to other devices.

NULL (default)

UTF-8 encoded string

`reg.x.address`

The user part (for example, 1002) or the user and the host part (for example, `1002@poly.com`) of the registration SIP URI or the H.323 ID/extension for registration x.

Null (default)

string address

`reg.x.displayname`

The display name used in SIP signaling and/or the H.323 alias used as the default caller ID for registration x.

Null (default)

UTF-8 encoded string

`reg.x.label`

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

The maximum number of characters for this parameter value is `256`; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters you use. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The rules for parameter `up.cfgLabelElide` determine how the label is truncated.

Null (default)

UTF-8 encoded string

`system.name`

The system name that displays at the top left corner of the monitor, and at the top of the Global menu of the phone.

Enter a string, maximum 96 characters.

Hide MAC Address Parameters

The following list includes parameters that configure the display of MAC address.

`device.mac.hide.set`

Enable or disable the `device.mac.hide` parameter to control the display of MAC address information of phones to users.

Null (default)

0 - Disabled

1 - Enabled

`device.mac.hide`

0 (default) - MAC address displays.

1 - MAC address is hidden.

Multilingual Parameters

The multilingual parameters included in the following list are based on string dictionary files downloaded from the provisioning server.

These files are encoded in XML format and include space for user-defined languages.

`lcl.ml.lang.clock.x.24HourClock`

1 (default) - Displays the time in 24-hour clock mode.

0 - Does not display the time in 24-hour clock mode.

Note: Overrides the `lcl.datetime.time.24HourClock` parameter.

`lcl.ml.lang.clock.x.dateTop`

1 (default) - Displays date above time.

0 - Displays date below time.

Note: Overrides the `lcl.datetime.date.dateTop` parameter.

`lcl.ml.lang.clock.x.format`

"D,dM" (default)

String

The field may contain 0, 1 or 2 commas which can occur only between characters and only one at a time.

For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday.

Note: Overrides the `lcl.datetime.date.format` parameter to display the day and date.

`lcl.ml.lang.clock.x.longFormat`

1 (default) - Displays the day and month in long format (Friday/November).

0 - Displays the day and month in abbreviated format (Fri/Nov).

Note: Overrides the `lcl.datetime.date.longFormat` parameter.

`lcl.ml.lang.japanese.font.enabled`

Enable or disable the use of Japanese kana format.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`lcl.ml.lang.list`

Displays the list of languages supported on the phone.

All (default)

String

Change causes system to restart or reboot.

The basic character support includes the Unicode character ranges listed in the next table.

Unicode Ranges for Basic Character Support

Name	Range
C0 Controls and Basic Latin	U+0000 - U+007F
C1 Controls and Latin-1 Supplement	U+0080 - U+00FF
Cyrillic (partial)	U+0400 - U+045F

Number and Label Parameters

You can configure display of the phone number or label on the Home screen using centralized provisioning parameters.

`homeScreen.placeACall.enable`

0 - Does not display the label on the home screen.

`homeScreen.customLabel`

Specify the label to display on the phone's Home screen when `homeScreen.labelType="Custom"`. The label can be 0 to 255 characters.

Null (default)

`homeScreen.labelLocation`

Specify where the label displays on the screen.

StatusBar (default) - The phone displays the custom label in the status bar at the top of the screen.

BelowDate - The phone displays the custom label on the Home screen only, just below the time and date.

`homeScreen.labelType`

Specify the type of label to display on the phone's Home screen.

PhoneNumber (default)

- When the phone is set to use Lync Base Profile, the phone number is derived from the Skype for Business server.

Custom - Enter an alphanumeric string between 0 and 255 characters into the `homeScreen.customLabel` parameter.

PrimaryPhoneNumber – The status bar displays only the first phone number rather than all of the phone numbers.

None - Don't display a label.

`reg.1.useteluriAsLineLabel`

1 - If `reg.x.label`="Null" the tel URI/phone number/address displays as the label of the line key.

0 - If `reg.x.label`="Null" the value for `reg.x.displayName`, if available, displays as the label. If `reg.x.displayName` is unavailable, the user part of `reg.x.address` is used.

`up.formatPhoneNumbers`

1 (default) - Enables automatic number formatting.

0 - Disables automatic number formatting and numbers display separated by "-".

Olson Time Zone Parameters

Use the following parameters to configure an Olson time zone.

`tcpIpApp.sntp.olsonTimezoneID`

Enter an Olson time zone ID. If set to an invalid or unrecognized value, the time zone is be set to GMT with daylight saving disabled.

Null (default)

When set, this parameter overrides existing GMT offset and DST rules.

Phone Language Parameters

You can select the language that displays on the phone using the parameters in the following list.

`device.spProfile`

Set the default language that displays on the phone.

NULL (default) - The default language is an empty string (lcl.ml.lang=""), which is English.

DT - The default language is German (lcl.ml.lang="DTGerman_Germany").

`lcl.ml.lang`

Null (default) - Sets the phone language to US English.

String - Sets the phone language specified in the `lcl.ml.lang.menu.x.label` parameter.

`lcl.ml.lang.menu.x`

Specifies the dictionary files for the supported languages on the phone. Dictionary files must be sequential. The dictionary file cannot have capital letters, and the strings must exactly match a folder name of a dictionary file.

Null (default)

String

`lcl.ml.lang.menu.x.label`

Specifies the phone language menu label. The labels must be sequential.

Null (default)

String

Poly Trio Visual+ and Trio VisualPro Display Parameters

Use the following parameters to hide or display icons and features on monitors connected with Poly Trio Visual+ or Trio VisualPro systems when paired with a Poly Trio system.

`feature.exchangeVoiceMail.menuLocation`

Default (default) - Show the Voicemail menu in the global menu only when unread voicemails are available. After the voicemail is accessed, the Voicemail option no longer displays in the global menu and is accessible in the phone menu.

Everywhere - Always show the Voicemail menu in the global menu and phone menu.

MenusOnly - Show the Voicemail menu only in the phone Features menu.

`mr.bg.dimCustomImages`

Set to automatically dim a custom background image when the system is idle to improve the readability of text on a monitor connected to a Visual+ accessory.

1 (default) - The background image is dimmed automatically.

0 - The background image is not dimmed.

`mr.bg.selection`

Sets a background image for the connected monitor(s).

Poly (default)

Auto - Automatically cycles through HallstatterSeeLake, BavarianAlps, and ForgetMeNotPond. The background image changes each time a video call ends.

BlueGradient

BavarianAlps

ForgetMeNotPond

HallstatterSeeLake

Custom - Use a custom background specified by `mr.bg.url`.

A custom background image must be a JPEG with 1920x1080 resolution and a maximum size of 2.9 MB for it to display correctly on the paired Trio VisualPro or RealPresence Group Series system monitor(s). (PNG images, which typically are supported on Poly Trio, are not supported in this setup.)

`mr.bg.showPlcmLogo`

1 (default) - The Poly logo shows on the connected monitor(s).

0 - Hides the Poly logo.

`mr.bg.showWelcomeInstructions`

All (default) - Display the content-sharing graphic and welcome message on the connected monitor(s).

TextOnly - Hide the content-sharing graphic.

None - Hide both the content-sharing graphic and welcome message.

`mr.bg.url`

Specify an HTTP URL location of a background image to use on the connected monitor(s).

The Poly Trio system supports PNG and JPEG images up to 2.9 MB.

A custom background image must be a JPEG with 1920x1080 resolution and a maximum size of 2.9 MB for it to display correctly on the paired Trio VisualPro or RealPresence Group Series system monitor(s). (PNG images, which typically are supported on Poly Trio, are not supported in this setup.)

This background image is used only if `mr.bg.selection= "Custom"`

Null (default)

String (maximum 256 characters)

`up.hideSystemIpAddress`

Specify where the IP address of the Poly Trio system and Poly Trio Visual+ system are hidden from view.

You can access the IP address from the phone Advanced menu if you set this parameter to 'Menus' or 'Everywhere'.

Poly Trio Home Screen Parameters

Use the following parameters to configure the phone's Home screen display.

`homeScreen.application.enable`

0 (default) - Enable display of the Applications icon on the phone Home screen.

1 - Enable display of the Applications icon on the phone Home screen.

`homeScreen.calendar.enable`

1 (default) - Enable display of the Calendar icon on the phone Home screen.

0 - Disable display of the Calendar icon on the phone Home screen.

`homeScreen.contacts.enable`

1 (default) - The Contacts icon displays on the Home screen.

0 - The Contacts icon does not display on the Home screen.

`homeScreen.diagnostics.enable`

0 (default) - A Diagnostics icon does not show on the Home screen.

1 - A Diagnostics icon shows on the Home screen to provide quick access to the Diagnostics menu.

`homeScreen.directories.enable`

1 (default) - Enable display of the Directories menu icon on the phone Home screen.

0 - Disable display of the Directories menu icon on the phone Home screen.

`homeScreen.doNotDisturb.enable`

0 (default) - Disable display of the DND icon on the phone Home screen.

1 - Enable display of the DND icon on the phone Home screen.

`homeScreen.forward.enable`

0 (default) - Disable display of the call forward icon on the phone Home screen.

1 - Enable display of the call forward icon on the phone Home screen.

`homeScreen.messages.enable`

1 (default) - Enable display of the Messages menu icon on the phone Home screen.

0 - Disable display of the Messages menu icon on the phone Home screen.

`homeScreen.newCall.enable`

1 (default) - Enable display of the New Call icon on the phone Home screen.

0 - Disable display of the New Call icon on the phone Home screen.

`homeScreen.present.enable`

Control whether the Content icon displays on the Polycom Trio system Home screen when Content Sharing is enabled and the system is paired with a supported video and content accessory.

1 (default)

0

`homeScreen.redial.enable`

0 (default) - Disable display of the Redial menu icon on the phone Home screen.

1 - Enable display of the Redial menu icon on the phone Home screen.

`homeScreen.settings.enable`

1 (default) - Enable display of the Settings menu icon on the phone Home screen.

0 - Disable display of the Settings menu icon on the phone Home screen.

`softkey.feature.redial`

1 (default) - Displays the **Redial** softkey on the Home screen of the Poly Trio 8300 system.

0 - The **Redial** softkey doesn't display on the Home screen.

Poly Trio System Number Formatting Parameters

Use the following parameter to enable or disable number formatting.

`up.formatPhoneNumbers`

1 (default) - Enable automatic number formatting.

0 - Disable automatic number formatting.

Poly Trio System Status Message Parameters

Use the following parameters to configure status messages on the Poly Trio system.

`up.status.message.flash.rate`

Specify the number of seconds to display a message before moving to the next message.

2 seconds (default)

1 - 8 seconds

`up.status.message.x`

<message line one>

<message line two>

<message line three>

<message line four>

<message line five>

Theme Parameter

The following parameter configures the phone's theme.

`up.uiTheme`

Specifies the colors and icons used in the phone user interface. By default, the theme changes based on the set base profile. Poly doesn't recommend you change the default themes.
Default (default) - The phone displays the default Poly theme. Available for Generic and Skype for Business base profiles.

SkypeForBusiness - The phone displays the Skype for Business theme. Available for Generic and Skype for Business base profiles.

MSTeams - The phone displays the Microsoft Teams theme. Available for the MSTeams base profile only.

SettingsMenu - The phone displays the Settings menu for Zoom Rooms. Available for the Zoom Rooms base profile only.

GenericUSB - The phone displays the Generic theme for USB optimized mode. Available for the Microsoft USB Optimized base profile only.

Time Zone Location Parameters

The following parameters configure time zone location.

Time Zone Location Parameter Values

Permitted Value	Time Zone Description
0	(GMT -12:00) Eniwetok,Kwajalein
1	(GMT -11:00) Midway Island
2	(GMT -10:00) Hawaii
3	(GMT -9:00) Alaska
4	(GMT -8:00) Pacific Time (US & Canada)
5	(GMT -8:00) Baja California
6	(GMT -7:00) Mountain Time (US & Canada)
7	(GMT -7:00) Chihuahua,La Paz
8	(GMT -7:00) Mazatlan
9	(GMT -7:00) Arizona
10	(GMT -6:00) Central Time (US & Canada)

Permitted Value	Time Zone Description
11	(GMT -6:00) Mexico City
12	(GMT -6:00) Saskatchewan
13	(GMT -6:00) Guadalajara
14	(GMT -6:00) Monterrey
15	(GMT -6:00) Central America
16	(GMT -5:00) Eastern Time (US & Canada)
17	(GMT -5:00) Indiana (East)
18	(GMT -5:00) Bogota,Lima
19	(GMT -5:00) Quito
20	(GMT -4:30) Caracas
21	(GMT -4:00) Atlantic Time (Canada)
22	(GMT -4:00) San Juan
23	(GMT -4:00) Manaus,La Paz
24	(GMT -4:00) Asuncion,Cuiaba
25	(GMT -4:00) Georgetown
26	(GMT -3:30) Newfoundland
27	(GMT -3:00) Brasilia
28	(GMT -3:00) Buenos Aires
29	(GMT -3:00) Greenland
30	(GMT -3:00) Cayenne,Fortaleza
31	(GMT -3:00) Montevideo
32	(GMT -3:00) Salvador
33	(GMT -3:00) Santiago
34	(GMT -2:00) Mid-Atlantic
35	(GMT -1:00) Azores
36	(GMT -1:00) Cape Verde Islands
37	(GMT 0:00) Western Europe Time
38	(GMT 0:00) London,Lisbon
39	(GMT 0:00) Casablanca
40	(GMT 0:00) Dublin
41	(GMT 0:00) Edinburgh
42	(GMT 0:00) Monrovia
43	(GMT 0:00) Reykjavik
44	(GMT +1:00) Belgrade
45	(GMT +1:00) Bratislava
46	(GMT +1:00) Budapest
47	(GMT +1:00) Ljubljana
48	(GMT +1:00) Prague
49	(GMT +1:00) Sarajevo,Skopje
50	(GMT +1:00) Warsaw,Zagreb

Permitted Value	Time Zone Description
51	GMT +1:00) Brussels
52	(GMT +1:00) Copenhagen
53	(GMT +1:00) Madrid,Paris
54	(GMT +1:00) Amsterdam,Berlin
55	(GMT +1:00) Bern,Rome
56	(GMT +1:00) Stockholm,Vienna
57	(GMT +1:00) West Central Africa
58	(GMT +1:00) Windhoek
59	(GMT +2:00) Bucharest,Cairo
60	(GMT +2:00) Amman,Beirut
61	(GMT +2:00) Helsinki,Kyiv
62	(GMT +2:00) Riga,Sofia
63	(GMT +2:00) Tallinn,Vilnius
64	(GMT +2:00) Athens
65	(GMT +2:00) Damascus
66	(GMT +2:00) E.Europe
67	(GMT +2:00) Harare,Pretoria
68	(GMT +2:00) Jerusalem
69	(GMT +2:00) Kaliningrad (RTZ 1)
70	(GMT +2:00) Tripoli
71	(GMT +3:00) Moscow
72	(GMT +3:00) St.Petersburg
73	(GMT +3:00) Volgograd (RTZ 2)
74	(GMT +3:00) Kuwait,Riyadh
75	(GMT +3:00) Nairobi
76	(GMT +3:00) Baghdad
77	(GMT +3:00) Minsk, Istanbul
78	(GMT +3:30) Tehran
79	(GMT +4:00) Abu Dhabi,Muscat
80	(GMT +4:00) Baku,Tbilisi
81	(GMT +4:00) Izhevsk,Samara (RTZ 3)
82	(GMT +4:00) Port Louis
83	(GMT +4:00) Yerevan
84	(GMT +4:30) Kabul
85	(GMT +5:00) Yekaterinburg (RTZ 4)
86	(GMT +5:00) Islamabad
87	(GMT +5:00) Karachi
88	(GMT +5:00) Tashkent
89	(GMT +5:30) Mumbai,Chennai
90	(GMT +5:30) Kolkata,New Delhi

Permitted Value	Time Zone Description
91	(GMT +5:30) Sri Jayawardenepura
92	(GMT +5:45) Kathmandu
93	(GMT +6:00) Astana,Dhaka
94	(GMT +6:00) Almaty
95	(GMT +6:00) Novosibirsk (RTZ 5)
96	(GMT +6:30) Yangon (Rangoon)
97	(GMT +7:00) Bangkok,Hanoi
98	(GMT +7:00) Jakarta
99	(GMT +7:00) Krasnoyarsk (RTZ 6)
100	(GMT +8:00) Beijing,Chongqing
101	(GMT +8:00) Hong Kong,Urumqi
102	(GMT +8:00) Kuala Lumpur
103	(GMT +8:00) Singapore
104	(GMT +8:00) Taipei,Perth
105	(GMT +8:00) Irkutsk (RTZ 7)
106	(GMT +8:00) Ulaanbaatar
107	(GMT +9:00) Tokyo,Seoul,Osaka
108	(GMT +9:00) Sapporo,Yakutsk (RTZ 8)
109	(GMT +9:30) Adelaide,Darwin
110	(GMT +10:00) Canberra
111	(GMT +10:00) Magadan (RTZ 9)
112	(GMT +10:00) Melbourne
113	(GMT +10:00) Sydney,Brisbane
114	(GMT +10:00) Hobart
115	(GMT +10:00) Vladivostok
116	(GMT +10:00) Guam,Port Moresby
117	(GMT +11:00) Solomon Islands
118	(GMT +11:00) New Caledonia
119	(GMT +11:00) Chokurdakh (RTZ 10)
120	(GMT +12:00) Fiji Islands
121	(GMT +12:00) Auckland,Anadyr
122	(GMT +12:00) Petropavlovsk-Kamchatsky (RTZ 11)
123	(GMT +12:00) Wellington
124	(GMT +12:00) Marshall Islands
125	(GMT +13:00) Nuku'alofa
126	(GMT +13:00) Samoa

Time and Date Display Parameters

Use the parameters in the following list to configure time and display options.

`up.localClockEnabled`

Specifies whether or not the date and time are shown on the idle display.

1 (Default) - Date and time are shown.

0 - Date and time are hidden.

`lcl.datetime.date.dateTop`

1 - Displays the date above time.

0 (default) - Displays the time above date.

`lcl.datetime.date.format`

The phone displays day and date. The field may contain 0, 1, or 2 commas which can occur only between characters and only one at a time.

For example: D,dM = Thursday, 3 July or Md,D = July 3, Thursday.

"D,dM" (default)

String

`lcl.datetime.date.longFormat`

1 (default) - Displays the day and month in long format (Friday/November).

0 - Displays the day and month in abbreviated format (Fri/Nov).

`lcl.datetime.time.24HourClock`

1 (default) - Displays the time in 24-hour clock mode.

0 - Displays the time in 12-hour clock mode.

`tcpIpApp.sntp.address`

Specifies the SNTP server address.

NULL (default)

Valid hostname or IP address.

`tcpIpApp.sntp.AQuery`

Specifies a query to return hostnames.

0 (default) - Queries to resolve the SNTP hostname are performed using DNS SRV.

1 - Query the hostname for a DNS A record.

`tcpIpApp.sntp.address.overrideDHCP`

0 (Default) - DHCP values for the SNTP server address are used.

1 - SNTP parameters override the DHCP values.

`tcpIpApp.sntp.daylightSavings.enable`

Enable or disable Daylight Savings Time rules to the displayed time.

1 (Default) - Enabled

0 - Disabled

`tcpIpApp.sntp.daylightSavings.fixedDayEnable`

0 (Default) - `Month` , `date` , and `dayOfWeek` are used in the DST calculation.

1 - Only `month` and `date` are used in the DST calculation.

`tcpIpApp.sntp.daylightSavings.start.date`

Start date for daylight savings time. Range is 1 to 31.

8 (Default) - Second occurrence in the month after DST starts.

0 - If `fixedDayEnable` is set to 0, this value specifies the occurrence of `dayOfWeek` when DST should start.

1 - If `fixedDayEnable` is set to 1, this value is the day of the month to start DST.

15 - Third occurrence.

22 - Fourth occurrence.

Example: If value is set to 15, DST starts on the third `dayOfWeek` of the month.

`tcpIpApp.sntp.daylightSavings.start.dayOfWeek`

Specifies the day of the week to start DST. This parameter is not used if `fixedDayEnable` is set to 1.

1 (Default) - Sunday

1-7 where the integer entered corresponds to a day of the week. For example, 1 = Sunday, 2 = Monday, and so on to 7 = Saturday.

`tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth`

0 (Default)

1 - DST starts on the last `dayOfWeek` of the month and the `start.date` is ignored.

Note: This parameter is not used if `fixedDayEnable` is set to 1.

`tcpIpApp.sntp.daylightSavings.start.month`

Specifies the month to start DST.

3 (Default) - March

1-12 where the integer entered corresponds to a month of the year. For example, 1 = January, 2 = February and so on to 12 = December.

`tcpIpApp.sntp.daylightSavings.start.time`

Specifies the time of day to start DST in 24-hour clock format. Range is 0 to 23.

2 (Default) - 2 a.m.

0 - 23 where the integer entered corresponds to the hour on in a 24 span. For example, 0 = 12 AM, 1 = 1 AM, and so on to 23 = 11 PM.

`tcpIpApp.sntp.daylightSavings.stop.date`

Specifies the stop date for daylight savings time. Range is 1 to 31.

1 (Default) - If `fixedDayEnable` is set to 1, the value of this parameter is the day of the month to stop DST. Set 1 for the first occurrence in the month.

0 - If `fixedDayEnable` is set to 0, this value specifies the `dayOfWeek` when DST should stop.

8 - Second occurrence.

15 - Third occurrence.

22 - Fourth occurrence.

Example: If set to 22, DST stops on the fourth `dayOfWeek` in the month.

`tcpIpApp.sntp.daylightSavings.stop.dayOfWeek`

Day of the week to stop DST.

1 (default) - Sunday

1-7 where the integer entered corresponds to a day of the week. For example, 1 = Sunday, 2 = Monday, and so on to 7 = Saturday.

Note: Parameter is not used if `fixedDayEnable` is set to 1.

`tcpIpApp.sntp.daylightSavings.stop.dayOfWeek.lastInMonth`

1 - DST stops on the last `dayOfWeek` of the month and the `stop.date` is ignored).

Parameter is not used if `fixedDayEnable` is set to 1.

`tcpIpApp.sntp.daylightSavings.stop.month`

Specifies the month to stop DST. Range is 1 to 12.

11 (Default) - November

1-12 where the integer entered corresponds to a month of the year. For example, 1 = January, 2 = February and so on to 12 = December.

`tcpIpApp.sntp.daylightSavings.stop.time`

Specifies the time of day to stop DST in 24-hour clock format. Range is 0 to 23.

2 (Default) - 2 a.m.

0 - 23 where the integer entered corresponds to the hour on in a 24 span. For example, 0 = 12 AM, 1 = 1 AM, and so on to 23 = 11 PM.

`tcpIpApp.sntp.gmtOffset`

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

0 (Default) - GMT

3600 seconds = 1 hour

-3600 seconds = -1 hour

Positive or negative integer

`tcpIpApp.sntp.gmtOffsetcityID`

You must disable `tcpIpApp.sntp.daylightSavings.enable` for the phone to display daylight savings time according to `gmtOffsetcityID`.

NULL (Default)

For descriptions of all values, refer to Time Zone Location Description.

0 to 127

`tcpIpApp.sntp.gmtOffset.overrideDHCP`

0 (Default) - The DHCP values for the GMT offset are used.

1 - The SNTP values for the GMT offset are used.

`tcpIpApp.sntp.resyncPeriod`

Specifies the period of time (in seconds) that passes before the phone resynchronizes with the SNTP server.

86400 (Default). 86400 seconds is 24 hours.

Positive integer

`tcpIpApp.sntp.retryDnsPeriod`

Sets a retry period for DNS queries. The DNS retry period is affected by other DNS queries made on the phone. If the phone makes a query for another service during the retry period, such as SIP registration, and receives no response, the Network Time Protocol (NTP) DNS query is omitted to limit the retry attempts to the unresponsive server. If no other DNS attempts are made by other services, the retry period is not affected. If the DNS server becomes responsive to another service, NTP immediately retries the DNS query.

86400 (Default). 86400 seconds is 24 hours.

60 - 2147483647 seconds

Date Formats

Use the following table to choose values for the `lcl`.

`datetime.date.format` and `lcl.datetime.date.longformat` parameters. The table shows values for Friday, August 19, 2011 as an example.

Date Formats

lcl.datetime.date.format	lcl.datetime.date.longformat	Date Displayed on Phone
dM,D	0	19 Aug, Fri
dM,D	1	19 August, Friday
Md,D	0	Aug 19, Fri
Md,D	1	August 19, Friday
D,dM	0	Fri, 19 Aug
D,dM	1	Friday, August 19
DD/MM/YY	n/a	19/08/11
DD/MM/YYYY	n/a	19/08/2011
MM/DD/YY	n/a	08/19/11
MM/DD/YYYY	n/a	08/19/2011
YY/MM/DD	n/a	11/08/19
YYYY/MM/DD	n/a	2011/08/11

Unique Line Labels for Registration Lines Parameters

When using this feature with the parameter `reg.x.label.y` where x=2 or higher, multiple line keys display for the registered line address.

`reg.x.line.y.label`

Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1`. If `reg.x.linekeys=1`, this parameter does not have any effect.

x = the registration index number starting from 1.

y = the line index from 1 to the value set by `reg.x.linekeys`. Specifying a string sets the label used for the line key registration on phones with multiple line keys.

If no parameter value is set for `reg.x.line.y.label`, the phone automatically numbers multiple lines by prepending "<y>_" where <y> is the line index from 1 to the value set by `reg.x.linekeys`.

`up.cfgLabelElide`

Controls the alignment of the line label. By default when the line label is an alphanumeric or alphabetic string, the label aligns right. When the line label is a numeric string, the label aligns left.

None (Default)

Right

Left

`up.cfgUniqueLineLabel`

Allow unique labels for a registration that is split across multiple line keys using `reg.X.linekeys`.

0 (Default) - Use the same label on all line keys.

1 - Display a unique label as defined by `reg.X.line.Y.label`.

If `reg.X.line.Y.label` is not configured, then a label of the form <integer>_ will be applied in front of the applied label automatically.

Power Management Parameters

This section provides information on the power management parameters for the Poly Trio.

Poly Trio System Power Management Parameters

You can use the parameters listed to manage the Poly Trio system's power usage.

`poe.pse.class`

Specify the LAN OUT PoE class.

0 (default)

0 - 3

`poe.pse.enabled`

1 (default) - The Poly Trio LAN OUT interface provides PoE power to a connected device.

0 - PoE power is not provided by the LAN OUT port.

`usb.charging.enabled`

0 (default) - You can't charge USB-connected devices from the USB charging port.

1 - Enable fast charging of devices connected by USB port up to 7.5 W power / 1.5 A current.

Power-Saving Parameters

Use the following parameters to configure power-saving features and feature options.

`powerSaving.Enable`

0 (default) - The paired device does not into power-saving mode.

1 - When the Poly Trio system enters power-saving mode, the paired device display switches to standby mode and powers up when the system exits power-saving mode.

`powerSaving.cecEnable`

Enable to synchronize paired devices and displays connected via HDMI with the Trio system when it enters power-saving mode.

0 (default) - The paired device display behavior is controlled only by the value set for `powerSaving.tvStandbyMode`.

1 - When the Poly Trio system enters power-saving mode, the paired device display switches to standby mode and powers up when the system exits power-saving mode.

`powerSaving.Enable`

1 (default) - Enable the LCD power-saving feature.

0 - Disable The LCD power-saving feature.

`powerSaving.idleTimeout.offHours`

The number of idle minutes during off hours after which the phone enters power saving.

1 (default)

1 - 10

`powerSaving.idleTimeout.officeHours`

The number of idle minutes during office hours after which the phone enters power saving.

30 (default)

1 - 600

`powerSaving.idleTimeout.userInputExtension`

The number of minutes after the phone is last used after which the phone enters power saving.

10 (default)

1 - 20

`powerSaving.officeHours.duration.x`

Append the day of the week for *x*. For example, `powerSaving.officeHours.duration.Monday`.

Set the duration of the office working hours by weekday.

Monday - Friday = 12 (default)

Saturday - Sunday = 0

0 - 24

`powerSaving.officeHours.startHour.x`

Specify the starting hour for the day's office working hours.

7 (default)

0 - 23

Set *x* to Monday, Tuesday, Wednesday, Thursday, Friday, Saturday, and Sunday (see `powerSaving.officeHours.duration` for an example).

`powerSaving.tvStandbyMode`

black (default) - The paired device displays a black screen after entering power-saving mode.

noSignal - Power-saving mode turns off the HDMI signal going to the paired device monitor(s).

Security Options Parameters

This section provides information about the security options parameters for Poly Trio.

802.1X Authentication Parameters

To set up an EAP method that requires a device or CA certificate, you need to configure TLS Platform Profile 1 or TLS Platform Profile 2 to use with 802.1X.

You can use the parameters in the following list to configure 802.1X Authentication.

For more information on EAP authentication protocol, see RFC 3748: Extensible Authentication Protocol.

`device.net.dot1x.enabled`

Enable or disable 802.1X authentication

0 - Disabled

1 - Enabled

Change causes system to restart or reboot.

`device.net.dot1x.identity`

Set the identity (user name) for 802.1X authentication

String

Change causes system to restart or reboot

`device.net.dot1x.method`

Specify the 802.1X EAP method

EAP-None - No authentication

EAP-TLS,

EAP-PEAPv0-MSCHAPv2,

EAP-PEAPv0-GTC,

EAP-TTLS-MSCHAPv2,

EAP-TTLS-GTC,

EAP-FAST,

EAP-MD5

`device.net.dot1x.password`

Set the password for 802.1X authentication. This parameter is required for all methods except EAP-TLS

String

Change causes system to restart or reboot.

`device.net.dot1x.eapFastInBandProv`

Enable EAP In-Band Provisioning for EAP-FAST

0 (default) - Disabled

1 - Unauthenticated, active only when the EAP method is EAP-FAST

`device.pacfile.data`

Specify a PAC file for EAP-FAST (optional)

Null (default)

0-2048 - String length

`device.pacfile.password`

The optional password for the EAP-FAST PAC file.

Null (default)

0-255 - String length

Administrator and User Password Parameters

Use the following parameters to set the administrator and user password and configure password settings.

`sec.pwd.length.admin`

The minimum character length for administrator passwords changed using the phone. Use 0 to allow null passwords.

1 (default)

0 - 32

Change causes system to restart or reboot.

`sec.pwd.length.user`

The minimum character length for user passwords changed using the phone. Use 0 to allow null passwords.

2 (default)

0 - 32

Change causes system to restart or reboot.

`up.echoPasswordDigits`

1 (default) - The phone briefly displays password characters before masking them with an asterisk.

0 - The phone displays only asterisks for the password characters.

`device.auth.localAdminPassword`

Specify a local administrator password.

0 - 32 characters

You must use this parameter with: `device.auth.localAdminPassword.set="1"`

`device.auth.localAdminPassword.set`

0 (default) - Disables overwriting the local admin password when provisioning using a configuration file.

1 - Enables overwriting the local admin password when provisioning using a configuration file.

Basic Settings Menu Lock Parameter

Use the parameter below to lock the Basic settings menu.

`up.basicSettingsPasswordEnabled`

Specifies that a password is required or not required to access the **Basic Settings** menu.

0 (Default) - No password is required to access the **Basic Settings** menu.

1 - Password is required for access to the **Basic Settings** menu.

Configuration File Encryption Parameters

The following list provides the parameters you can use to encrypt your configuration files.

`device.sec.configEncryption.key`

Set the configuration encryption key used to encrypt configuration files.

string

Change causes system to restart or reboot.

`sec.encryption.upload.callLists`

0 (default) - The call list is uploaded without encryption.

1 - The call list is uploaded in encrypted form.

Change causes system to restart or reboot.

`sec.encryption.upload.config`

0 (default) - The file is uploaded without encryption and replaces the phone-specific configuration file on the provisioning server.

1 - The file is uploaded in encrypted form and replaces the existing phone-specific configuration file on the provisioning server.

`sec.encryption.upload.dir`

0 (default) - The contact directory is uploaded without encryption and replaces the phone-specific contact directory on the provisioning server.

1 - The contact directory is uploaded in encrypted form and replaces the existing phone-specific contact directory on the provisioning server.

Change causes system to restart or reboot.

`sec.encryption.upload.overrides`

0 (default) - The MAC address configuration file is uploaded without encryption and replaces the phone-specific MAC address configuration file on the provisioning server.

1 - The MAC address configuration file is uploaded in encrypted form and replaces the existing phone-specific MAC address configuration file on the provisioning server.

Disable Unused Ports and Features Parameters

Use the parameters in the following list to disable external ports or specific features.

`device.net.etherModePC`

0 (default) - Disable the PC port mode that sets the network speed over Ethernet.

1 - Enable the PC port mode that sets the network speed over Ethernet.

`device.auxPort.enable`

-1 - Disabled

0 - Auto (default)

1 - 10HD

2 - 10FD

3 - 100HD

4 - 100FD

5 - 1000FD

`httpd.enabled`

Base Profile = Generic

1 (default) - The web server is enabled.

0 - The web server is disabled.

Base Profile = Skype

0 (default) - The web server is disabled.

1 - The web server is enabled.

Change causes system to restart or reboot.

`ptt.pttMode.enable`

0 (default) - Disable push-to-talk mode.

1 - Enable push-to-talk mode.

`feature.callRecording.enabled`

0 (default) - Disable the phone USB port for local call recording.

1 - Enable the phone USB port for local call recording.

Change causes system to restart or reboot.

`up.handsfreeMode`

1(default) - Enable handsfree mode.

0 - disable handsfree mode.

`feature.forward.enable`

1(default) - Enable call forwarding.

0 - Disable call forwarding.

`feature.doNotDisturb.enable`

1(default) - Enable Do Not Disturb (DND).

0 - Disable Do Not Disturb (DND).

Change causes system to restart or reboot.

`homeScreen.doNotDisturb.enable`

1 (default) - Enables the display of the DND icon on the phone's Home screen.

0 - Disables the display of the DND icon on the phone's Home screen.

`call.autoAnswerMenu.enable`

1 (default) - Enables the phone's Autoanswer menu.

0 - Disables the phone's Autoanswer menu.

FIPS 140-2 Parameter

The following parameter enables or disables the FIPS 140-2 feature.

`device.sec.TLS.FIPS.enabled`

0 (default) - Disables the FIPS-compliant cryptography feature.

1 - Enables the FIPS-compliant cryptography feature.

Phone Lock Parameters

Use the parameters in the following list to enable the phone lock feature, set authorized numbers for users to call when a phone is locked, and set scenarios when the phone should be locked.

`phoneLock.Allow.AnswerOnLock`

1 (default) - Users can answer any incoming call without needing to unlock the phone.

0 - Users must unlock the phone before answering an incoming call.

`phoneLock.authorized.x.description`

The name or description of an authorized number.

Null (default)

String

Up to five (x=1 to 5) authorized contacts that a user can call while their phone is locked. Each contact needs a description to display on the screen, and a phone number or address value for the phone to dial.

`phoneLock.authorized.x.value`

The number or address for an authorized contact.

Null (default)

String

Up to five (x=1 to 5) authorized contacts that a user can call while their phone is locked. Each contact needs a description to display on the screen, and a phone number or address value for the phone to dial.

`phoneLock.browserEnabled`

0 (default) - The microbrowser or browser is not displayed while the phone is locked.

1 - The microbrowser or browser is displayed while the phone is locked.

`phoneLock.dndWhenLocked`

0 (default) - The phone can receive calls while it is locked

1 - The phone enters Do-Not-Disturb mode while it is locked

`phoneLock.enabled`

0 (default) - The phone lock feature is disabled

1 - The phone lock feature is enabled.

`phoneLock.idleTimeout`

The amount of time (in seconds) the phone can be idle before it automatically locks. If 0, automatic locking is disabled.

0 (default)

0 to 65535

`phoneLock.lockState`

0 (default) - The phone is unlocked.

1 - The phone is locked.

The phone stores and uploads the value each time it changes via the MAC-phone.cfg. You can set this parameter remotely using the Web Configuration Utility.

`phoneLock.powerUpUnlocked`

Overrides the `phoneLock.lockState` parameter.

0 (default) - The phone retains the value in `phoneLock.lockState` parameter.

1 - You can restart, reboot, or power cycle the phone to override the value for `phoneLock.lockState` in the MAC-phone.cfg and start the phone in an unlocked state.

You can then lock or unlock the phone locally. Poly recommends that you do not leave this parameter enabled

Secondary Port Link Status Report Parameters

You can use the parameters in the following list to configure options for the Secondary Port Link Status Report feature, including the required elapse or sleep time between two CDP UPs dispatching.

`sec.dot1x.eapollogoff.enabled`

0 (default) - The phone does not send an EAPOL Logoff message.

1 - The phone sends an EAPOL Logoff message.

Change causes system to restart or reboot.

`sec.dot1x.eapollogoff.lanlinkreset`

0 (default) - The phone does not reset the LAN port link.

1 - The phone resets the LAN port link.

Change causes system to restart or reboot.

`sec.hostmovedetect.cdp.enabled`

0 (default) - The phone does not send a CDP packet.

1 - The phone sends a CDP packet.

Change causes system to restart or reboot.

`sec.hostmovedetect.cdp.sleepTime`

Controls the frequency between two consecutive link-up state change reports.

1000 (default)

0 to 60000

If `sec.hostmovedetect.cdp.enabled` is set to 1, there is an x microsecond time interval between two consecutive link-up state change reports, which reduces the frequency of dispatching CDP packets.

Change causes system to restart or reboot.

Simple Certificate Enrollment Protocol Parameters

Use the following parameters to configure Simple Certificate Enrollment Protocol (SCEP).

`SCEP.CAFingerprint`

Configure the CA certificate fingerprint to confirm the authenticity of the CA response during enrollment.

null (default)

0 - 255 characters

`SCEP.certPoll.retryCount`

Specify the number of times to poll the SCEP server when the SCEP server returns a Certificate Enrollment Response Message with pkiStatus set to `pending`.

12 (default)

1 - 24

`SCEP.certPoll.retryInterval`

Specify the number of seconds to wait between poll attempts when the SCEP server returns a Certificate Enrollment Response Message with pkiStatus set to `pending`.

300 (default)

300 - 3600

`SCEP.certRenewalRetryInterval`

Specify the time interval to retry certificate renewal.

86400 seconds (default)

28800 - 259200 seconds

`SCEP.certRenewalThreshold`

Specify the percentage of the certificate validity interval to initiate a renewal.

80 (default)

50 - 100

`SCEP.challengePassword`

Specify the challenge password to send with the Certificate Signing Request (CSR) when requesting a certificate.

null (default)

0 - 255 characters

`SCEP.csr.commonName`

Specify the common name to use for CSR generation.

Note: If you use the default setting, the phone uses its own MAC address for the CN value in the generated CSR.

null (default)

0 - 64

`SCEP.csr.country`

Specify the country name to use for CSR generation.

null (default)

0 - 2

`SCEP.csr.email`

Specify the email address to use for CSR generation.

null (default)

0 - 64

`SCEP.csr.locality`

Specify the phone's locality (L) to use for CSR generation.

null (default)

0-64 characters

`SCEP.csr.organization`

Specify the organization name to use for CSR generation.

null (default)

0 - 64

`SCEP.csr.organizationUnit`

Specify the phone's organizational unit (OU) to use for CSR generation.

null (default)

0-64 characters

`SCEP.csr.state`

Specify the state name to use for CSR generation.

null (default)

0 - 128 characters

`SCEP.enable`

0 (default) - Disable the SCEP feature.

1 - Enable the SCEP feature.

`SCEP.enrollment.retryCount`

Specify the number of times to retry the enrollment process in case of enrollment failure.

12 (default)

1 - 24

`SCEP.enrollment.retryInterval`

Specify the time interval to retry the enrollment process.

300 seconds (default)

300 - 3600 seconds

`SCEP.http.password`

Specify the password that authenticates with the SCEP server.

null (default)

string, max 255 characters

`SCEP.http.username`

Specify the user name that authenticates with the SCEP server.

null (default)

string, max 255 characters

`SCEP.url`

Specify the URL address of the SCEP server accepting requests to obtain a certificate.

null (default)

0 - 255 characters

`SCEP.verifyWithScepCaCert`

Connect to the SCEP server with TLS verified with a CA cert provided by the server.

1 (default)

0 - Use settings from TLS Provisioning Profile.

SRTP Parameters

Use the session parameters in the following list to enable or disable authentication and encryption for RTP and RTCP streams.

You can also turn off the session parameters to reduce the phone's processor usage.

`mr.srtp.audio.require`

Enable or disable a requirement for SRTP encrypted audio media between MR hubs and devices.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`mr.srtp.video.require`

Enable or disable a requirement for SRTP encrypted video media between hubs and devices.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`sec.srtp.answerWithNewKey`

1 (default) - Provides a new key when answering a call.

0 - Does not provide a new key when answering the call.

`sec.srtp.enable`

1 (default) - The phone accepts the SRTP offers.

0 - The phone declines the SRTP offers.

The defaults for SIP 3.2.0 is 0 when Null or not defined.

Change causes system to restart or reboot.

`sec.srtp.key.lifetime`

Specifies the lifetime of the key used for the cryptographic parameter in SDP.

Null (default)

0 - The primary key lifetime is not set.

Positive integer minimum 1024 or power of 2 notation - The primary key lifetime is set.

Setting this parameter to a non-zero value may affect the performance of the phone.

Change causes system to restart or reboot.

`sec.srtp.mki.enabled`

0 (default) - The phone sends two encrypted attributes in the SDP, one with MKI and one without MKI when the base profile is set as Generic.

1 - The phone sends only one encrypted value.

Change causes system to restart or reboot.

`sec.srtp.mki.startSessionAtOne`

0 (default) - The phone uses MKI value of 1.

1 - The MKI value increments for each new crypto key.

`sec.srtp.offer`

0 (default) - The secure media stream is not included in SDP of an SIP invite.

1 - The phone includes secure media stream along with the non-secure media description in SDP of an SIP invite.

Change causes system to restart or reboot.

`sec.srtp.offer.AES_ICM_256`

0 (default) - The system doesn't accept or offer `AES_256_CM_HMAC_SHA1_80` encryption.

1 - The system includes the `AES_256_CM_HMAC_SHA1_80` crypto suite in SRTP offers, and accepts it when offered in SIP calls.

Change causes system to restart or reboot.

`sec.srtp.offer.AES_GCM_256`

0 (default) - The system doesn't accept or offer `AEAD_AES_256_GCM` encryption.

1 - The system includes the `AEAD_AES_256_GCM` crypto suite in SRTP offers, and accepts it when offered in SIP calls.

Change causes system to restart or reboot.

`sec.srtp.offer.HMAC_SHA1_32`

0 (default) - The `AES_CM_128_HMAC_SHA1_32` crypto suite in SDP is not included.

1 - The `AES_CM_128_HMAC_SHA1_32` crypto suite in SDP is included.

Change causes system to restart or reboot.

`sec.srtp.offer.HMAC_SHA1_80`

1 (default) - The `AES_CM_128_HMAC_SHA1_80` crypto suite in SDP is included.

0 - The `AES_CM_128_HMAC_SHA1_80` crypto suite in SDP is not included.

Change causes system to restart or reboot.

`sec.srtp.padRtpToFourByteAlignment`

0 (default) - The RTP packet padding is not required when sending or receiving video.

1 - The RTP packet padding is required when sending or receiving video.

Change causes system to restart or reboot.

`sec.srtp.require`

0 (default) - The secure media streams are not required.

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

Change causes system to restart or reboot.

`sec.srtp.requireMatchingTag`

1 (default) - The tag values must match in the crypto parameter.

0 - The tag values are ignored in the crypto parameter.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noAuth.offer`

0 (default) - The authentication for RTP offer is enabled.

1 - The authentication for RTP offer is disabled.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noAuth.require`

0 (default) - The RTP authentication is required.

1 - The RTP authentication is not required.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noEncrypRTCP.offer`

0 (default) - The encryption for RTCP offer is enabled.

1 - The encryption for RTCP offer is disabled.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noEncrypRTCP.require`

0 (default) - The RTCP encryption is required.

1 - The RTCP encryption is not required.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noEncrypRTP.offer`

0 (default) - The encryption for RTP offer is enabled.

1 - The encryption for RTP offer is disabled.

Change causes system to restart or reboot.

`sec.srtp.sessionParams.noEncrypRTP.require`

0 (default) - The RTP encryption is required.

1 - The RTP encryption is not required.

Change causes system to restart or reboot.

`sec.srtp.simplifiedBestEffort`

1 (default) - The SRTP is supported with Microsoft Description Protocol Version 2.0 Extensions.

0 - The SRTP is not supported with Microsoft Description Protocol Version 2.0 Extensions.

`reg.x.secureTransportRequired`

0 (Default) - The phones register based on the transport priority received in the DNS response.

1 - The phones register only on the TLS transport in the DNS response if the transport is configured as DNSNaptr.

If the transport is configured as TLSOnly, then the phone registers to the configured SIP server. The phone doesn’t register if the transport is either TCP or UDP.

Visual Security Classification Parameters

To enable the visual security classification feature, you must configure settings on the BroadSoft BroadWorks server v20 or higher and on the phones.

If a phone has multiple registered lines, administrators can assign a different security classification to each line.

An administrator can configure security classifications as names or strings, then set the priority of each classification on the server in addition to the default security classification level Unclassified. The default security classification Unclassified displays until you set classifications on the server. When a user establishes a call to a phone not connected to this feature, the phone displays as Unclassified.

The following list includes the parameters you can use to configure visual security classification.

`voIpProt.SIP.serverFeatureControl.securityClassification`

0 (default) - The visual security classification feature for all lines on a phone is disabled.

1 - The visual security classification feature for all lines on a phone is enabled.

Change causes system to restart or reboot.

`reg.x.serverFeatureControl.securityClassification`

0 (default) - The visual security classification feature for a specific phone line is disabled.

1 - The visual security classification feature for a specific phone line is enabled.

VoSIP Parameter

The following table lists parameters to configure VoSIP.

`reg.X.rfc3329MediaSec.enable`

0 (default) – Disables the media security mechanisms negotiated between Phone and Outbound proxy without the need of multiple m-lines in the Session Description Protocol.

1 – Enables the media security mechanisms negotiated between Phone and Outbound proxy without the need of multiple m-lines in the Session Description Protocol.

Shared Lines Parameters

This section provides information about the shared lines parameters for Poly Trio.

Shared Call Appearances Parameters

This feature is dependent on support from a SIP call server. To enable shared call appearances on your phone, you must obtain a shared line address from your SIP service provider.

A shared line is an address of record managed by a call server. The server allows multiple endpoints to register locations against the address of record.

Poly devices support Shared Call Appearance (SCA) using the SUBSCRIBE-NOTIFY method specified in [RFC 6665](). The events used are:

- Call-info for call appearance state notification
- Line-seize for the phone to ask to seize the line

Use the parameters in the following list to configure options for this feature.

`reg.x.address`

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

Null (default)

String address

`reg.x.type`

private (default) - Use standard call signaling.

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

`call.shared.reject`

For shared line calls on the BroadWorks server.

0 - The phone displays a Reject soft key to reject an incoming call to a shared line.

1 - The Reject soft key does not display.

`call.shared.disableDivert`

1 (default) - Enable the diversion feature for shared lines.

0 - Disable the diversion feature for shared lines. Note that this feature is disabled on most call servers.

Change causes system to restart or reboot.

`call.shared.exposeAutoHolds`

0 (default) - No re-INVITE is sent to the server when setting up a conference on a shared line.

1 - A re-INVITE is sent to the server when setting up a conference on a shared line.

`call.shared.preferCallInfoCID`

0 (default) - The Caller-ID information received in the 200 OK status code is not ignored if the NOTIFY message received with caller information includes display information.

1 - The Caller-ID information received in the 200 OK status code is ignored if the NOTIFY message received with caller information includes display information.

call.shared.remoteActiveHoldAsActive

1 (default) - Shared remote active/hold calls are treated as a active call on the phone.

0 - Shared remote active/hold calls are not treated as a active call on the phone.

call.shared.seizeFailReorder

1 (default) - Play a re-order tone locally on shared line seize failure.

0 - Do not play a re-order tone locally on shared line seize failure.

Change causes system to restart or reboot.

divert.x.sharedDisabled

1 (default) - Disables call diversion features on shared lines.

0 - Enables call diversion features on shared lines.

Change causes system to restart or reboot.

voIpProt.SIP.specialEvent.lineSeize.nonStandard

Controls the response for a line-seize event SUBSCRIBE.

1 (default) - This speeds up the processing of the response for line-seize event.

0 - This will process the response for the line seize event normally

Change causes system to restart or reboot.

reg.x.ringType

The ringer to be used for calls received by this registration. The default is the first non-silent ringer.

If you use the configuration parameters ringer13 and ringer14 on a single registered line, the phone plays SystemRing.wav.

default (default)

ringer1 to ringer24

reg.x.line.y.label

Configure a unique line label for a shared line that has multiple line key appearances. This parameter takes effect when `up.cfgUniqueLineLabel=1` . If `reg.x.linekeys=1` , this parameter does not have any effect.

x = the registration index number starting from 1.

y = the line index from 1 to the value set by `reg.x.linekeys` . Specifying a string sets the label used for the line key registration on phones with multiple line keys.

If no parameter value is set for `reg.x.line.y.label` , the phone automatically numbers multiple lines by prepending “<y>_” where <y> is the line index from 1 to the value set by `reg.x.linekeys` .

reg.x.callsPerLineKey

Set the maximum number of concurrent calls for a single registration x. This parameter applies to all line keys using registration x. If registration x is a shared line, an active call counts as a call appearance on all phones sharing that registration.

12 (default)

1-24

Note: This per-registration parameter overrides `call.callsPerLineKey`.

`reg.x.header.pearlymedia.support`

0 (Default) - The p-early-media header is not supported on the specified line registration.

1 - The p-early-media header is supported by the specified line registration.

`reg.X.insertOBPAddressInRoute`

1 (Default) - The outbound proxy address is added as the topmost route header.

0 - The outbound proxy address is not added to the route header.

`reg.x.path`

0 (Default) - The path extension header field in the Register request message is not supported for the specific line registration.

1 - The phone supports and provides the path extension header field in the Register request message for the specific line registration.

`reg.x.regevent`

0 (default) - The phone is not subscribed to registration state change notifications for the specific phone line.

1 - The phone is subscribed to registration state change notifications for the specific phone line.

This parameter overrides the global parameter `voIpProt.SIP.regevent`.

`reg.x.rejectNDUBInvite`

Specify whether or not the phone accepts a call for a particular registration in case of a Network Determined User Busy (NDUB) event advertised by the SIP server.

0 (Default) - If an NDUB event occurs, the phone does not reject the call.

1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code.

`reg.x.server.y.specialInterop`

Specify the server-specific feature set for the line registration.

Standard (Default)

Standard

GENBAND

ALU-CTS

ocs2007r2

lcs2005

`reg.x.gruu`

1 - The phone sends sip.instance in the REGISTER request.

0 (default) - The phone does not send sip.instance in the REGISTER request.

`reg.x.serverFeatureControl.securityClassification`

0 (default) - The visual security classification feature for a specific phone line is disabled.

1 - The visual security classification feature for a specific phone line is enabled.

`reg.x.acd-login-logout reg.x.acd-agent-available`

0 (default) - The ACD feature is disabled for registration.

1 - If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.

`reg.x.auth.domain`

The domain of the authorization server that is used to check the user names and passwords.

Null (default)string

`reg.x.auth.optimizedInFailover`

The destination of the first new SIP request when failover occurs.

0 (default) - The SIP request is sent to the server with the highest priority in the server list.

1 - The SIP request is sent to the server which sent the proxy authentication request.

`reg.x.auth.password`

The password to be used for authentication challenges for this registration.

Null (default)

string - It overrides the password entered into the Authentication submenu on the Settings menu of the phone.

`reg.x.auth.userId`

User ID to be used for authentication challenges for this registration.

Null (default)

string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication submenu on the Settings menu of the phone.

`reg.x.auth.useLoginCredentials`

0 - (default) The Login credentials are not used for authentication to the server on registration x.

1 - The login credentials are used for authentication to the server.

`reg.x.broadsoft.userId`

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

Null (default)

string

`reg.x.broadsoft.useXspCredentials`

If this parameter is disabled, the phones use standard SIP credentials to authenticate.

1 (default) - Use this value, if phone lines are registered with a server running BroadWorks R19 or earlier.

0 - Set to 0, if phone lines are registered with a server running BroadWorks R19 SP1 or later.

`reg.x.broadsoft.xsp.password`

Enter the password associated with the BroadSoft user account for the line. Required only when `reg.x.broadsoft.useXspCredentials=1` .

Null (default)

string

`reg.x.displayName`

The display name used in SIP signaling and/or the H.323 alias used as the default caller ID.

Null (default)

UTF-8 encoded string

`reg.x.enablePvtHoldSoftKey`

This parameter applies only to shared lines.

0 (default) - To disable user on a shared line to hold calls privately.

1 - To enable users on a shared line to hold calls privately.

`reg.x.filterReflectedBlaDialogs`

1 (default) - bridged line appearance NOTIFY messages are ignored.

0 - bridged line appearance NOTIFY messages is not ignored

`reg.x.fwd.busy.contact`

The forward-to contact for calls forwarded due to busy status.

Null (default) - The contact specified by `divert.x.contact` is used.

string - The contact specified by `divert.x.contact` is not used

`reg.x.fwd.busy.status`

0 (default) - Incoming calls that receive a busy signal is not forwarded

1 - Busy calls are forwarded to the contact specified by `reg.x.fwd.busy.contact` .

`reg.x.fwd.noanswer.contact`

Null (default) - The forward-to contact specified by `divert.x.contact` is used.

string - The forward to contact used for calls forwarded due to no answer.

`reg.x.fwd.noanswer.ringCount`

The number of seconds the phone should ring for before the call is forwarded because of no answer. The maximum value accepted by some call servers is 20.

0 - (default)

1 to 65535

`reg.x.fwd.noanswer.status`

0 (default) - The calls are not forwarded if there is no answer.

1 - The calls are forwarded to the contact specified by `reg.x.noanswer.contact` after ringing for the length of time specified by `reg.x.fwd.noanswer.ringCount` .

`reg.x.gruu`

Specify if the phone sends sip.instance in the REGISTER request.

0 (default)

1

`reg.x.label`

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

The maximum number of characters for this parameter value is 256; however, the maximum number of characters that a phone can display on its user interface varies by phone model and by the width of the characters you use. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The rules for parameter up.cfgLabelElide determine how the label is truncated.

Null (default) - the label is determined as follows:

- If `reg.1.useteluriAsLineLabel=1` , then the tel URI/phone number/address displays as the label.
- If *`reg.1.useteluriAsLineLabel=0`*, then the value for `reg.x.displayName` , if available, displays as the label. If `reg.x.displayName` is unavailable, the user part of `reg.x.address` is used.

UTF-8 encoded string

`reg.x.lineAddress`

The line extension for a shared line. This parameter applies to private lines and BroadSoft call park and retrieve. If there is no extension provided for this parameter, the call park notification is ignored for the shared line.

Null (default)

String

`reg.x.lineKeys`

Specify the number of line keys to use for a single registration. The maximum number of line keys you can use per registration depends on your phone model.

1 (default)

1 to max

`reg.x.lisdisclaimer`

This parameter sets the value of the location policy disclaimer. For example, the disclaimer may be “Warning: If you do not provide a location, emergency services may be delayed in reaching your location should you need to call for help.”

Null (default)

string, 0 to 256 characters

`reg.x.musicOnHold.uri`

A URI that provides the media stream to play for the remote party on hold.

Null (default) - This parameter does not overrides `voIpProt.SIP.musicOnHold.uri` .

a SIP URI - This parameter overrides `voIpProt.SIP.musicOnHold.uri` .

`reg.x.offerFullCodecListUponResume`

1 (default) - The phone sends full audio and video capabilities after resuming a held call irrespective of the audio and video capabilities negotiated at the initial call answer.

0 - The phone does not send full audio and video capabilities after resuming a held call.

`reg.x.outboundProxy.address`

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

Null (default)

IP address or hostname

`reg.x.outboundProxy.failOver.failBack.mode`

The mode for failover failback (overrides `reg.x.server.y.failOver.failBack.mode`).

duration - (default) The phone tries the primary server again after the time specified by `reg.x.outboundProxy.failOver.failBack.timeout` expires.

newRequests - All new requests are forwarded first to the primary server regardless of the last used server.

DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.

`reg.x.outboundProxy.failOver.failBack.timeout`

3600 (default) -The time to wait (in seconds) before failback occurs (overrides `reg.x.server.y.failOver.failBack.timeout`).

0, 60 to 65535 - The phone does not fail back until a failover event occurs with the current server.

`reg.x.outboundProxy.failOver.failRegistrationOn`

1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration.

0 - The reRegisterOn parameter is enabled, existing registrations remain active.

`reg.x.outboundProxy.failOver.onlySignalWithRegistered`

1 (default) - The reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs.

0 - The reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed.

`reg.x.outboundProxy.failOver.reRegisterOn`

This parameters overrides `reg.x.server.y.failOver.reRegisterOn` .

0 (default) - The phone won't attempt to register with the secondary server.

1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server.

`reg.x.outboundProxy.port`

The port of the SIP server to which the phone sends all requests.

0 - (default)

1 to 65535

`reg.x.outboundProxy.transport`

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

DNSnaptr (default)

DNSnaptr, TCPpreferred, UDPOnly, TLS, TCPOnly

`reg.x.proxyRequire`

Null (default) - No Proxy-Require is sent.

string - Needs to be entered in the Proxy-Require header.

`reg.x.ringType`

The ringer to be used for calls received by this registration.

ringer2 (default) - Is the first non-silent ringer.

ringer1 to ringer24 - To play ringer on a single registered line.

`reg.x.serverFeatureControl.callRecording`

1 (default) - BroadSoft BroadWorks v20 call recording feature for individual phone lines is enabled.

0 - BroadSoft BroadWorks v20 call recording feature for individual phone lines is disabled.

`reg.x.serverFeatureControl.cf`

0 (default) - The server-based call forwarding is disabled.

1 - server based call forwarding is enabled.

Note: This parameter overrides `voIpProt.SIP.serverFeatureControl.cf` .

Change causes system to restart or reboot.

`reg.x.serverFeatureControl.dnd`

0 (default) - server-based do-not-disturb (DND) is disabled.

1 - server-based DND is enabled and the call server has control of DND.

Note: This parameter overrides `voIpProt.SIP.serverFeatureControl.dnd.` .

Change causes system to restart or reboot.

`reg.x.serverFeatureControl.localProcessing.cf`

0 (default) - If `reg.x.serverFeatureControl.cf` is set to 1 the phone does not perform local Call Forward behavior.

1 - The phone performs local Call Forward behavior on all calls received.

Note: This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.cf` .

`reg.x.serverFeatureControl.localProcessing.dnd`

0 (default) - If `reg.x.serverFeatureControl.dnd` is set to 1, the phone does not perform local DND call behavior.

1 - The phone performs local DND call behavior on all calls received.

Note: This parameter overrides `voIpProt.SIP.serverFeatureControl.localProcessing.dnd` .

`reg.x.serverFeatureControl.securityClassification`

0 (default) - The visual security classification feature for a specific phone line is disabled.

1 - The visual security classification feature for a specific phone line is enabled.

`reg.x.serverFeatureControl.signalingMethod`

Controls the method used to perform call forwarding requests to the server.

serviceMsForwardContact (default)

string

`reg.x.srtp.enable`

1 (default) - The registration accepts SRTP offers.

0 - The registration always declines SRTP offers.

Change causes system to restart or reboot.

`reg.x.srtp.offer`

This parameter applies to the registration initiating (offering) a phone call.

0 (default) - No secure media stream is included in SDP of a SIP INVITE.

1 - The registration includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE.

Change causes system to restart or reboot.

`reg.x.srtp.require`

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

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

Change causes system to restart or reboot.

`reg.x.srtp.simplifiedBestEffort`

1 (default) - Negotiation of SRTP compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported.

0 - No SRTP is supported.

Note: This parameter overrides `sec.srtp.simplifiedBestEffort` .

`reg.x.strictLineSeize`

0 (default) - Dial prompt is provided immediately without waiting for a successful OK from the call server.

1 - The phone is forced to wait for 200 OK on registration x when receiving a TRYING notify.

Note: This parameter overrides `voIpProt.SIP.strictLineSeize` for registration x.

`reg.x.tcpFastFailover`

0 (default) - A full 32 second RFC compliant timeout is used.

1 - failover occurs based on the values of `reg.x.server.y.retryMaxCount` and `voIpProt.server.x.retryTimeOut` .

`reg.x.thirdPartyName`

Null (default) - In all other cases.

string address -This field must match the `reg.x.address` value of the registration which makes up the part of a bridged line appearance (BLA).

`reg.x.useCompleteUriForRetrieve`

1 (default) - The target URI in BLF signaling uses the complete address as provided in the XML dialog document.

0 - Only the user portion of the XML dialog document is used and the current registrar's domain is appended to create the full target URI.

Note: This parameters overrides `voipPort.SIP.useCompleteUriForRetrieve` .

`reg.x.server.y.address`

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

Null (default) - SIP server does not accepts registrations.

IP address or hostname - SIP server that accepts registrations. If not Null, all of the parameters in this list override the parameters specified in `voIpProt.server.*`

`reg.x.server.y.expires`

The phone's requested registration period in seconds.

The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period.

3600 - (default)

positive integer, minimum 10

`reg.x.server.y.expires.lineSeize`

Requested line-seize subscription period.

30 - (default)

0 to 65535

`reg.x.server.y.expires.overlap`

The number of seconds before the expiration time returned by server x at which the phone should try to re-register.

The phone tries to re-register at half the expiration time returned by the server if the server value is less than the configured overlap value.

60 (default)

5 to 65535

`reg.x.server.y.failOver.failBack.mode`

duration (default) - The phone tries the primary server again after the time specified by `reg.x.server.y.failOver.failBack.timeout` .

newRequests - All new requests are forwarded first to the primary server regardless of the last used server.

DNSTTL - The phone tries the primary server again after a timeout equal to the DNS TTL configured for the server that the phone is registered to.

registration - The phone tries the primary server again when the registration renewal signaling begins.

This parameter overrides `voIpProt.server.x.failOver.failBack.mode)`

`reg.x.server.y.failOver.failBack.timeout`

3600 (default) - The time to wait (in seconds) before failback occurs.

0 - The phone does not fail back until a failover event occurs with the current server.

60 to 65535 - If set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.

`reg.x.server.y.failOver.failRegistrationOn`

1 (default) - The reRegisterOn parameter is enabled, the phone silently invalidates an existing registration (if it exists), at the point of failing over.

0 - The reRegisterOn parameter is disabled, existing registrations remain active.

`reg.x.server.y.failOver.onlySignalWithRegistered`

1 (default) - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, no signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs. If the phone attempts to send signaling associated with an existing call via an unregistered server (for example, to resume or hold a call), the call ends. No SIP messages are sent to the unregistered server.

0 - Set to this value and reRegisterOn and failRegistrationOn parameters are enabled, signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred).

`reg.x.server.y.failOver.reRegisterOn`

0 (default) - The phone does not attempt to register with the secondary server, since the phone assumes that the primary and secondary servers share registration information.

1 - The phone attempts to register with (or via, for the outbound proxy scenario), the secondary server. If the registration succeeds (a 200 OK response with valid expires), signaling proceeds with the secondary server.

This parameter overrides `voIpProt.server.x.failOver.reRegisterOn` .

`reg.x.server.y.port`

Null (default) - The port of the SIP server does not specifies registrations.

0 - The port used depends on `reg.x.server.y.transport` .

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

`reg.x.server.y.register`

1 (default) - Calls can't be routed to an outbound proxy without registration.

0 - Calls can be routed to an outbound proxy without registration.

See `voIpProt.server.x.register` for more information, see *SIP Server Fallback Enhancements on Polycom Phones - Technical Bulletin 5844* on Poly Engineering Advisories and Technical Notifications.

`reg.x.server.y.registerRetry.baseTimeOut`

For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Used in conjunction with `reg.x.server.y.registerRetry.maxTimeOut` to determine how long to wait.

60 (default)

10 - 120 seconds

`reg.x.server.y.registerRetry.maxTimeout`

For registered line x, set y to the maximum time period the phone waits before trying to re-register with the server. Use in conjunction with r `eg.x.server.y.registerRetry.baseTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.

180 - (default)

60 - 1800 seconds

`reg.x.server.y.retryMaxCount`

The number of retries attempted before moving to the next available server.

3 - (default)

0 to 20 - 3 is used when the value is set to 0.

`reg.x.server.y.retryTimeOut`

0 (default) - Use standard RFC 3261 signaling retry behavior.

0 to 65535 - The amount of time (in milliseconds) to wait between retries.

`reg.x.server.y.subscribe.expires`

The phone's requested subscription period in seconds after which the phone attempts to resubscribe at the beginning of the overlap period.

3600 seconds - (default)

10 - 2147483647 (seconds)

You can use this parameter in conjunction with `reg.x.server.y.subscribe.expires.overlap` .

`reg.x.server.y.subscribe.expires.overlap`

The number of seconds before the expiration time returned by server x after which the phone attempts to resubscribe. If the server value is less than the configured overlap value, the phone tries to resubscribe at half the expiration time returned by the server.

60 seconds (default)

5 - 65535 seconds

`reg.x.server.y.transport`

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

DNSnaptr (default) - If `reg.x.server.y.address` is a hostname and `reg.x.server.y.port` is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If `reg.x.server.y.address` is an IP address, or a port is given, then UDP is used.

TCPpreferred - TCP is the preferred transport; UDP is used if TCP fails.

UDPOnly - Only UDP is used.

TLS - If TLS fails, transport fails. Leave port field empty (defaults to `5061`) or set to `5061` .

TCPOnly - Only TCP is used.

`reg.x.server.y.useOutboundProxy`

1 (default) - Enables to use the outbound proxy specified in `reg.x.outboundProxy.address` for server x.

0 - Disable to use the outbound proxy specified in `reg.x.outboundProxy.address` for server x.

`divert.x.sharedDisabled`

1 (default) - Disables call diversion features on shared lines.

0 - Enables call diversion features on shared lines.

Change causes system to restart or reboot.

`call.shared.distinctiveLedOnHold`

0 (default) - The LED blinks red for both remotely held calls and locally held calls.

1 - The LED blinks as red and green for local hold calls, and blinks only red for remotely held calls.

Private Hold on Shared Lines Parameters

You can configure private hold only using configuration files; you cannot configure the feature on the Web Configuration Utility or from the local phone interface.

Use the parameters in the following list to configure this feature.

`call.shared.exposeAutoHolds`

Enable to send a re-INVITE to the server when setting up a conference on a shared line.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`reg.x.enablePvtHoldSoftKey`

Enable to allow users on a shared line to hold calls privately.

0 (default) - Disabled

1 - Enabled

Note: This parameter applies only to shared lines.

Intercom Calls Parameters

Use the parameters in the list below to configure the behavior of the calling and answering phone.

`feature.intercom.enable`

Enable or disable the intercom feature.

0 (default) - Disabled

1 - Enabled

`homeScreen.intercom.enable`

Enable to display the **Intercom** icon on the phone's **Home** screen.

1 (default) - Enabled

0 - Disabled

`voIpProt.SIP.intercom.alertInfo`

The string you want to use in the Alert-Info header. You can use the following characters: '@', '-' ,'_' , '.' .

If you use any other characters, NULL, or empty spaces, the call is sent as normal without the Alert-Info header.

Intercom (default)

Alpha - Numeric string

Group Paging Parameters

Administrators must enable paging and PTT before users can subscribe to a page group.

Use the parameters in the following list to configure this feature.

Note:
The default port used by Group Paging conflicts with the UDP port 5001 used by Polycom People+Content on the Poly Trio system. Since the port used by People+Content is fixed and cannot be configured, configure one of the following workarounds:

- Configure a different port for Group Paging using parameter `ptt.port` or
- Disable People+Content IP using parameter `content.ppcipServer.enabled='0'` .

`ptt.address`

The multicast IP address to send page audio to and receive page audio from.

224.0.1.116 (default)

Multicast IP address.

`ptt.pageMode.allowOffHookPages`

Enable to play group pages on handsets while they are on active calls.

0 (default) - Disabled. Priority and Emergency pages still play while handsets are on active calls.

1 - Enabled.

`ptt.pageMode.defaultGroup`

The paging group used to transmit an outgoing page if the user does not explicitly specify a group.

1 (default)

1 to 25

`ptt.pageMode.transmit.timeout.continuation`

The time (in seconds) to add to the initial timeout (`ptt.pageMode.transmit.timeout.initial`) for terminating page announcements. If this value is non-zero, **Extend** displays on the phone. Pressing **Extend** continues the initial timeout for the time specified by this parameter. If 0, announcements cannot be extended.

60 (default)

0 to 65535

`ptt.pageMode.transmit.timeout.initial`

The number of seconds to wait before automatically terminating an outgoing page announcement.

0 (default) -The page announcements do not automatically terminate.

0 to 65535 - The page announcements automatically terminate.

`ptt.pageMode.priorityGroup`

The paging group to use for priority pages.

24 (default)

1 to 25

`ptt.pageMode.payloadSize`

The page mode audio payload size.

20 (default)

10, 20, ..., 80 milliseconds

`ptt.pageMode.emergencyGroup`

The paging group used for emergency pages.

25 (default)

1 to 25

`ptt.pageMode.codec`

The audio codec to use for outgoing group pages. Incoming pages are decoded according to the codec specified in the incoming message.

G.722 (default)

G.711Mu, G.726QI, or G.722

`ptt.pageMode.displayName`

This display name is shown in the caller ID field of outgoing group pages. If Null, the value from `reg.1.displayName` is used.

NULL (default)

up to 64 octet UTF-8 string

`ptt.pageMode.enable`

Enable or disable group paging.

0 (default) - Disabled

1 - Enabled

`ptt.pageMode.group.x.available`

Enable to make the group (x) available to the user.

1 (default) - Enabled

0 - Disabled

`ptt.pageMode.group.x.allowReceive`

Enable to allow the phone to receive pages from the group (x).

1 (default) - Enabled

0 - Disabled

`ptt.pageMode.group.x.allowTransmit`

Enable to allow outgoing announcements to the group.

1 (default) - Enabled

0 - Disabled

`ptt.pageMode.group.x.label`

The label to identify the group.

ch24: Priority, ch25: Emergency, others: Null ch1, 24, 25: 1, others: 0 (default)

String

`ptt.pageMode.group.x.subscribed`

Subscribe the phone to the group.

A page mode group x, where x= 1 to 25. The `label` is the name used to identify the group during pages.

If `available` is disabled (0), the user cannot access the group or subscribe and the other page mode group parameters is ignored. If enabled, the user can access the group and choose to subscribe.

If `allowTransmit` is disabled (0), the user cannot send outgoing pages to the group. If enabled, the user may send outgoing pages.

1 (default) - If enabled, the phone subscribes to the group.

0 - If disabled, the phone does not subscribe to the group.

System Logging Parameters

This section provides information about the system logging parameters for Poly Trio.

Log File Collection and Storage Parameters

You can configure log file collection and storage using the parameters in the following list.

You must contact Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure log file collection and storage.

The Poly Trio solution uploads a system log file [MAC address]-plcmsyslog.tar.gz that contains Android logs and diagnostics. This file can be ignored but does contain minimal data that may be useful to investigate Android issues.

There is no way to prevent the system log file [MAC address]-plcmsyslog.tar.gz from uploading to the server and you cannot control it using the parameters `log.render.file.upload.append.sizeLimit` and `log.render.file.upload.append.limitMode`. However, you can control the frequency of uploads using `log.render.file.upload.system.period`.

`log.render.level`

Specify the events to render to the log files. Severity levels are indicated in brackets.

0 - SeverityDebug (7)

1 - SeverityDebug (7) - default

2 - SeverityInformational (6)

3 - SeverityInformational (6)

4 - SeverityError (3)

5 - SeverityCritical (2)

6 - SeverityEmergency (0)

`log.render.file.size`

Set the maximum file size of the log file. When the maximum size is about to be exceeded, the phone uploads all logs that have not yet been uploaded and erases half of the logs on the phone. You can use a web browser to read logs on the phone.

512 kb (default)

1 - 10240 kB for Trio 8500 and 8800

1 - 1000 for Trio 8300

`log.render.file.upload.period`

Specify the frequency in seconds between log file uploads to the provisioning server.

Note: The log file is not uploaded if no new events have been logged since the last upload.

172800 seconds (default) - 48 hours

`log.render.file.upload.append`

1 (default) - Log files uploaded from the phone to the server are appended to existing files. You must set up the server to append using HTTP or TFTP.

0 - Log files uploaded from the phone to the server overwrite existing files.

Note that this parameter is not supported by all servers.

`log.render.file.upload.append.sizeLimit`

Specify the maximum size of log files that can be stored on the provisioning server.

512kb (default)

Note that this parameter is not supported by HTTP/HTTPS or TFTP protocols. Logs generated and uploaded via HTTP/HTTPS or TFTP protocol must be deleted manually if needed.

`log.render.file.upload.append.limitMode`

Specify whether to stop or delete logging when the server log reaches its maximum size.

delete (default) - Delete logs and start logging again after the file reaches the maximum allowable size specified by `log.render.file.upload.append.sizeLimit`.

stop - Stop logging and keep the older logs after the log file reaches the maximum allowable size.

Note that this parameter is not supported by HTTP/HTTPS or TFTP protocols. Logs generated and uploaded via HTTP/HTTPS or TFTP protocol must be deleted manually if needed.

`log.render.file.upload.system.period`

Specify the frequency in seconds the Poly Trio system uploads the Android system log file MAC address]-plcmsyslog.tar.gz to the server.

86400 seconds (default)

0 - 2147483647 seconds

Logging Level, Change, and Render Parameters for Poly Trio

Use the following parameters to configure logging features.

`log.level.change.app`

Initial logging level for the Apps log module.

4 (default)

0 - 6

`log.level.change.bfcp`

Initial logging level for the BFCP content log module.

4 (default)

0 - 6

`log.level.change.dasvc`

Initial logging level for Device analytics logs.

4 (default)

0 - 6

`log.level.change.fec`

Sets the log level for video FEC.

4 (default)

0 - 6

`log.level.change.fecde`

Sets high volume log level to decode video FEC.

4 (default)

0 - 6

`log.level.change.fecen`

Sets high volume log level to encode video FEC.

4 (default)

0 - 6

`log.level.change.flk`

Sets the log level for the FLK logs.

4 (default)

0 - 6

`log.level.change.mr`

Initial logging level for the Networked Devices log module.

4 (default)

0 - 6

`log.level.change.mraud`

Initial logging level for the Networked Devices Audio log module.

4 (default)

0 - 6

`log.level.change.mrcam`

Initial logging level for the Networked Devices Camera log module.

4 (default)

0 - 6

`log.level.change.mrci`

Initial logging level for the Networked Devices HDMI/VGA Content Input log module.

4 (default)

0 - 6

`log.level.change.mrcon`

Initial logging level for the Networked Devices Connection log module.

4 (default)

0 - 6

`log.level.change.mrdis`

Initial logging level for the Networked Devices Display log module.

4 (default)

0 - 6

`log.level.change.mrmgr`

Initial logging level for the Networked Devices Manager log module.

4 (default)

0 - 6

`log.level.change.opxy`

Define the log level for the OPXY ** log. The default logging level for the OPXY ** log is Minor Error.

4 (default)

0 - 6

`log.level.change.ppcip`

Initial logging level for the People+Content IP log module.

4 (default)

0 - 6

`log.level.change.prox`

Initial logging level for the Proximity log module.

4 (default)

0 - 6

`log.level.change.ptp`

Initial logging level for the Precision Time Protocol log module.

4 (default)

0 - 6

`log.level.change.usba`

Sets the logging detail level for the USB audio log.

4 (default)

0 - 6

`log.level.change.usbh`

Sets the logging detail level for the USB HID log.

4 (default)

0 - 6

`log.level.change.xxx`

Controls the logging detail level for individual components. These are the input filters into the internal memory-based log system.

4 (default)

0 - 6

Possible values for xxx are acom, ares, app1, appsp, bluet, bdiag, brow, bsdir, cap, cdp, cert, cfg, cipher, clink, clist, cmp, cmr, copy, curl, daa, dapi, dasvc, dbs, dbuf, dhcpc, dis, dock, dot1x, dns, drvbt, ec, efk, ethf, flk, fec,fecde, fecen, fur, h323, hset, httpa, httpd, hw, ht, ib, key, ldap, lic, lldp, loc, log, mb, mcu, mobil, mrci, net, niche, ocsp, osd, pcap, pcd, pdc, peer, pgui, pkt, pmt, poll, pps, pres, pstn, ptt, push, pwrsv, rdisk, res, restapi, rtos, rtls, sec, sig, sip, slog, so, srtp, sshc, ssps, statc, statn, style, sync, sys, ta, task, tls, trace, ttrs, usb, usbio, util, utilm, vsr, wdog, wmgr, and xmpp.

`log.render.file`

Polycom recommends that you do not change this value.

1 (default)

0

`log.render.realtime`

Polycom recommends that you do not change this value.

1 (default)

0

`log.render.stdout`

Polycom recommends that you do not change this value.

0 (default)

1

`log.render.type`

Refer to the Event Time Stamp Formats table for time stamp type.

2 (default)

0 - 2

Logging Parameters

The phone can be configured so certain advanced logging tasks take place scheduled basis.

Poly recommends that you set the parameters listed below with consultation with Poly Technical Support. Each scheduled log task is controlled by a unique parameter set starting with `log.sched.x` where `x` identifies the task. A maximum of 10 schedule logs is allowed.

`log.sched.x.level`

The event class to assign to the log events generated by this command.

3 (default)

0 - 5

This needs to be the same or higher than `log.level.change.slog` for these events to display in the log.

`log.sched.x.period`

Specifies the time in seconds between each command execution.

15 (default)

positive integer

`log.sched.x.startDay`

When startMode is abs, specifies the day of the week to start command execution. 1=Sun, 2=Mon, ..., 7=Sat

7 (default)

0 - 7

`log.sched.x.startMode`

Starts at an absolute or relative time to boot.

Null (default)

0 - 64

`log.sched.x.startTime`

Displays the start time in seconds since boot when startMode is rel or displays the start time in 24-hour clock format when startMode is abs.

Null (default)

positive integer, hh:mm

Severity of Logging Event Parameter

You can configure the severity of the events that are logged independently for each module of PVOS.

This enables you to capture lower severity events in one part of the application, and high severity events for other components. Severity levels range from 0 to 6, where 0 is the most detailed logging and 6 captures only critical errors.

Note: User passwords display in level 1 log files.

You must contact Poly Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure log levels.

`log.level.change.module_name`

Set the severity level to log for the module name you specify. Not all modules are available for all phone models.

For a list of available module names, module descriptions, and log level severity, see refer to the Web Configuration Utility at **Settings > Logging > Module Log Level Limits**.

USB Logging Parameter

The following parameters configure the USB logging feature.

`feature.usbLogging.enabled`

0 (default) - Disables collecting logs using a USB flash drive.

1 - Enables collecting logs using a USB flash drive.

Third-Party Servers Parameters

This section provides information about third-party servers parameters for Poly Trio.

Anonymous Call Rejection Parameters

Use the parameters below to configure Anonymous Call Rejection Parameters.

Use the parameters in the following list to enable this feature.

`feature.broadsoft.xsi.AnonymousCallReject.enabled`

0 (default) - Does not display the Anonymous Call Rejection menu to users.

1 - Displays the Anonymous Call Rejection menu and the user can turn the feature on or off from the phone.

`feature.broadsoftUcOne.enabled`

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

`reg.x.broadsoft.userId`

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

Null (default)

string

Authentication for BroadWorks XSP Parameters

Use these parameters for authenticate Poly phones with BroadWorks server.

`reg.x.broadsoft.xsp.password`

Enter the password associated with the BroadSoft user account for the line. Required only when `reg.x.broadsoft.useXspCredentials=1`.

Null (default)

string

`reg.x.broadsoft.userId`

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

Null (default)

string

`reg.x.broadsoft.useXspCredentials`

If this parameter is disabled, the phones use standard SIP credentials to authenticate.

1 (default) - Use this value, if phone lines are registered with a server running BroadWorks R19 or earlier.

0 - Set to 0, if phone lines are registered with a server running BroadWorks R19 SP1 or later.

`reg.x.auth.userId`

User ID to be used for authentication challenges for this registration.

Null (default)

string - If the User ID is non-Null, it overrides the user parameter entered into the Authentication sub-menu on the Settings menu of the phone.

`reg.x.auth.password`

The password to be used for authentication challenges for this registration.

Null (default)

string - It overrides the password entered into the Authentication sub-menu on the Settings menu of the phone.

Basic Authentication for One Touch Dial Exchange Parameter

Use the following parameter to allow basic authentication for One Touch Dial exchange services.

`feature.exchange.allowBasicAuth`

Set to determine whether One Touch Dial (OTD) Exchange services use basic authentication.
0 (default) – OTD Exchange services do not use basic authentication.

1 – OTD Exchange services use basic authentication.

Change causes system to restart or reboot.

BroadSoft UC-One Configuration Parameters

The following list includes all parameters available to configure features in the BroadSoft UC-One application.

`feature.qml.enabled`

0 (default) - Disable the QML viewer on the phone. Note that the UC-One directory user interface uses QML as the user interface framework and the viewer is used to load the QML applications.

1 - Enable the QML viewer on phone.

Change causes system to restart or reboot.

`feature.broadsoftdir.enabled`

0 (default) - Disable simple search for Enterprise Directories.

1 - Enable simple search for Enterprise Directories.

Change causes system to restart or reboot.

`feature.broadsoftUcOne.enabled`

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

`feature.presence.enabled`

0 (default) - Disable the presence feature – including buddy managements and user status.

1 - Enable the presence feature with the buddy and status options.

`homeScreen.UCOne.enable`

1 (default) - Enable the UC-One Settings icon to display on the phone Home screen.

0 - Disable the UC-One Settings icon to display on the phone Home screen.

`dir.broadsoft.xsp.address`

Set the IP address or hostname of the BroadSoft directory XSP home address.

Null (default)

IP address

Hostname

FQDN

`dir.broadsoft.xsp.username`

To set the BroadSoft Directory XSP home address.

`dir.broadsoft.xsp.password`

Set the password used to authenticate to the BroadSoft Directory XSP server.

Null (default)

UTF-8 encoding string

`xmpp.1.auth.password`

Specify the password used for XMPP registration.

Null (Default)

UTF-8 encoded string

`xmpp.1.dialMethod`

For SIP dialing, the destination XMPP URI is converted to a SIP URI, and the first available SIP line is used to place the call.

SIP (default)

String min 0, max 256

`xmpp.1.jid`

Enter the Jabber identity used to register with the presence server, for example: `presence.test2@polycom-alpha.eu.bc.im`.

Null (default)

String min 0, max 256

`xmpp.1.roster.invite.accept`

Choose how phone users receive the BroadSoft XMPP invitation to be added to a buddy list.

prompt (default) - phone displays a list of users who have requested to add you as a buddy and you can accept or reject the invitation.

Automatic

xmpp.1.server

Sets the BroadSoft XMPP presence server to an IP address, host name, or FQDN, for example: `polycom-alpha.eu.bc.im`.

Null (default)

dotted-decimal IP address, host name, or FQDN.

xmpp.1.verifyCert

Enable or disable verification of the TLS certificate provided by the BroadSoft XMPP presence server.

1 (default) - Enabled

0 - Disabled

BroadWorks Anywhere Parameters

You can configure BroadWorks Anywhere using configuration files or the Web Configuration Utility.

Use the parameters in the following list to enable this feature.

feature.broadsoft.xsi.BroadWorksAnywhere.enabled

0 (default) - Disables and does not display the BroadWorks Anywhere feature menu on the phone.

1 - Enables the BroadWorks Anywhere feature menu on the phone.

feature.broadsoftUcOne.enabled

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

BroadSoft UC-One Directory Parameters

Use the parameters in the following list to configure the Polycom BroadSoft UC-One directory.

dir.broadsoft.regMap

Specify the registration line credentials you want to use for BroadSoft R20 Server or later to retrieve directory information from the BroadSoft UC-One directory when `dir.broadsoft.useXspCredentials` =0.

1 (default)

0 - Const_NumLineReg

dir.broadsoft.useXspCredentials

Specify which method of credentials the phone uses to sign in with the BroadSoft server.

1 (default) - Uses BroadSoft XSP credentials.

0 - Uses SIP credentials from `dir.broadsoft.regMap`.

BroadSoft UC-One Credential Parameters

Use the parameters in the following list to enable this feature.

`dir.broadsoft.xsp.address`

Set the IP address or hostname of the BroadSoft directory XSP home address.

Null (default)

IP address

Hostname

FQDN

`reg.x.broadsoft.userId`

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

Null (default)

string

`feature.broadsoftUcOne.enabled`

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

`dir.broadsoft.xsp.username`

To set the BroadSoft Directory XSP home address.

`dir.broadsoft.xsp.password`

Set the password used to authenticate to the BroadSoft Directory XSP server.

Null (default)

UTF-8 encoding string

`feature.broadsoftdir.enabled`

0 (default) - Disable simple search for Enterprise Directories.

1 - Enable simple search for Enterprise Directories.

Change causes system to restart or reboot.

Exchange Impersonation for Calendaring Parameter

When you configure Exchange Impersonation for calendaring, enter the Exchange account whose calendar should display on the Poly Trio into the parameter below.

`exchange.targetMailbox`

This parameter configures the calendaring Exchange account sign-in address.

NULL (default) - The calendar for the account entered in `exchange.auth.email` displays on the Poly Trio system.

String (maximum of 255 characters) - The calendar for the entered Exchange account displays on the Poly Trio system.

Line ID Blocking Parameters

Use the parameters below to configure Line ID Blocking.

Use the parameters in the following list to enable this feature.

`feature.broadsoft.xsi.LineIdblock.enabled`

0 (default) - Disables and does not display the Line ID Blocking feature menu on the phone.

1 - Enables the Line ID Blocking feature menu on the phone.

`feature.broadsoftUcOne.enabled`

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

Meeting SIP URI Parameters

The following table lists the parameters that configure dial-in information.

`exchange.meeting.join.promptWithList`

Specifies the behavior of the Join button on meeting reminder pop-ups.

0 (default) - Tapping Join on a meeting reminder should show a list of numbers to dial rather than immediately dialing the first one.

1 - A meeting reminder does not show a list of numbers to dial.

`exchange.meeting.parseWhen`

Specifies when to scan the meeting's subject, location, and description fields for dialable numbers.

NonSkypeMeeting (default)

Always

Never

`exchange.meeting.parseOption`

Select a meeting invite field to fetch a VMR or meeting number from.

All (default)

Location

LocationAndSubject

Description

Change causes a reboot.

`exchange.meeting.parseEmailsAsSipUris`

List instances of text like user@domain or user@ipaddress in the meeting description or subject under the More Actions pane as dialable SIP URIs.

0 (default) - it does not list the text as a dialable SIP URI

1 - it treats user@domain

`exchange.meeting.parseAllowedSipUriDomains`

List of comma-separated domains that will be permitted to be interpreted as SIP URIs

Null (default)

String (maximum of 255 characters)

Microsoft Exchange Parameters

The following parameters configure Microsoft Exchange integration.

`exchange.meeting.alert.followOfficeHours`

1 (default) - Enable audible calendar alerts during business hours.

0 - Disable audible calendar alerts.

`exchange.meeting.alert.tonePattern`

positiveConfirm (default) - Set the tone pattern of the reminder alerts using any tone specified by `se.pat.*`.

`exchange.meeting.alert.toneVolume`

10 (default) - Set the volume level of reminder alert tones.

0 - 17

`exchange.meeting.allowScrollingToPast`

0 (default) - Do not allow scrolling up in the Day calendar view to see recently past meetings.

1 - Allow scrolling up in the Day calendar view to see recently past meetings.

`exchange.meeting.hideAllDayNotification`

0 (default) - All-day and multi-day meeting notifications display on the **Calendar** screen.

`exchange.meeting.parseOption`

Select a meeting invite field to fetch a VMR or meeting number from.

All (default)

Location

LocationAndSubject

Description

Change causes a reboot.

`exchange.meeting.parseWhen`

NonSkypeMeeting (default) - Disable number-searching on the Calendar for additional numbers to dial while in Skype Meetings.

Always - Enable number-searching on the Calendar for additional numbers to dial while in Skype Meetings.

`exchange.meeting.phonePattern`

NULL (default)

string

The pattern used to identify phone numbers in meeting descriptions, where "x" is a digit or an asterisk(*) and "|" separates alternative patterns (for example, xxx-xxx-xxxx|604.xxx.xxxx).

`exchange.meeting.realConnectProcessing.outboundRegistration`

Choose a line number to use to make calls on Polycom RealConnect technology.

2 (default)

1 - 34

Change causes system to restart or reboot.

`exchange.meeting.realConnectProcessing.prefix.domain`

Define the One-Touch Dial meeting invite prefix domain. Example: "`mypolycom.com`"

`exchange.meeting.realConnectProcessing.prefix.value`

Define the One-Touch Dial meeting invite prefix value.

`exchange.meeting.realConnectProcessing.skype.enabled`

0 (default) – Disable the Skype for Business meeting on Polycom RealConnect technology.

1 - Enable the Skype for Business meeting on Polycom RealConnect technology.

Change causes system to restart or reboot.

`exchange.meeting.reminderEnabled`

1 (default) - Meeting reminders are enabled.

0 - Meeting reminders are disabled.

`exchange.meeting.reminderInterval`

Set the interval at which phones display reminder messages.

300 seconds (default)

60 - 900 seconds

`exchange.meeting.reminderSound.enabled`

1 (default) - The phone makes an alert sound when users receive reminder notifications of calendar events. Note that when enabled, alert sounds take effect only if `exchange.meeting.reminderEnabled` is also enabled.

0 - The phone does not make an alert sound when users receive reminder notifications of calendar events.

`exchange.meeting.reminderType`

Customize the calendar reminder and tone.

2 (default) - The reminder is always audible and visual.

1 - The first reminder is audible and visual reminders are silent.

0 - All reminders are silent.

`exchange.meeting.reminderWake.enabled`

1 (default) - The phone wakes from low power mode after receiving a calendar notification.

0 - The phone stays in low power mode after receiving a calendar notification.

`exchange.meeting.showAttendees`

1 (default) - Show the names of the meeting invitees.

0 - Hide the names of the meeting invitees.

`exchange.meeting.showDescription`

1 (default) - Show Agenda/Notes in Meeting Details that display after you tap a scheduled meeting on the Poly Trio system calendar.

0 - Hide the meeting Agenda/Notes.

`exchange.meeting.showLocation`

1 (default) - Show the meeting location.

0 - Hide the meeting location.

`exchange.meeting.showMoreActions`

1 (default) - Show More Actions in Meeting Details to allow users to choose a dial-in number.

0 - Hide More Actions in Meeting Details.

`exchange.meeting.showOnlyCurrentOrNext`

0 (default) - Disable the limitation to display only the current or next meeting on the Calendar.

1 - Display only the current or next meeting on the Calendar.

`exchange.meeting.showOrganizer`

1 (default) - Show the meeting organizer in the meeting invite.

0 - Hide the meeting organizer in the meeting invite.

`exchange.meeting.showSubject`

1 (default) - Show the meeting Subject.

0 - Hide the meeting Subject.

`exchange.meeting.showTomorrow`

1 (default) - Show meetings scheduled for tomorrow as well as for today.

0 - Show only meetings scheduled for today.

`exchange.menu.location`

Features (default) - Displays the Calendar in the global menu under **Settings > Features**.

Administrator - Displays the Calendar in the **Admin** menu at **Settings > Advanced > Administration Settings**.

`exchange.pollInterval`

The interval, in milliseconds, to poll the Exchange server for new meetings.

30000 (default)

4000 minimum

60000 maximum

`exchange.reconnectOnError`

1 (default) - The phone attempts to reconnect to the Exchange server after an error.

0 - The phone does not attempt to reconnect to the Exchange server after an error.

`exchange.server.url`

NULL (default)

string

The Microsoft Exchange server address.

`feature.EWSAutodiscover.enabled`

If you configure `exchange.server.url` and set this parameter to 1, preference is given to the value of `exchange.server.url`.

1 (default) - Exchange autodiscovery is enabled and the phone automatically discovers the Exchange server using the email address information.

0 - Exchange autodiscovery is disabled on the phone and you must manually configure the Exchange server address.

`feature.exchangeCalendar.enabled`

Generic Base Profile default is 0.

0 - The calendaring feature is disabled.

1 - The calendaring feature is enabled.

You must enable this parameter if you also enable `feature.exchangeCallLog.enabled`. If you disable `feature.exchangeCalendar.enabled`, also disable `feature.exchangeCallLog.enabled` to ensure call log functionality.

`exchange.multipleCalendarEvents.enabled`

1 (default) - Multiple calendar events display if at least two events begin within 15 minutes of each other.

0 - Only the next calendar event displays.

`feature.exchangeContacts.enabled`

Generic Base Profile default is 0.

1 - The Exchange call log feature is enabled and users can retrieve the call log histories for missed, received, and outgoing calls.

0 - The Exchange call log feature is disabled and users cannot retrieve call logs histories.

You must also enable the parameter `feature.exchangeCallLog.enabled` to use the Exchange call log feature.

`feature.exchangeVoiceMail.enabled`

Generic Base Profile default is 0.

1 - The Exchange voicemail feature is enabled and users can retrieve voicemails stored on the Exchange server from the phone.

0 - The Exchange voicemail feature is disabled and users cannot retrieve voicemails from Exchange Server on the phone.

You must also enable `feature.exchangeCalendar.enabled` to use the Exchange contact feature.

`feature.exchangeVoiceMail.skipPin.enabled`

0 (default) - Enable PIN authentication for Exchange Voicemail. Users are required to enter their PIN before accessing Exchange Voicemail.

1 - Disable PIN authentication for Exchange Voicemail. Users are not required to enter their PIN before accessing Exchange Voicemail.

`feature.lync.abs.enabled`

Generic Base Profile default is 0.

1 - Enable comprehensive contact search in the Skype for Business address book service.

0 - Disable comprehensive contact search in the Skype for Business address book service.

`feature.lync.abs.maxResult`

Define the maximum number of contacts to display in a Skype for Business address book service contact search.

12 (default)

5 - 50

`feature.wad.enabled`

Do not disable this parameter if you are using Skype Online or Web Sign-In.

1 (default) – The phone attempts to use Web auto-discovery and if no FQDN is available, falls back to DNS.

0 - The phone uses DNS to locate the server FQDN and does not use Web auto-discovery. Do not disable this parameter when using Skype for Business Online and Web Sign In.

`feature.contacts.readonly`

0 (default) - Skype for Business Contacts are editable.

1 - Skype for Business are read-only.

`up.oneTouchVoiceMail1`

Generic Base Profile default is 0.

0 - The phone displays a summary page with message counts. The user must press the Connect soft key to dial the voicemail server.

1 - The phone dials voicemail services directly (if available on the call server) without displaying the voicemail summary.

Microsoft Exchange Advanced Login Parameters

Use the following parameters to configure Microsoft Exchange Advanced Login.

`exchange.showSeparateAuth`

0 (default) - Phone disables the dual user sign-in mode.

1 - Phone enables the dual user sign-in mode.

`exchange.auth.email`

This parameter configures the email address of the Exchange account.

NULL (default)

String (maximum of 255 characters)

`device.loginAltCred.domain`

This parameter configures the domain of the Exchange account.

NULL (default)

String (maximum of 255 characters)

`device.loginAltCred.user`

This parameter configures the User ID of Exchange account.

NULL (default)

String (maximum of 255 characters)

`device.loginAltCred.password`

This parameter configures the password of Exchange Account.

NULL (default)

String (maximum of 32 characters)

`device.loginAltCred.domain.set`

This parameter overrides the value set for `device.loginAltCred.domain` using other configuration methods like the phone menu or the Web Configuration Utility.

0 (default)

`device.loginAltCred.user.set`

This parameter overrides the value set for `device.loginAltCred.user` using other configuration methods like the phone menu or the Web Configuration Utility.

0 (default)

`device.loginAltCred.password.set`

This parameter overrides the value set for `device.loginAltCred.password` using other configuration methods like the phone menu or the Web Configuration Utility.

0 (default)

Remote Office Parameters

Use the parameters in the following list to enable this feature.

`feature.broadsoft.xsi.RemoteOffice.enabled`

0 (default) - Disables the Remote Office feature menu on the phone.

1 - Enables and displays the Remote Office feature menu on the phone.

`reg.x.broadsoft.userId`

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

Null (default)

string

`feature.broadsoftUcOne.enabled`

0 (default) - Disables the BroadSoft UC-One feature.

1 - Enables the BroadSoft UC-One feature.

Change causes system to restart or reboot.

`dir.broadsoft.xsp.password`

Set the password used to authenticate to the BroadSoft Directory XSP server.

Null (default)

UTF-8 encoding string

Simultaneous Ring Parameters

Use the parameters below to configure Simultaneous Ring.

Use the parameters in the following list to enable this feature.

`feature.broadsoft.xsi.SimultaneousRing.enabled`

0 (default) - Disables and does not display the Simultaneous Ring Personal feature menu on the phone.

1 - Enables the Simultaneous Ring Personal feature menu on the phone.

`feature.broadsoftUcOne.enabled`

Enable or disable all BroadSoft UC-One features.

0 - Disabled

1 - Enabled

Skype for Business Private Meeting Parameters

Use the following parameters to configure Skype for Business private meetings.

`exchange.meeting.private.showAttendees`

0 (default) - Meetings marked as private in Outlook don't show the list of meeting attendees and invitees on the Poly Trio calendar.

1 - Meetings marked as private in Outlook show the list of meeting attendees and invitees on the Poly Trio calendar.

`exchange.meeting.private.showDescription`

0 (default) - Meetings marked as private in Outlook don't display a meeting description on the Poly Trio calendar.

1 - Meetings marked as private in Outlook display a meeting description on Poly Trio calendar.

`exchange.meeting.private.showLocation`

0 (default) - Meetings marked as private in Outlook don't display the meeting location on the Poly Trio calendar.

1 - Meetings marked as private in Outlook display the meeting location on the Poly Trio calendar.

`exchange.meeting.private.showSubject`

0 (default) - Meetings marked as private in Outlook don't display a subject line on Poly Trio calendar.

1 - Meetings marked as private in Outlook display a subject line on Poly Trio calendar.

`exchange.meeting.private.showMoreActions`

1 (default) - Meetings marked as private in Outlook display the **More Actions** button, when applicable.

0 - Meetings marked as private in Outlook don't display the **More Actions** button.

`exchange.meeting.private.showOrganizer`

1 (default) - Meetings marked as private in Outlook display the name of the meeting organizer on the Poly Trio calendar.

0 - Meetings marked as private in Outlook don't display the name of the meeting organizer on the Poly Trio calendar.

`exchange.meeting.private.enabled`

1 (default) - The Poly Trio system considers the private meeting flag for meetings marked as private in Outlook.

0 - Treat meetings marked as private in Outlook the same as other meetings.

`exchange.meeting.private.promptForPIN`

0 (default) - Disable the Skype for Business Conference ID prompt that allows users to join meetings marked as private.

1 - Enable the Skype for Business Conference ID prompt that allows users to join meetings marked as private.

User-Controlled Software Update Parameters

This section provides information about the user-controlled software update parameters for Poly Trio.

User-Controlled Software Update Parameters

You can set a polling policy and polling time period at which the phone polls the server for software updates and displays a notification on the phone to update software.

For example, if you set the polling policy to poll every four hours, the phone polls the server for new software every four hours and displays a notification that says a software update is available. Users can choose to update the software right then, or they can postpone it a maximum of three times for up to six hours. The phone automatically updates the software after three postponements or after six hours, whichever comes first.

The polling policy is disabled after the phone displays the software update notification.

After the software postponement ends, the phone displays the software update notification again.

`prov.usercontrol.enabled`

0 (default) - The phone doesn't display the software update notification and options and the phone reboots automatically to update the software.

1 - The phone displays the software update notification and options and the user can control the software download.

`prov.usercontrol.optionToIgnore`

You can configure the phone to give the user the ability to ignore software updates completely, or ignore until the next reboot or sync event.

1 - The Ignore and Ignore until next Reboot/Sync softkeys display on the phone’s local interface during a software upgrade alert.

0 (default) - Users can defer software upgrades up to three times.

`prov.usercontrol.postponeTime`

Sets the time interval for software update notification using the HH:MM format.

02:00 (default)

00:15

01:00

02:00

04:00

06:00

User Profiles Parameters

This section provides information about the user profiles parameters for Poly Trio.

User Profile Parameters

Before you configure user profiles, you must complete the following:

- Create a phone configuration file, or update an existing file, to enable the feature's settings.
- Create a user configuration file in the format <user>.cfg to specify the user's password, registration, and other user-specific settings that you want to define.

  Important: You can reset a user's password by removing the password parameter from the override file. This causes the phone to use the default password in the <user>.cfg file.

When you set up the user profile feature, you can set the following conditions:

- If users are required to always log in to use a phone and access their personal settings.
- If users are required to log in and have the option to use the phone as is without access to their personal settings.
- If users are automatically logged out of the phone when the phone restarts or reboots.
- If users remain logged in to the phone when the phone restarts or reboots.

Use the parameters in the following list to enable users to access their personal phone settings from any phone in the organization.

`prov.login.automaticLogout`

Specify the amount of time before a non-default user is logged out.

0 minutes (default)

0 to 46000 minutes

`prov.login.defaultOnly`

0 (default) - The phone can't have users other than the default user.

1 - The phone can have users other than the default user.

`prov.login.defaultPassword`

Specify the default password for the default user.

NULL (default)

`prov.login.defaultUser`

Specify the name of the default user. If a value is present, the user is automatically logged in when the phone boots up and after another user logs out.

NULL (default)

`prov.login.enabled`

0 (default) - The user profile is disabled.

1 - The user profile feature is enabled.

`prov.login.localPassword.hashed`

0 (default) - The user's local password is formatted and validated as clear text.

1 - The user's local password is created and validated as a hashed value.

`prov.login.localPassword`

Specify the password used to validate the user login. The password is stored either as plain text or as an encrypted SHA1 hash.

123 (default)

`prov.login.persistent`

0 (default) - Users are logged out if the handset reboots.

1 - Users remain logged in when the phone reboots.

`prov.login.required`

Set whether the phone requires the user to log in to the phone to use it.

0 (default) - Login not required.

1 - Login is required.

`prov.login.useProvAuth`

0 (default) - The phone doesn't use server authentication.

1 - The phones use server authentication and user login credentials are used as provisioning server credentials.

`voIpProt.SIP.specialEvent.checkSync.downloadCallList`

0 (default) - The phone does not download the call list for the user after receiving a checksync event in the NOTIFY.

1 - The phone downloads the call list for the user after receiving a checksync event in the NOTIFY.

Video Parameters for Poly Trio

This section provides information about the video parameters for Poly Trio.

Audio-only or Audio-Video Call Parameters

The following parameters configure whether the phone starts a call with audio and video.

`up.homeScreen.audioCall.enabled`

Devices that support video calling show an 'Audio Call' button on the Home screen to initiate audio-only calls.

0 (default) - Disable a Home screen icon that allows users to make audio-only calls.

1 - Enable a Home screen icon that allows users to make audio-only calls.

`video.autoStartVideoTx`

1 (default) - Automatically begin video to the far side when you start a call.

0 - Video to the far side does not begin.

`audioVideoToggle.callMode.persistent`

0 (default) - Resets the call mode set by a user to the default.

1 - Maintains the call mode set by a user.

`video.callMode.default`

Allow the user to begin calls as audio-only or with video.

video (default)

audio - Set the initial call to audio only and video may be added during a call.

On a Poly Trio solution, you can combine this parameter with `video.autoStartVideoTx`.

`video.enable`

1 (default) - Enables video in outgoing and incoming calls.

0 - Disables video in outgoing and incoming calls.

`video.localCameraView.idleState`

1 (default) – Enables the default local camera view state while not on a call.

0 – Disables the default local camera view state while not on a call.

`video.localCameraView.userControl`

Set to enable users to control how the local call view (LCV) displays during a call.

Persistent (default) - Enables users to access the Layout menu and control how the LCV displays before a call.

PerSession - Enables users to access the Layout menu and control how the LCV displays before or during a call.

Hidden - Hides the Layout menu so that users cannot control how the LCV displays.

Camera Recalibration Parameters

Use the following parameters to configure recalibration activity.

`video.camera.recalibrate.uponIdle`

Set to enable the EagleEye IV USB camera to recalibrate when the system enters an idle state.

0 (default) - Disabled

1 - Enabled

`video.camera.recalibrate.uponWakeUp`

Set to enable the EagleEye IV USB camera to recalibrate when the system wakes up from Low Power Mode.

0 (default) - Disabled

1 - Enabled

Note: If enabled and an incoming call wakes the camera, it skips recalibration to immediately begin the video call.

Camera Specific Preset Parameters

Use the following parameters to set presets to automatically select a camera associated with a preset.

`video.camera.multiCamera.enabled`

Set to indicate if one or more cameras are paired with the Poly Trio system and you want to automatically switch to a paired camera when a user selects a preset.
0 (default) A single camera is paired with the Poly Trio system.

1 - More than one camera is paired with the Poly Trio system.

`video.camera.selection.x.uniqueId`

Enter the unique ID of the camera.
NULL (default)

String

`video.camera.preset.x.selection`

Enter the unique ID of the camera associated with the selected preset. Leave blank if preset applies to the active camera.

NULL (default)

String

`video.camera.preset.home.selection`

Enter the unique ID of the camera associated with the selected preset. Leave blank if preset applies to the active camera.
NULL (default)

String

H.323 Protocol Parameters

Using the configuration parameters, you do the following:

- Configure SIP and H.323 protocols
- Set up a SIP and H.323 dial plan
- Set up manual protocol routing using soft keys
  If the phone can't determine the protocol to place a call, the **Use SIP** and **Use H.323** soft keys display, and users must select one to place the call.
- Configure auto-answering on H.323 calls only
- Set the preferred protocol to SIP
- Configure one SIP line, one H.323 line, and a dual protocol line—phones can use both SIP and H.323
- Set the preferred protocol for off-hook calls on the third (dual protocol) line to SIP

H.323 is supported only on Poly Trio systems.

`up.manualProtocolRouting`

Specifies whether to choose a protocol routing or use a default protocol.

1 (Default) - User is presented with a protocol routing choice in situations where a call can be placed using either protocol (for example, with SIP and H.323 protocols).

0 - Default protocol is used.

`up.manualProtocolRouting.softKeys`

Display soft keys that control **Manual Protocol Routing** options.

1 (Default) - Soft keys are enabled. Use soft keys to choose between the SIP or H.323 protocol.

0 - Soft keys for protocol routing do not display.

`call.autoAnswer.H323`

Enables and disables auto-answer for H.323 calls.

0 (default) - Disabled

1 - Enabled

`call.enableOnNotRegistered`

Enable or disable calls on the phone when it is not registered. When enabled, the phones can make calls using the H.323 protocol even though an H.323 gatekeeper is not configured.

Lync Base Profile – 0 (default)

Generic Base Profile – 1 (default)

1 - Enabled

0 - Disabled

Change causes system to restart or reboot.

`call.autoAnswer.videoMute`

0 (default) - Video begins transmitting (video Tx) automatically after a call is auto-answered.

1 - User must start video transmission (video Tx) manually when a call is auto-answered.

`call.autoRouting.preferredProtocol`

SIP (default) - Calls are placed via SIP if available or via H.323 if SIP is not available.

H323 - Calls are placed via H.323 if available, or via SIP if H.323 is not available.

`call.autoRouting.preference`

line - Calls are placed via the first available line, regardless of its protocol capabilities. If the first available line has both SIP and H.323 capabilities, the preferred protocol is used (`call.autoRouting.preferredProtocol`).

protocol - The first available line with the preferred protocol activated is used, if available. If not available, the first available line is used. Note that auto-routing is used when manual routing selection features (`up.manualProtocolRouting`) are disabled.

`reg.x.protocol.H323`

Enable or disable H.323 signaling for registration x.

0 (default) - Disabled

1 - Enabled

`reg.x.server.H323.y.address`

Address of the H.323 gatekeeper.

Null (default)

IP address or host name

`reg.x.server.H323.y.port`

Port to be used for H.323 signaling. If set to Null, 1719 (H.323 RAS signaling) is used.

0 (default)

0 to 65535

`reg.x.server.H323.y.expires`

Desired registration period.

3600

Positive integer

`voIpProt.H323.autoGateKeeperDiscovery`

1 (default) - The phone will attempt to discover an H.323 gatekeeper address via the standard multicast technique, provided that a statically configured gatekeeper address is not available.

0 - The phone will not send out any gatekeeper discovery messages.

Change causes system to restart or reboot.

`voIpProt.H323.blockFacilityOnStartH245`

0 (default) - Facility messages when using H.245 are not removed.

1 - Facility messages when using H.245 are removed.

Change causes system to restart or reboot.

`voIpProt.H323.dtmfViaSignaling.enabled`

Enable or disable use of H.323 signaling channel for DTMF key press transmission.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`voIpProt.H323.dtmfViaSignaling.H245alphanumericMode`

1 (default) - The phone supports H.245 signaling channel alphanumeric mode DTMF transmission.

0 - The phone does not support H.245 signaling channel alphanumeric mode DTMF transmission

Note: If both alphanumeric and signal modes can be used, the phone gives priority to DTMF.

Change causes system to restart or reboot.

`voIpProt.H323.dtmfViaSignaling.H245signalMode`

1 (default) - The phone will support H.245 signaling channel signal mode DTMF transmission.

0 - The phone will not support H.245 signaling channel signal mode DTMF transmission.

Change causes system to restart or reboot.

`voIpProt.H323.enable`

0 (default) - The H.323 protocol is not used for call routing, dial plan, DTMF, and URL dialing.

1 - The H.323 protocol is used for call routing, dial plan, DTMF, and URL dialing.

Change causes system to restart or reboot.

`voIpProt.H323.local.port`

Local port for sending and receiving H.323 signaling packets.

0 - 1720 is used for the local port but is not advertised in the H.323 signaling.

0 to 65535 - The value is used for the local port and it is advertised in the H.323 signaling.

Change causes system to restart or reboot.

`voIpProt.H323.local.RAS.port`

Specifies the local port value for RAS signaling.

1719 (default)

1 to 65535

Change causes system to restart or reboot.

`voIpProt.server.H323.x.address`

Specify the address of the H.323 gatekeeper. Only one H.323 gatekeeper per phone is supported. If more than one is configured, only the first is used.

Null (default)

IP address or host name

`voIpProt.server.H323.x.port`

Designate a port to be used for H.323 signaling. The H.323 gatekeeper RAS signaling uses UDP, while the H.225/245 signaling uses TCP.

1719 (default)

0 to 65535

`voIpProt.server.H323.x.expires`

Set a desired registration period.

3600 (default)

positive integer.

`sec.H235.mediaEncryption.enabled`

Enable or disable H.235 media encryption.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`sec.H235.mediaEncryption.offer`

0 (default) - The media encryption offer is not initiated with the far-end.

1 - If `sec.H235.mediaEncryption.enabled` is also set to `1`, media encryption negotiations are initiated with the far-end; however, successful negotiations are not a requirement for the call to complete.

Change causes system to restart or reboot.

`sec.H235.mediaEncryption.require`

0 (default) - The media encryption requirement is not required.

1 - If `sec.H235.mediaEncryption.enabled` is also set to `1`, media encryption negotiations are initiated or completed with the far end, but if negotiations fail, the call is dropped.

Change causes system to restart or reboot.

Per-Camera Video Parameters

You can use the following per-camera parameters to configure camera and video options for each type of Poly camera used in your environment..

Using the parameter video.camera.x.type, you can configure parameters differently for the following cameras supported with Poly Trio 8500 and 8800 systems paired with Poly Trio Visual+:

- Polycom EagleEye IV USB camera
- Polycom EagleEye Mini USB camera
- Poly EagleEye Cube USB camera
- Polycom EagleEye Director II camera (Poly Trio 8800 only)
- Logitech C930e webcam

You also can use the parameter video.camera.x.type to configure the following cameras that are supported with a paired Trio VisualPro system:

- Polycom EagleEye IV camera
- Polycom EagleEye Director II camera

- Polycom EagleEye Producer camera
- Polycom EagleEye Acoustic camera
- Poly EagleEye Cube HDCI camera
- Poly Studio USB video bar

Below is an example of per-camera configurations for the EagleEye IV USB camera and EagleEye Mini USB camera.

Example Per-Camera Configuration

Camera Type Configuration	Per-Camera Configuration
video.camera.1.type = EagleEyeIVUSB	video.camera.1.focus.auto=1 video.camera.1.orientation=Inverted
video.camera.2.type=EagleEyeMini	video.camera.2.focus.auto=0 video.camera.2.focus.range=150

For information about camera parameters with a paired RealPresence Group Series system, see the *Poly Trio with Polycom RealPresence Group Series Integration Guide* in the Polycom Documentation Library.

`call.singleKeyPressCameraControls`

1 (default) - Tapping **Camera** in the call view directly shows the **Camera Controls** menu for Poly USB cameras.

0 - Tapping **Camera** in the call view shows a menu with camera preferences, presets, and camera controls menu items for Poly USB cameras.

`feature.fecc.allowLocalCameraFarEndControl`

1 (default) – Allow the far-end user to control the local near-end camera controls, such as pan, tilt, and zoom, for the Poly USB cameras.

0 - The far-end user cannot control the local near-end camera.

`video.camera.x.attendeesCountEnable`

Enable to allow camera to count attendees on a video call.

0 (default) - Disabled

1 - Enabled

`video.camera.gamma`

Set the factor to use for gamma correction applied to each frame of video captured by the Poly USB cameras. You can use this setting to correct for video that appears too dark or too light.

NULL (default)

0 -1000

`video.camera.hue`

Use to correct the color of video captured by the Poly USB cameras.

NULL (default)

0 - 1000

`video.camera.invertPanControl`

Invert the direction of the pan control for the Poly USB camera.

0 (default)

`video.camera.orientation`

Specify the camera mounting orientation for Poly USB cameras: Normal or Inverted (upside down)

Normal (default)

Inverted

`video.camera.x.presetIndex`

Set the number of presets available.

NULL (default)

0-1000

`video.camera.preset.x.label`

Enter a label for the Poly USB camera preset.

NULL (default)

String 0 – 12 characters

`video.camera.preset.x.pan`

Set the pan for the Poly USB camera presets, where x equals the preset.

NULL (default)

0 - 1000

`video.camera.preset.x.tilt`

Set the tilt for the Poly USB camera presets, where x equals the preset.

NULL (default)

0 - 1000

`video.camera.preset.x.zoom`

Set the zoom for the EagleEye IV USB camera presets, where x equals the preset.

NULL (default)

0 - 1000

`video.camera.x.autoWhiteBalance`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

1 – Enable auto white balance. This parameter overrides `video.camera.whiteBalance`.

0 – Disable auto white balance.

`video.camera.x.backlightCompensation`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Set the backlight compensation of the video stream.

NULL (default)

0 - 1000

This parameter overrides `video.camera.backlightCompensation`.

`video.camera.x.brightness`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Sets the brightness level of the video stream. The value range is from 0 (dimmest) to 1000 (brightest).

NULL (default)

3

0 - 1000

This parameter overrides `video.camera.brightness`.

`video.camera.x.contrast`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Sets the contrast level of the video stream. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast).

NULL (default)

0 - 1000

This parameter overrides `video.camera.contrast`.

`video.camera.x.flickerAvoidance`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Null (default) - Flicker avoidance is automatic.

50hz AC power frequency flicker avoidance (Europe/Asia).

60hz AC power frequency flicker avoidance (North America).

Disabled

This parameter overrides `video.camera.flickerAvoidance`.

`video.camera.x.focus.auto`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

NULL (default)

0 - Disable the camera's automatic focus.

1 - Enable the camera's automatic focus. This parameter overrides `video.camera.focus.auto`.

Change causes system to restart or reboot.

`video.camera.x.focus.range`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Specify the distance to the camera's optimally-focused target.

NULL (default)

0

255

This parameter overrides `video.camera.focus.range`.

`video.camera.x.gamma`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Set the factor to use for gamma correction applied to each frame of video captured. You can use this setting to correct video that appears too dark or too light.

NULL (default)

0

1000

This parameter overrides `video.camera.gamma`.

`video.camera.x.hue`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Use to correct the color of video captured.

NULL (default)

500

1000

This parameter overrides `video.camera.hue`.

`video.camera.x.orientation`

Specify the camera mounting orientation for Poly USB cameras: Normal or Inverted (upside down)

Normal (default)

Inverted

This parameter overrides `video.camera.orientation`.

`video.camera.x.saturation`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Sets the saturation level of video captured by any supported USB camera.

NULL (default)

0 - 1000

This parameter overrides `video.camera.saturation`.

`video.camera.x.sharpness`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Sets the sharpness level of video captured.

NULL (default)

0 - 1000

This parameter overrides `video.camera.sharpness`.

`video.camera.x.trackingEnabled`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Enables automatic camera tracking. You can then set the tracking type, speed, and size.

0 - Disables camera tracking.

`video.camera.x.trackingFramingMode`

For the EagleEye Director II or EagleEye Producer camera:

0 (default) - Frame Speaker: Frames the active speaker.

1 - Frame Group: Frames the participants in the room (camera movement is seen on the far end).

2 - Frame Group with Transition (EagleEye Producer only): Frames the participants in the room (camera movement is seen on the far end).

`video.camera.x.trackingFramingSize`

For the EagleEye Director II or EagleEye Producer camera:

1 - Medium (default): Average-sized frame.

0 - Wide: Most expansive frame.

2 - Tight: Close-up frame.

`video.camera.x.trackingPipEnabled`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Enables People in Picture (PIP), which displays a group or room view to far-end participants.

0 - Disables PIP.

`video.camera.x.trackingSpeed`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Normal: Tracks transitions at a medium rate.

0 - Slow: Tracks transitions slowly.

2 - Fast: Tracks transitions quickly.

`video.camera.x.type`

Choose a camera type that corresponds to x, where x = 1 to 3. The value set for x in this parameter and related parameters determines the configuration settings for the camera type you specify.

NULL (default)

EagleEyeMini

EagleEyeDirectorII

EagleEyeProducer

EagleEyeAcoustic

EagleEyeCubeHDCI

EagleEyeCubeUSB

EagleEyeIVUSB

EagleEyeIV12x

EagleEyeIV4x

LogitechC930e

PolycomStudio

`video.camera.x.whiteBalance`

Set for per-camera configuration when you specify the camera type using the `video.camera.x.type` parameter.

Use to correct the white balance tint of video captured.

NULL (default)

0 - 1000

This parameter overrides `video.camera.whiteBalance`.

Video Call Overlay Parameters

Use the following parameters to configure the video call overlay that displays during video calls on monitors connected to he Poly Trio Visual+ or Poly VisualPro accessory.

`video.conf.galleryView.overlayPosition`

Use this parameter to set the location of the video overlay.

Bottom (default) - The video overlay displays at the bottom of the monitor screen.

Top - The video overlay displays at the top of the monitor screen.

None - The video overlay does not display.

Note: For Poly Trio VisualPro, you can only choose the values Bottom or None to configure `video.conf.galleryView.overlayPosition`.

`video.conf.galleryView.overlayTimeout`

Set the timer for the video overlay on the connected monitor. The overlay disappears after the time you set.

0 (default) - The video overlay does not time out and displays indefinitely.

0 - 60000 ms

Video Codec Parameters for Poly Trio

To support Poly Trio solution video interoperability with Cisco, set the following parameters:

- video.codecPref.H264HP="0"
- video.codecPref.H264HP.packetizationMode0="0"
- video.codecPref.H264="0"

Use the parameters in the following table to prioritize and adjust the video codecs used by the Poly Trio solution.

`video.codecPref.H264HP`

Sets the H.264 High Profile video codec preference priority.

0 - 8

2 (default)

`video.codecPref.H264HP.packetizationMode0`

0 - 8

5 (default)

`video.codecPref.H264SVC`

`video.codecPref.Xdata`

Sets the Remote Desktop Protocol (RDP) codec preference priority. A value of 1 indicates the codec is the most preferred and has highest priority.

0 - 8

7 (default)

`video.codecPref.XH264UC`

Sets the Microsoft H.264 UC video codec preference priority.

0 - 8

1 (default)

`video.codecPref.XUlpFecUC`

Set the forward error correction (FEC) codec priority.

8 (default)

0 - 8

Video and Camera Parameters

Use the following parameters to configure video and camera options for all Poly cameras.

`feature.fecc.enabled`

1 (default) - Enable far-end camera control.

0 - Disable far-end camera control.

Change causes system to restart or reboot.

`feature.fecc.payload`

Set the RTP payload used to receive far-end camera control data.

124 (default)

100 - 127

`homeScreen.camera.enable`

0 (default) - A **Camera** menu item is shown on the main menu.

1 - A **Camera** menu item displays on the Home Screen allowing users to pan, tilt, or zoom.

`mr.video.camera.focus.auto`

NULL (default)

0 - Disable the camera's automatic focus.

1 - Enable the camera's automatic focus.

Change causes system to restart or reboot.

`mr.video.camera.focus.range`

Specify the distance to the camera's optimally-focused target.

NULL (default)

0

0 - 255

`reg.x.fecc.enabled`

1 (default) - Enable far-end camera control for the line you specify with x.

0 - Disable far-end camera control for the line.

`up.arrow.repeatDelay`

Choose the milliseconds (ms) an arrow button must be held before the arrow starts repeating in the **Camera Controls** menu for supported Poly USB cameras.

500 ms (default)

100 – 5000 ms

`up.arrow.repeatRate`

Choose the milliseconds (ms) between repeated simulated presses while an arrow button is being held down. This applies to the arrows in the Camera Controls menu for supported Poly USB cameras.

80 ms (default)

50 – 2000 ms

`video.camera.attendeesCountEnable`

Enable to allow camera to count attendees on a video call.

0 (default) - Disabled

1 - Enabled

`video.camera.autoWhiteBalance`

0 – Disable auto white balance.

1- Enable auto white balance.

`video.camera.backlightCompensation`

NULL (default)

0 - 1000

`video.camera.brightness`

Sets the brightness level of the video stream. The value range is from 0 (dimmest) to 1000 (brightest).

NULL (default)

0 - 1000

`video.camera.contrast`

Sets the contrast level of the video stream for all supported USB cameras. The value range is from 0 (no contrast increase) to 3 (most contrast increase), and 4 (noise reduction contrast).

NULL (default)

0 - 1000

`video.camera.controlStyle`

Choose whether to control pan and tilt Poly USB cameras with directional arrow buttons or separate pan/tilt sliders.

Default (default)

Alternate

`video.camera.flickerAvoidance`

Sets the flicker avoidance for all supported USB cameras.

Null (default) - Flicker avoidance is automatic.

50hz AC power frequency flicker avoidance (Europe/Asia).

60hz AC power frequency flicker avoidance (North America).

Disabled

`video.camera.focus.auto`

NULL (default)

0 - Disable the camera's automatic focus.

1 - Enable the camera's automatic focus.

Change causes system to restart or reboot.

`video.camera.focus.range`

Specify the distance to the camera's optimally-focused target.

NULL (default)

0 - 255

`video.camera.frameRate`

Sets the target frame rate (frames per second) for all supported USB cameras. Values indicate a fixed frame rate from 5 (least smooth) to 30 (most smooth).

25 (default)

5 - 30

If `video.camera.frameRate` is set to a decimal number, the value 25 is used instead.

`video.camera.menuLocation`

Specify if camera settings, including preset storage and modifications, display under the **Advanced** menu for administrators or the **Basic** menu for users.

Basic (default)

Advanced

`video.camera.preset.home.pan`

Set the pan coordinate for a camera home preset.

Default values are set by and depend on the camera you are using.

0 – 1000

`video.camera.preset.home.tilt`

Set the tilt coordinate for a camera home preset.

Default values are set by and depend on the camera you are using.

0 - 1000

`video.camera.preset.home.uponIdle.delay`

Set the number of minutes after the idle timeout expires to move the camera to the home preset.

0 (default)

0 - 3600

`video.camera.preset.home.uponIdle.enabled`

0 (default) – Do not move the camera to the home preset when the camera is idle.

1 - Move the camera to the home preset when the camera is idle.

`video.camera.preset.home.zoom`

Set the zoom coordinate for a camera home preset.

Default values are set by and depend on the camera you are using.

0 - 1000

`video.camera.presetIndex`

Set the number of presets available.

NULL (default)

0 - 1000

`video.camera.saturation`

Sets the saturation level of video captured by any supported USB camera.

NULL (default)

0 - 1000

`video.camera.sharpness`

Sets the sharpness level of video captured.

NULL (default)

0 - 1000

`video.camera.trackingEnabled`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Enables automatic camera tracking. You can then set the tracking type, speed, and size.

0 - Disables camera tracking.

`video.camera.trackingFramingMode`

For the EagleEye Director II or EagleEye Producer camera:

0 (default) - Frame Speaker: Frames the active speaker.

1 - Frame Group: Frames the participants in the room (camera movement is seen on the far end).

2 - Frame Group with Transition (EagleEye Producer only): Frames the participants in the room (camera movement is seen on the far end).

`video.camera.trackingFramingSize`

For the EagleEye Director II or EagleEye Producer camera:

1 - Medium (default): Average-sized frame.

0 - Wide: Most expansive frame.

2 - Tight: Close-up frame.

`video.camera.trackingPipEnabled`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Enables People in Picture (PIP), which displays a group or room view to far-end participants.

0 - Disables PIP.

`video.camera.trackingSpeed`

For the EagleEye Director II or EagleEye Producer camera:

1 (default) - Normal: Tracks transitions at a medium rate.

0 - Slow: Tracks transitions slowly.

2 - Fast: Tracks transitions quickly.

`video.camera.whiteBalance`

Use to correct the white balance tint of video captured by any supported USB camera.

NULL (default)

0 - 1000

`video.localCameraView.callState`

Set to determine how the local call view (LCV) displays when the phone is in a call.

1 (default) - The local camera view displays on the connected monitor.

0 - The local camera view does not display on the connected monitor.

This parameter applies only when `video.localCameraView.userControl` is set to PerSession or Hidden.

`video.localCameraView.idleState`

Set to determine how the local call view (LCV) displays when the phone is idle.

0 (default) - Does not display the LCV when the phone is idle.

1 - Displays the LCV in picture-in-picture (PIP) when the phone is idle.

This parameter applies only when `video.localCameraView.userControl` is set to PerSession or Hidden.

`video.localCameraView.fullScreen.callState`

Set to determine how the local call view (LCV) displays in full screen when the phone is in a call.

0 (default) - Displays the LCV in PIP during a call.

1 - Displays the LCV in full screen during a call.

`video.localCameraView.fullScreen.idleState`

Set to determine how the local call view (LCV) displays in full screen when the phone is idle.

0 (default) - Does not display the LCV when the phone is idle.

1 - Displays the LCV in picture-in-picture (PIP) when the phone is idle.

`video.localCameraView.fullScreen.userControl`

Set to enable users to control how the local call view (LCV) displays during a call.

Persistent (default) - Enables users to access the **Layout** menu and control how the LCV displays before a call.

PerSession - Enables users to access the **Layout** menu and control how the LCV displays before or during a call.

Hidden - Hides the **Layout** menu so that users cannot control how the LCV displays.

`video.vc4Decode.overrunTolerance`

Set the overrun errors per second for video decoder tolerance. If the decoder generates more overrun errors than the number you set, the Poly Trio drops SVC video layer 1 to reduce the decoder load.

0 (default) – Disable tolerance for decoder overrun errors.

0 – 100 overrun errors per second

Video Parameters

Use the following parameters to configure video features on video-capable phones.

`video.allowWithSource`

Restricts sending video codec negotiation in Session Description Protocol (SDP).

0 (default)

0 or 1

`video.enable`

1 (default) - Enables video calling capabilities for outgoing and incoming calls.

0 - Disables video calling capabilities.

`video.autoFullScreen`

0 (default) - Video calls use the full screen layout, only if explicitly selected by the user.

1 - Video calls use the full screen layout by default.

`video.conf.profile`

Sets the video resolution to large window in all layouts.

540p (default)

1080p

720p

360p

240p

180p

`video.dynamicControlMethod`

0 (default)

1 - The first I-Frame request uses the method defined by `video.forceRtcpVideoCodecControl` and subsequent requests alternate between RTCP-FB and SIP INFO.

To set other methods for I-frame requests, refer the parameter `video.forceRtcpVideoCodecControl.`

`video.iFrame.delay`

0 (default)

1 - 10 seconds - Transmits an extra I-frame after the video starts.

You can configure the amount of delay from the start of video until the I-frame is sent up to 10 seconds.

Change causes system to restart or reboot.

`video.iFrame.minPeriod`

Time taken before sending a second I-frame in response to requests from the far end.

2 (default)

1 - 60

`video.iFrame.onPacketLoss`

0 (default)

1 - Transmits an I-frame to the far end when video RTP packet loss occurs.

`video.iFrame.period.onBoard`

Set the I-Frame interval used for the VC4 encoder.

180 (default)

300 maximum

`video.mute.sendCannedVideo`

1 (default) - The Poly Trio system sends a custom image to the far end when you press Stop my video.

0 - The Poly Trio system doesn't send a video to the far end when you press **Stop my video** and displays a no video graphic, by default.

Video Codec Preference Parameters

Use the following video codec parameters to specify video codec preferences.

To disable codecs, set the value to `0`. A value of `1` indicates the codec is the most preferred and has highest priority.

`video.codecPref.H261`

Sets the H.261 payload type.

6 (default)

0 - 8

`video.codecPref.H264`

Sets the H.264 payload type.

4 (default)

0 - 8

`video.codecPref.H263 1998`

Sets the H.263 payload type.

5 (default)

0 - 8

`video.codecPref.H263`

5 (default)

0 - 8

`video.codecPref.H264`

4 (default)

0 - 8

`video.codecPref.H264.packetizationMode0`

Sets the H.264 payload type when packetization mode is set to 0.

5 (default)

0 - 8

`video.codecPref.H264HP`

Sets the H.264 High Profile video codec preference priority.

2 (default)

0 - 8

`video.codecPref.H264HP.packetizationMode0`

Sets the H.264 high profile payload type when packetization mode is set to 0.

3 (default)

0 - 8

`video.codecPref.H264SVC`

`video.codecPref.Xdata`

Sets the Remote Desktop Protocol (RDP) codec preference priority.

7 (default)

0 - 8

1 - Codec has highest priority.

`video.codecPref.XH264UC`

Sets the Microsoft H.264 UC video codec preference priority.

1 (default)

0 - 8

`video.codecPref.XUlpFecUC`

Sets the forward error correction (FEC) codec priority.

8 (default)

0 - 8

Video Profile Parameters for Poly Trio

These settings include a group of low-level video codec parameters.

For most use cases, the default values are appropriate. does not recommend changing the default values unless specifically advised to do so.

`video.profile.H264.payloadType`

Specifies the RTP payload format type for H264/90000 MIME type.

109 (default)

96 to 127

Change causes system to restart or reboot.

`video.profile.H264.packetizationMode0.payloadType`

Sets the H.264 payload type when packetization mode is set to 0.

99 (default)

0 - 127

`video.profile.H264.packetizationMode1.payloadType`

Sets the H.264 payload type when packetization mode is set to 1.

109 (default)

0 - 127

`video.profile.H264.profileLevel`

Specifies the highest profile level within the baseline profile supported in video calls.

4.1 (default)

1, 1b, 1.1, 1.2, 1.3, and 2

Change causes system to restart or reboot.

`video.profile.H264HP.jitterBufferMax`

The largest jitter buffer depth to be supported (in milliseconds).

2000 (default)

533 - 2500 milliseconds

This parameter should be set to the smallest possible value that supports the expected network jitter. Jitter above this size always causes packet loss.

`video.profile.H264HP.jitterBufferMin`

The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time.

150 milliseconds (default)

33 - 1000 milliseconds

Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.

`video.profile.H264HP.jitterBufferShrink`

The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks.

70 milliseconds (default)

33 - 1000 milliseconds

Use smaller values (33 ms) to minimize the delay from trusted networks. Use larger values (1000ms) to minimize packet loss on networks with large jitter (3000 ms).

`video.profile.H264HP.payloadType`

Specifies the RTP payload format type for H264/90000 MIME type.

100 (default)

0 - 127

`video.profile.H264HP.profileLevel`

Specifies the highest profile level within the baseline profile supported in video calls.

4.1 (default)

String (1 - 5 characters)

`video.profile.H264HP.packetizationMode0.payloadType`

Sets the H.264 high profile payload type when packetization mode is set to 0.

113 (default)

0 - 127

`video.profile.Xdata.payloadType`

Specifies the payload type to use in SDP negotiations of the payload used for Skype for Business desktop content sharing.

127 (default)

0 - 127

`video.profile.XH264UC.jitterBufferMax`

The largest supported jitter buffer depth.

2000 (default)

533 - 2500 milliseconds

Jitter above 2500ms always causes packet loss. This parameter should be set to the smallest possible value that supports the network jitter.

`video.profile.XH264UC.jitterBufferMin`

The smallest jitter buffer depth that must be achieved before play out begins for the first time.

150 (default)

33 - 1000 milliseconds

Even if this depth is achieved initially, it may fall and the play out might still continue. This parameter should be set to the smallest possible value, at least two packet payloads, and larger than the expected short term average jitter.

`video.profile.XH264UC.jitterBufferShrink`

Specifies the minimum duration in milliseconds of Real-time Transport Protocol (RTP) packet Rx, with no packet loss to trigger jitter buffer size shrinks.

70 (default)

33 - 1000

Use smaller values (1000 ms) to minimize the delay on known good networks.

`video.profile.XH264UC.mstMode`

Specifies the multi-session transmission packetization mode.

NI-TC (default)

String

The value of NI-TC identifies non-interleaved combined timestamp and CS-DON mode.

This value should not be modified for interoperation with other Skype for Business devices.

`video.profile.XH264UC.payloadType`

Specifies the RTP payload format type for H.264 MIME type.

122 (default)

0 - 127

`video.profile.XUlpFecUC.alwaysOn`

1 (default) - Enable Forward Error Correction during video calls even when it is not needed.

0 - Disable Forward Error Correction.

`video.profile.XUlpFecUC.debug.rxDropBurst`

1 (default)

1 - 100

`video.profile.XUlpFecUC.debug.rxDropOnlyLayer0`

1 (default)

0 or 1

`video.profile.XUlpFecUC.debug.rxDropRate`

0 (default)

0 - 40000

`video.profile.XUlpFecUC.debug.txDropBurst`

1 (default)

1 - 100

`video.profile.XUlpFecUC.debug.txDropRate`

0 (default)

0 - 40000

`video.profile.XUlpFecUC.noLossTurnOffTimeout`

300 (default)

10 - 7200

video.profile.XUlpFecUC.payloadType

123 (default)

0 - 127

video.profile.XUlpFecUC.rxEnabled

1 (default)

0 or 1

video.profile.XUlpFecUC.txEnabled

1 (default)

0 or 1

video.simpleJB.enable

1 (default)

0 or 1

video.simpleJB.lipSyncDelayMs

0 (default)

0 -250 ms

video.simpleJB.timeoutMs

100 ms (default)

0 - 250 ms

video.rtcpbandwidthdetect.enable

0 (default)

1 - Poly Trio 8800 uses an estimated bandwidth value from the RTCP message to control Tx/Rx video bps.

Video Quality Parameters

Use the following parameters to configure quality settings for video calls.

video.quality

The optimal quality for video that is sent in a call or a conference.

motion (default) — For outgoing video that has motion or movement.

sharpness — For outgoing video that has little or no movement.

Note: If you don't select `motion`, moderate to heavy motion can cause the phone to drop some frames.

video.quality.content

motion (default) - For outgoing video that has motion or movement.

sharpness - For outgoing video that has little or no movement.

`video.autoFullScreen`

0 (default) - Video calls only use the full screen layout if it is explicitly selected by the user.

1 - Video calls use the full screen layout by default, such as when a video call is first created or when an audio call transitions to a video call

`video.callRate`

The default call rate (in Kbps) to use when initially negotiating bandwidth for a video call.

2048 (default)

128 - 6144

`video.forceRtcpVideoCodecControl`

0 (default) - RTCP feedback messages depend on a successful SDP negotiation of a=rtcp-fb and are not used if that negotiation is missing.

1 - The phone is forced to send RTCP feedback messages to request fast I-frame updates along with SIP INFO messages for all video calls irrespective of a successful SDP negotiation of a=rtcp-fb.

For an account of all parameter dependencies when setting I-frame requests, refer to the section I-Frames.

`video.maxCallRate`

Sets the maximum call rate that the users can select. The value set on the phone cannot exceed this value. If `video.callRate` exceeds this value, this parameter overrides `video.callRate` and this value is used as the maximum.

4096 (default)

128 - 6144

`video.mute.maxFrameRate`

Set the maximum frame rate that the system will allow for when the system is on video mute.

5 (default)

1 - 60

`video.mute.maxResolution`

Set the maximum resolution (height) that video mute will be encoded to on the system.

180 (default)

Valid values:

- 180
- 240
- 270
- 448
- 540
- 576
- 720