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Poly Voice Software
Poly CCX Business Media Phones Parameter Reference Guide

Contents

Before You Begin

This *Poly CCX Business Media Phones with OpenSIP Parameter Reference Guide* contains overview information for navigating and performing tasks on Poly CCX phones.

This guide contains information for the following Poly products and accessories:

- Poly CCX 350 Business Media Phones
- Poly CCX 400 Business Media Phones
- Poly CCX 500 Business Media Phones
- Poly CCX 505 Business Media Phones
- Poly CCX 600 Business Media Phones
- Poly CCX 700 Business Media Phones

Audience, Purpose, and Required Skills

This guide is intended for beginning users, as well as intermediate and advanced users who want to learn how to use the features available with CCX.

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.
- The Poly Documentation Library provides support documentation for active products, services, and solutions. The documentation displays in responsive HTML5 format so that you can easily access and view installation, configuration, or administration content from any online device.
- The Poly Community provides access to the latest developer and support information. Create an account to access Poly support personnel and participate in developer and support forums. You can find the latest information on hardware, software, and partner solutions topics, share ideas, and solve problems with your colleagues.
- The Poly Partner Network is a program where resellers, distributors, solutions providers, and unified communications providers deliver high-value business solutions that meet critical customer needs, making it easy for you to communicate face-to-face using the applications and devices you use every day.
- Poly Services help your business succeed and get the most out of your investment through the benefits of collaboration. Enhance collaboration for your employees by accessing Poly service solutions, including Support Services, Managed Services, Professional Services, and Training Services.
- 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.

Privacy Policy

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Getting Started

This parameter reference guide lists the parameters available to provision and configure Poly CCX Business Media Phones.

You can use the configuration parameters described in this guide to provision single or multiple phones.

For more information on provisioning Poly phones, see the Poly CCX Business Media Phones with OpenSIP Provisioning Guide.

Parameter List Conventions

For each feature, Poly provides a list of parameters in XML that you can use to configure feature settings.

This guide documents parameters using parameter lists. Be sure to familiarize yourself with basic XML and parameter list conventions to successfully change configurations.

Using XML

Poly parameters are attributes of XML elements. Element names don't affect the behavior of parameters or operation of your phone, and you can customize as needed.

When configuring the parameters as XML, you must enter parameter names as attributes of a well-formed XML syntax. You can organize parameters into any well-formed XML element structure.

A **parameter="value"** pair is equivalent to an XML **attribute="value"** pair. For example:

```
<element1>
       <element2 feature.acousticFenceUI.enabled="1" />
</element1>
```

Understanding Parameter Lists

The following describes a general convention for details you can find in parameter lists. Parameter details can vary depending on the complexity of the parameter.

`parameter.name`

A parameter's description, applicability, or dependencies, as needed.

The parameter's permitted values, the default value, and the value's unit of measure, such as seconds, Hz, or dB.

An indication when a change in a parameter's value causes a phone restart or reboot.

Note: A note that highlights critical information you need to know.

Parameter List Template and Examples

Parameter details can vary depending on the complexity of the parameter.

The following template shows the general parameter list conventions and details.

`parameter.name`

A parameter's description, applicability, or dependencies, as needed.

The parameter's permitted values, the default value, and the value's unit of measure, such as seconds, Hz, or dB.

An indication when a change in a parameter's value causes a phone restart or reboot.

Note: A note that highlights critical information you need to know.

The following sample parameter lists show a few example parameters and some XML representations showing how to use them.

`feature.acousticFenceUI.enabled`

0 (default) - Hide the Acoustic Fence configuration setting on the phone.

1 - Display the Acoustic Fence configuration setting on the phone.

Change causes system to reboot or restart.

XML Representation

```
<element feature.acousticFenceUI.enabled="1" />
```

`reg.x.callsPerLineKey`

Set the maximum number of concurrent calls for a single registration *x* that you specify. 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 global parameter `call.callsPerLineKey`.

24 (default)

1 - 24

1 - 8

XML Representation

```
<registration
    reg.1.callsPerLineKey="3"
    reg.2.callsPerLineKey="1"
    reg.3.callsPerLineKey="1"
/>
```

Audio Parameters

Use the following parameters to configure audio settings on your phones.

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 - Alerts and sound effects play through the 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.

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.G7221.24kbps

0 (default)

voice.codecPref.G7221_C.24kbps

0 (default)

voice.codecPref.G7221.32kbps

5 (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)

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)

Context Sensitive Volume Control Parameters

The following list includes the parameters you can use to configure Context Sensitive Volume Control.

`voice.volume.persist.bluetooth.headset`

0 (default) - The Bluetooth headset volume does not persist between calls and resets to a nominal level each new call.

1 - The volume for each call is the same as the previous call.

`voice.volume.persist.handset`

0 (default) - The handset volume automatically resets to a nominal level after each call.

1 - The volume for each call is the same as the previous call.

`voice.volume.persist.handsfree`

1 (default) - The speakerphone volume at the end of a call persists between calls.

0 - The speakerphone volume does not persist between calls and resets to a nominal level each new call.

`voice.volume.persist.usb.handsfree`

0 (default) - Does not use USB headset automatically for calls.

1 - Uses the USB headset automatically for all calls.

`voice.volume.persist.usbHeadset`

0 (default) - The USB headset volume does not persist between calls and resets to a nominal level each new call.

1 - The USB headset volume at the end of a call persists between calls.

Distinctive Ringtone Parameters

The following list includes the parameters you can use to configure distinctive ringtones for a line, contact, or type of call.

`voIpProt.SIP.alertInfo.x.class`

Specify a ringtone for a 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 to parameters as specified (x=1, 2, ..., 22) and if a match is found, the behavior described in the corresponding ring class is applied.

default (default)

See the list of ring classes in Ringtone Parameters.

`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

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

Polycom Acoustic Fence Parameters

The following list includes the parameters you can use to configure Polycom Acoustic Fence noise suppression feature.

Note: When Acoustic Fence is enabled, Polycom recommends setting the parameter `video.disableAFOnFullScreen` to **1** to improve the phone's performance when the Polycom EagleEye Mini USB camera is connected to a phone.

`feature.acousticFenceUI.enabled`

0 (default) - Hide display of the Acoustic Fence Configuration setting on the phone.

1 - Displays the Acoustic Fence Configuration setting on the phone.

`voice.ns.hd.enable`

Enables or disables noise suppression for headsets.

0 (default) – Disabled

1 – Enabled

`voice.ns.hd.enhanced`

Enables or disables Acoustic Fence noise suppression for headsets.

0 (default) – Disabled

1 – Enabled

`voice.ns.hd.nonStationaryThresh`

Sets the Acoustic Fence noise suppression threshold for headsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. High values can suppress the speaker's voice and impact far-end audio quality.

1 to 10

8 (default)

`voice.ns.hs.enable`

Enables or disables noise suppression for handsets.

0 - Disabled

1 (default) - Enabled

`voice.ns.hs.enhanced`

Enables or disables Acoustic Fence noise suppression for handsets.

1 - Enabled

0 (default) - Disabled

`voice.ns.hs.nonStationaryThresh`

Sets the Acoustic Fence noise suppression threshold for handsets. A lower value allows more background sound to enter, and a higher value suppresses background noise. High values can suppress the speaker's voice and impact far-end audio quality.

1 to 10

8 (default)

`video.disableAFOnFullScreen`

Allows or disallows the phone to dynamically deactivate Acoustic Fence when the user changes the view to full screen mode while using a handset in a video call.

0 (default) - Disallowed

1 - Allowed

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>.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.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.callProg.secondaryDialTone.name`

1-255

`se.pat.callProg.secondaryDialTone.inst.1.type`

0-255

`se.pat.callProg.secondaryDialTone.inst.1.value`

0-50

`se.pat.callProg.secondaryDialTone.inst.1.atte`

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

UTF-8 encoded string

`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

`se.pat.callProg.stutter.inst.1.type`

chord (1-2) (default) - Type of sound effect

NULL (3-8) (default)

sampled - Sampled audio file number

silence - Silence duration in milliseconds branch - Number of instructions to advance

se.pat.callProg.stutter.inst.1.value

stutterLong (1) (default)

dialTone (2) (default)

NULL (3-8) (default)

se.pat.misc.callParkBLFAudioNotification.inst.x.type

Specify the sound effect type to play the audio tone.

Null (default)

chord

se.pat.misc.callParkBLFAudioNotification.inst.x.value

Specify the file to play the audio tone.

Null (default)

cs7

cs4

se.pat.misc.callParkBLFAudioNotification.inst.x.param

Specify the duration for how long the tone should play.

0 (default)

5000 ms

se.pat.misc.callParkBLFAudioNotification.inst.x.atten

Specify the tone attenuation.

0 (default)

-1000 Hz

5000 Hz

se.pat.misc.callParkBLFReminderTone.inst.x.type

Specify the sound effect type to play the audio tone.

Null (default)

chord

se.pat.misc.callParkBLFReminderTone.inst.x.value

Specify the file to play the audio tone.

Null (default)

cs3

cs4

cs6

`se.pat.misc.callParkBLFReminderTone.inst.x.param`

Specify the duration for how long the tone should play.

0 (default)

5000 ms

`se.pat.misc.callParkBLFReminderTone.inst.x.atten`

Specify the tone attenuation.

0 (default)

-1000 Hz

5000 Hz

Supported Ring Classes

Ring classes help you define which ringtone to play for certain function notifications.

The phones support the following ring classes:

- default
- visual
- answerMute
- autoAnswer
- ringAnswerMute
- ringAutoAnswer
- internal
- external
- emergency
- precedence
- splash
- custom<y> where y is 1 to 17.

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.

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

Call Control Parameters

Use the following parameters to configure call controls including microphone mute, call recording, conference features, and call hold.

Access URL in SIP Messages Parameters

You can configure the retrieval method for web content and enable users to choose to retrieve web content using either Active or Passive mode.

If your call server supports access URLs, you can also specify active or passive retrieval in the SIP header. If parameters in the SIP signal conflict with the file configuration, parameters in the SIP signaling take precedence.

You can also enable new web content to be added to the Settings menu on the phone, and users can set the default display mode for individual URLs to active or passive from the phone's menu.

`mb.ssawc.enabled`

0 (default) - Spontaneous display of web content is disabled.

1 - Spontaneous web content display is enabled.

`mb.ssawc.call.mode`

passive (default) - Web content is displayed only when requested by the user. Passive mode is recommended when the microbrowser is used for other applications. When passive mode is enabled, an icon displays beside a call appearance indicating that web content is available, and the user can press the **Web** softkey to view the content.

Active - Web content is retrieved spontaneously and displayed immediately.

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.

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

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.

Busy Lamp Field Configuration Parameters

The maximum number of BLF entries for phones is 50.

In the following list, x in a parameter is the number of the BLF entry in the list. If you are using static BLF, you need to configure the number of each entry.

`attendant.behaviors.automata.pickupOnBusy`

Set to allow an automata resource (static BLF) pickup on a busy BLF Resource.

1 (default) - Allows pick up on a Busy Lamp Field resource.

0 - Doesn't allow pick up on a Busy Lamp Field resource.

`attendant.behaviors.display.remoteCallerID.automata`

These parameters depend on the value set for the parameter `attendant.resourceList.x.type` . If the parameter `attendant.resourceList.x.type` is set to automata, use the parameter `attendant.behaviors.display.remoteCallerID.automata` .

1 (default) - Automata remote party caller ID information is presented to the attendant.

0 - The string `unknown` is substituted for both name and number information.

`attendant.behaviors.display.remoteCallerID.normal`

These parameters depend on the value set for the parameter `attendant.resourceList.x.type` . If the parameter `attendant.resourceList.x.type` is set to normal, use the parameter `attendant.behaviors.display.remoteCallerID.normal` .

1 (default) - Normal remote party caller ID information is presented to the attendant.

0 - The string `unknown` is substituted for both name and number information.

`attendant.behaviors.display.spontaneousCallAppearances.automata`

Specifies how call appearances display on the attendant phone when the configured attendant resource is the automata type.

Note: The values of these call appearance parameters depend on the values applied to `attendant.resourceList.x.type`.

0 (default) - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter.

1 - The automata call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played).

`attendant.behaviors.display.spontaneousCallAppearances.normal`

Specifies how call appearances display on the attendant phone when the configured attendant is the normal type.

Note: The values of these call appearance parameters depend on the values applied to `attendant.resourceList.x.type` .

1 (default) - The normal call appearance is spontaneously presented to the attendant when calls are alerting on a monitored resource (and a ring tone is played).

0 - The call appearance is not spontaneously presented to the attendant. The information displayed after a press and hold of a resource's line key is unchanged by this parameter.

`attendant.behaviors.preserveCallerIDOnPickup`

Preserve caller ID information for monitored lines when a call is picked up by an attendant.

0 (default) - The caller ID is not preserved for monitored lines when a call is picked up by an attendant.

1 - The caller ID is preserved for monitored lines when a call is picked up by an attendant.

`attendant.resourceList.x.display.spontaneousCallAppearances`

Specifies spontaneous call appearance property for an incoming call for this specific BLF.

This parameter will override the phone level configuration parameters `attendant.behaviors.display.spontaneousCallAppearances.normal` and `attendant.behaviors.display.spontaneousCallAppearances.automata` to show or hide the call appearance for any BLF incoming call.

Auto (default) – This value will use phone-level configuration depending on the BLF value set of parameters.

Show – This value will override phone-level configuration and will show the call appearance.

Hide – This value will override phone-level configuration and will hide the call appearance.

`attendant.callWaiting.enable`

0 (default) - The phone does not generate acoustic indication of call waiting for attendant calls monitored by BLF.

1 - The phone generates an acoustic indication of call waiting for attendant calls monitored by BLF.

`attendant.callWaiting.ring`

This parameter is valid only if `attendant.callWaiting.enable` is set to 1. Specifies the ring type to be used for notifying an attendant call if there is an active call already present on the phone.

Silent - No acoustic indication is provided.

beep - Beep tone is played when there is an active call on the phone and an attendant call is received.

ring - Ring tone configured in attendant.ringType is used to alert the user when there is an active call on the phone and an attendant call is received.

`attendant.reg`

Specifies the line registration to use to subscribe to the BLF resource. The index of the registration is used to send a SUBSCRIBE to the list SIP URI specified in `attendant.uri` . For example, `attendant.reg="2"` means the second registration is used.

1 (default)

Permitted value is any positive integer.

`attendant.resourceList.x.address`

The user referenced by `attendant.reg` subscribes to this URI for dialog. If a user part is present, the phone subscribes to a sip URI constructed from the user part and domain of the user referenced by `attendant.reg` . Transport for BLF subscriptions may be modified by including a transport parameter into the subscription address. For example: `sip:blf12345@domain.com;transport=tcp`

Permitted value is a string that constitutes a valid SIP URI (`sip:6416@polycom.com`) or contains the user part of a SIP URI (`6416`).

Null (default)

`attendant.resourceList.x.bargeInMode`

Enable or disable barge-in and choose the default barge-in mode. This parameter applies to the Alcatel-Lucent CTS only.

Null (default) - The Barge In feature is disabled.

All - Press and hold the BLF line to display all barge-in options. Quick press to barge-in as Normal.

Normal - Barge-in plays an audio tone to indicate the arrival of a new participant to the call and all call participants can interact.

Listen - The user barging in can listen on the call only. Their outbound audio is not transmitted to either party.

Whisper - The user barging in can hear all parties but their audio is only transmitted to the user they are monitoring.

`attendant.resourceList.x.callAddress`

Use this parameter when the call signaling address for the BLF line is different than the address set by `attendant.resourceList.x.address`.

Null (default)

Maximum 255 characters

attendant.resourceList.x.label

The text label displays adjacent to the associated line key for the BLF line. If set to Null, the label is derived from the user part of `attendant.resourceList.x.address` .

Null (default)

Permitted value is a UTF-8 encoded string.

attendant.resourceList.x.proceedingIsRecipient

A flag to determine if pressing the associated line key for the monitored user picks up the call.

1 - If the call server does not support inclusion of the direction attribute in its dialog XML.

0 (default)

attendant.resourceList.x.requestSilentBargeIn

0 (default) - A tone plays when a contact barges in on a call.

1 - No tone is played when a contact barges in on a call.

attendant.resourceList.x.type

The type of resource being monitored and the default action to perform when pressing the line key adjacent to monitored user x.

normal (default) -The default action is to initiate a call if the user is idle or busy and to perform a directed call pickup if the user is ringing. Any active calls are first placed on hold.

Note: The value `normal` applies the call appearance setting `attendant.behaviors.display.*.normal` .

automata -The default action is to perform a park/blind transfer of any currently active call. If there is no active call and the monitored user is ringing/busy, an attempt to perform a directed call pickup/park retrieval is made.

Note: That the value `automata` applies the call appearance setting `attendant.behaviors.display.*.automata`.

attendant.restrictPickup

0 (default) - The attendant can pick up calls to monitored users while they show as ringing.

1 - The attendant cannot pick up the monitored call.

attendant.ringType

The ringtone that plays when a BLF dialog is in the offering state.

ringer1 (default)

ringer1 - ringer24

attendant.uri

The list SIP URI on the server. If this is just a user part, the URI is constructed with the server hostname/IP.

Note: If this parameter is set, then the individually addressed users configured by `attendant.resourceList` and `attendant.behaviors` are ignored.

Null (default)

Strings are permitted.

`call.directedCallPickupMethod`

Specifies how the phone performs a directed call pick-up from a BLF contact.

legacy (default) - Indicates that the phone uses the method specified in `call.directedCallPickupString` .

native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header).

`call.directedCallPickupString`

The star code to initiate a directed call pickup.

*97 (default)

Note: The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.

`call.parkedCallRetrieveMethod`

The method the phone uses to retrieve a BLF resource's call which has dialog state confirmed.

legacy (default) - Indicates that the phone uses the method specified in `call.parkedCallRetrieveString` .

native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header).

`call.parkedCallRetrieveString`

The star code that initiates retrieval of a parked call.

Null (default)

Permitted values are star codes.

`voipPort.SIP.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 target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for retrieval.

`voIpProt.SIP.strictReplacesHeader`

This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.

1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when call.directedCallPickupMethod is configured as native.

0 - Call pick-up requires a call id only.

`voIpProt.SIP.useLocalTargetUriforLegacyPickup`

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 target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for pickup or retrieval.

`attendant.callAction`

Specify the call action behavior for an Active call.

Dial-Pick up (default) – An active call goes on hold and dials to monitor line or picks the incoming call on monitor line when you short press the monitored line keys.

Blind – Blind transfer an active call on the monitored line keys.

Park – Parks an active call on the monitored line keys. If there is already a parked call on a monitored line then it will retrieve the parked call.

`attendant.callActionMenu.enabled`

This parameter is configured to get the **Attendant Call Action** menu on the phone when you configure dynamic BLF on the phone.

0 (default) – **Attendant Call Action** menu will not appear on the phone.

1 - **Attendant Call Action** menu will appear on the phone.

`attendant.displayHoldState.enable`

Specifies the control of the display on the phone for BLF hold state.

0 (default) - The phone displays a busy state.

1 - The phone displays a hold state.

Note : This parameter is only applicable to static BLF

`attendant.resourceList.x.hold.ringer`

The ringtone that plays on the phone when BLF is in a hold state.

The parameter depends on the value set for the parameter attendant.displayHoldState.enabled . If the parameter attendant.displayHoldState.enable is set to 1, use the parameter attendant.resourceList.x.hold.ringer

Triplet (default) – Specifies the ringtone name for the parameter ringer11.

Ringtone for BLF Hold should play for only 10 sec.

`attendant.resourceList.x.groupPickUp.prefix`

`attendant.resourceList.x.ringType`

This parameter is applicable to Static BLF.

Specifies incoming ringtone for each static BLF line

defaultAll (default) – Specifies the ringtone type ring for the ringtone name.

ringer1 - ringer24.

If no ringtone is configured for any static BLF line, then phone level incoming ringtone defined with `attendant.ringType` parameter will be played.

Call Application Switching Parameters on CCX Phones

Use the following parameters to configure call application switching.

`apps.android.appSwitcher.enabled`

0 (default) – Call application switching is disabled.

1 – App Switcher icon appears on the on the navigation bar, enabling users to switch call applications.

Change causes the system to restart or reboot.

`apps.android.appSwitcher.MSTeams.enabled`

0 (default) – Microsoft Teams is not accessible via app switching.

1 – Microsoft Teams is accessible via app switching.

`apps.android.appSwitcher.ZoomPhone.enabled`

0 (default) – Zoom Phone is not accessible via app switching.

1 – Zoom Phone is accessible via app switching.

`apps.android.statusBar.UCS.enabled`

If your phone is configured with a third-party call application base profile, enable this parameter to switch to the Poly OpenSIP call application.

0 (default) - Poly OpenSIP is not accessible via app switching.

1 - Poly OpenSIP is accessible via app switching.

`apps.android.statusBar.enabled`

1 (default) - Enables the **Poly Control Panel**.

0 - Disables the **Poly Control Panel**.

`apps.android.statusBar.Bluetooth.enabled`

If `feature.bluetooth.enabled="1"`, enable `apps.android.statusBar.Bluetooth.enabled` to enable users to toggle Bluetooth functionality on and off from the **Poly Control Panel**.

0 (default) - Bluetooth doesn't appear in the **Poly Control Panel**.

1 - Users can toggle Bluetooth on and off from the **Poly Control Panel** while the system is in Hub mode.

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.

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 Hold Timer Parameter

Use the following parameters to configure how the call hold timer displays on the phone's local interface.

`up.holdTimerDisplay.enable`

0 (default) – Hold Timer will not display.

1 – Hold Timer will display.

`up.timerDisplayInSeconds`

0 (default) – The call timer and call hold timer are displayed in "hh:mm:ss" notation.

1 – Call timer is displayed in 5-digit second notation as "sssss" notations, and the call hold timer is displayed in 4-digit second notation as "ssss" notations.

Calling Line Identification Parameters

`voIpProt.SIP.CID.sourcePreference`

Specify the priority order for the sources of caller ID information. The headers can be in any order.

Null (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order.

From,P-Asserted-Identity, Remote-Party-ID

P-Asserted-Identity,From,Remote-Party-ID

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

Note: By default callee and caller will take identity order from `voIpProt.SIP.CID.sourcePreference`.

If `voIpProt.SIP.Caller.SourcePreference` or `voIpProt.SIP.Callee.SourcePreference` are configured then the order set by `voIpProt.SIP.CID.sourcePreference` is ignored.

`voIpProt.SIP.caller.sourcePreference`

Set the priority order to display the caller's identity for incoming calls.

Null (default)

0-120

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

String

`voIpProt.SIP.callee.sourcePreference`

Set the priority order to display the callee's identity for outgoing calls.

Null (default)

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

String

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 Park and Retrieve Parameters

Use the parameter below to configure Call Park and Retrieve.

`call.parkedCallString`

The star code to initiate the call park.

String

*68

Change causes system to restart or reboot.

`up.simplifiedPickup`

Enable/disable simplified pick up for directed, group, and call park pickups.

0 (default) - Disabled simplified pick up.

1 - Enable simplified pick up.

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 Recording Modes

Set the call recording modes on the BroadSoft BroadWorks R20 server using the following call recording modes:

- **Never Mode** – Call recording is never initiated and the phone never displays call recording soft keys.
- **Always Mode** – The entire incoming or outgoing call is recorded and no control options are available to users. During active calls, the phone displays a Record symbol. Call recording stops when the call ends and the call is stored on the server.
- **Always with Pause/Resume Support Mode** – Call recording starts automatically when the call connects and the Pause and Resume soft keys are available. The phone display indicates the status of the call recording state. Call recording stops when the call ends and the recorded part of the call is stored on the server.
- **On Demand Mode** – Call recording starts on the server when the call connects, but the recorded file is not saved until the user initiates the recording. When the user presses the Start soft key, the recording is saved to the server and the phone displays the Pause and Resume soft keys.

- **On Demand Mode with User-Initiated Start Mode** – Call recording does not begin automatically and a Record soft key displays. If users want to record an active call, they need to press Record > Start to start recording and save the recording to the server. While recording, the phone displays the Pause, Resume, and Stop soft keys.
- **Recording two separate calls and creating a conference** – This mode enables users to record two participants as separate call sessions when connected in a conference call. The server stores the conference call as two separate recording sessions.

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.

Centralized Call Recording Parameters

You must enable this feature on the BroadSoft BroadWorks r20 server and on the phones using the configuration parameters listed in the following list.

`voIpProt.SIP.serverFeatureControl.callRecording`

0 (default) - The BroadSoft BroadWorks v20 call recording feature for multiple phones is disabled.

1 - The BroadSoft BroadWorks v20 call recording feature for multiple phones is enabled.

Change causes system to restart or reboot.

`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.

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 - 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.

Directed Call Pickup Parameters

You can configure Directed Call Pickup using parameters in this section.

The parameters you use to configure this feature depends on your call server. To enable or disable this feature for Sylantro call servers, set the parameter `feature.directedCallPickup.enabled` to 1.

To configure this feature for all other call servers, use the following parameters:

- `call.directedCallPickupMethod`
- `call.directedCallPickupString`

Note that the pickup string can be different for different call servers, so check with your call server provider if you configure legacy mode for directed call pickup.

The following list includes the configuration parameters for the directed call pick-up feature.

`feature.directedCallPickup.enabled`

0 (default) - Disables the directed call pickup feature.

1 - Enables the directed call pickup feature.

Change causes system to restart or reboot.

`call.directedCallPickupMethod`

Specifies how the phone performs a directed call pick-up from a BLF contact.

legacy - Indicates that the phone uses the method specified in the `call.directedCallPickupString` parameter.

native - Indicates that the phone uses a native protocol method (in this case SIP INVITE with the Replaces header.

`call.directedCallPickupString`

The star code to initiate a directed call pickup.

*97

Note: The default value supports the BroadWorks calls server only. You must change the value if your organization uses a different call server.

`voIpProt.SIP.strictReplacesHeader`

This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.

1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when `call.directedCallPickupMethod` is configured as native.

0 - Call pick-up requires a call id only.

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.

`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

`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.

`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.

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 to the URL to request the location information from the server. For example, https://host.domain.com/held/request.

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.

`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 to the URL to request the location information from the server. For example, https://host.domain.com/held/request.

NULL (default)

0-255

`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.

Group Call Pickup Parameters

This feature requires support from a SIP server and setup of this feature depends on the SIP server.

For example, while some SIP servers implement group call pick-up using a particular star-code sequence, others implement the feature using network signaling.

`feature.groupCallPickup.enabled`

Enable or disable SIP-B Group Call Pickup.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

Group Paging with Call Application Switching Parameters

Use the following parameters to configure group paging for use with call application switching.

`apps.android.appSwitcher.Paging.enabled`

0 (default) - Group paging with call application switching is disabled.

1 - Group paging with call application switching is enabled.

Note: Before setting this parameter, set `apps.android.appSwitcher.enabled="1"`.

`apps.android.appSwitcher.Paging.useDefaultChannel`

1 (default) - When a user selects group paging on the Poly Control Panel, the phone opens group 1, or the paging channel defined by `ptt.defaultChannel`.

0 - When a user selects group paging on the Poly Control Panel, the phone displays the **Group Page List**.

Note: Before setting this parameter, set `apps.android.appSwitcher.enabled="1"` and `apps.android.appSwitcher.Paging.enabled="1"`.

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.

Instant Messaging Parameter

Use this parameter to enable or disable instant messaging on the phone.

`feature.messaging.enabled`

0 (default) - Disable instant messaging.

1 - Enable instant messaging.

Change causes system to restart or reboot.

Key System Emulation Parameters

Use the following parameters to configure the KSE feature on supported phones.

`attendant.keylineEmulation.enabled`

0 (default) - Disables the KSE feature.

1 - Enables the KSE feature.

`attendant.keylineEmulation.showParkedCallerId`

1 (default) - The display name of the parked caller (if available) is shown for a line whenever a call is parked.

0 - The display name of the parked caller is generated from BLF dialog resource list.

`feature.enhancedCallPark.allowBLFAudioNotification`

Allow call park audio notification on BLF monitored lines.

0 (default) - Disabled

1 - Enabled

This parameter is applicable only if KSE is enabled.

`attendant.callParkBLFReminder.StartDelay`

Time in seconds before the first reminder tone is played.

0 (default) - No reminder tone is played for calls parked by remote phones.

0 - 3600

This parameter is applicable only if KSE is enabled.

`attendant.callParkBLFReminder.RepeatTime`

Time in seconds between two reminder tones.

0 (default) - No repeat reminder tone is played.

When `attendant.callParkBLFReminder.StartDelay` parameter is not set to 0 and `attendant.callParkBLFReminder.RepeatTime` parameter is set to 0, a single start reminder tone is played.

0-3600

This parameter is applicable only if KSE is enabled.

Last Call Return Parameters

The last call return string value that you enter for parameter `call.lastCallReturnString` depends on the call server you use. Consult with your call server provider for the last call return string.

`feature.lastCallReturn.enabled`

0 (default) - Disable last call return feature.

1 - Enable last call return.

`call.lastCallReturnString`

Specify the string sent to the server when the user selects the last call return action. The string is usually a star code.

*69 (default)

string - maximum 32 characters

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 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

`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 —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`.

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.

`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.

`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.

Multiple Call Appearance Parameters

Use the parameters in the following table 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 that this parameter can be overridden by the per-registration parameter `reg.x.callsPerLineKey` .

24 (default)

1 - 24

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 `call.callsPerLineKey` .

24 (default)

1-24

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.

Multiple Line Registrations Parameters

Each registration can be mapped to one or more line keys, however, a line key can be used for only one registration.

The maximum number of call appearances you can set varies by phone model.

`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.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.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.address`

The user part (for example, 1002) or the user and the host part (for example, `1002@polycom.com`) of the registration SIP URI.

Null (default)

string address

`reg.x.advancedConference.maxParticipants`

Sets the maximum number of participants allowed in a push to conference for advanced conference calls. The number of participants configured must match the number of participants allowed on the ALU CTS.

3 (default)

0 - 25

`reg.x.advancedConference.pushToConference`

0 (default) - Disable push-to-conference functionality.

1 - Enable push-to-conference functionality.

`reg.x.advancedConference.subscribeForConfEvents`

1 (default) - Conference participants to receive notifications for conference events is enabled.

0 - Conference participants to receive notifications for conference events is disabled.

`reg.x.advancedConference.subscribeForConfEventsOnCCPE`

1 (default) - Enable the conference host to receive notifications for conference events.

0 - Disable the conference host to receive notifications for conference events.

`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.useLoginCredentials`

0 - (default) The Login credentials aren't used for authentication to the server on registration x.

1 - The login credentials are used for authentication to the server.

`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.bargeInEnabled`

0 (default) - barge-in is disabled for line x.

1 - barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls).

`reg.x.bridgeInEnabled`

0 (default) - Bridge In feature is disabled.

1 - Bridge In feature is enabled.

`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.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 `call.callsPerLineKey`.

24 (default)

1 - 24

`reg.x.displayName`

The display name used in SIP signaling.

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.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.enhancedCallPark.enabled`

0 (default) - To disable the BroadWorks Enhanced Call Park feature.

1 - To enable the BroadWorks Enhanced Call Park feature.

`reg.x.filterReflectedBlaDialogs`

1 (default) - bridged line appearance NOTIFY messages are ignored.

0 - bridged line appearance NOTIFY messages isn’t 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` isn’t used

`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` isn’t 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.busy.status`

0 (default) - Incoming calls that receive a busy signal isn’t 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.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.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 aren’t 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.fwd.noanswer.status`

0 (default) - The calls aren't 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`

1 - The phone sends sip.instance in the REGISTER request.

0 (default) - The phone doesn't send sip.instance in the REGISTER request.

`reg.x.gruu`

Specify if the phone sends sip.instance in the REGISTER request.

0 (default) - Disabled

1 - Enabled

`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 isn't added to the route header.

`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.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 doesn't 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`.

- The following examples show labels for line 1 on a phone with user registration 1234, where `reg.x.linekeys="2"`:
  - If no label is configured for registration, the labels are “1_1234” and “2_1234”.

`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 doesn’t 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.lineAddress`

The line extension for a shared line. This parameter applies to private lines and BroadSoft call park and retrieve. If there’s 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)

48

`reg.x.lisdisclaimer`

This parameter sets the value of the location policy disclaimer. For example, the disclaimer may be “Warning: If you don’t 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 doesn’t override `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 doesn’t send full audio and video capabilities after resuming a held call.

`reg.x.offerFullCodecListUponHold`

0 (default) - The phone doesn't send full audio and video capabilities after a hold call.

1 - The phone sends full audio and video capabilities after a hold 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 doesn't 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 parameter overrides `reg.x.server.y.failOver.reRegisterOn`.

0 (default) - The phone doesn'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.path`

0 (Default) - The path extension header field in the Register request message isn't 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.protocol.SIP`

1 (default) - SIP signaling is enabled for this registration.

0 - SIP signaling isn't enabled for this registration.

`reg.x.proxyRequire`

Null (default) - No Proxy-Require is sent.

string - Needs to be entered in the Proxy-Require header.

`reg.x.regevent`

0 (default) - The phone isn't 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 doesn't reject the call.

1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code.

`reg.1.ringdelay`

Set a timer in seconds to delay ringing on an incoming call. When set, the timer withholds all visual and audible information from the user until the time elapses.

0 (default) - No delay timer is set.

0 to 75 seconds - Set the timer to any length of time between these two values.

`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.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.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 host name - 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.

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

`reregisterg.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 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 Poly Phones - Technical Bulletin 5844* at Poly Support.

`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 `reg.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.specialInterop

Specify the server-specific feature set for the line registration.

Standard (Default)

All other phones:

Standard

GENBAND

ALU-CTS

ocs2007r2

lcs2005

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

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.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.

`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.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`

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.

`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.

`reg.x.serverFeatureControl.dnd`

This parameter overrides `voIpProt.SIP.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.

Change causes system to restart or reboot.

`reg.x.serverFeatureControl.localProcessing.cf`

This parameter overrides `voIpProt.SIP.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.

`reg.x.serverFeatureControl.localProcessing.cf`

This parameter overrides `voIpProt.SIP.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.

`reg.x.serverFeatureControl.localProcessing.dnd`

This parameter overrides `voIpProt.SIP.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.

`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.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`

This parameter overrides `sec.srtp.simplifiedBestEffort`.

0 (default) - SRTP negotiation compliant with Microsoft Session Description Protocol Version 2.0 Extensions is not supported.

1 - SRTP negotiation compliant with Microsoft Session Description Protocol Version 2.0 Extensions is supported.

`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.

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.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.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.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.useCompleteUriForRetrieve`

This parameter overrides `voipPort.SIP.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.

`voipProt.server.x.address`

The IP address or hostname and port of a SIP server that accepts registrations. Multiple servers can be listed starting with x=1 to 4 for fault tolerance.

Null (default), IP address, or hostname

`voIpProt.server.x.expires`

The phone's requested registration period in seconds.

3600 (default)

positive integer, minimum 10

The period negotiated with the server may be different. The phone attempts to re-register at the beginning of the overlap period. For example, if expires="300" and overlap="5", the phone re-registers after 295 seconds (300-5).

`voIpProt.server.x.expires`

The phone's requested registration period in seconds. Note: The period negotiated with the server may be different. The phone will attempt to re-register at the beginning of the `overlap` period.

3600 (default)

positive integer, minimum 10

`voIpProt.server.x.expires.lineSeize`

Requested line-seize subscription period.

30 (default)

positive integer, minimum 10

`voIpProt.server.x.expires.lineSeize`

Requested line-seize subscription period.

30 (default)

positive integer, minimum 0 was 10

`voIpProt.server.x.expires.overlap`

The number of seconds before the expiration time returned by server x at which the phone should try to re-register. If the server value is less than the configured overlap value, the phone tries to re-register at half the expiration time returned by the server.

60 (default)

5 to 65536

`voIpProt.server.x.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 will try 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

`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 to. Values between 1 and 59 result in a timeout of 60 and 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 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 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 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 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.

`voIpProt.server.x.port`

The port of the server that specifies registrations.

0 (default) - If 0, the port used depends on `voIpProt.server.x.transport.`

1 to 65535

`voIpProt.server.x.protocol.SIP`

1 (default) - Server is a SIP proxy/registrar

0 - If set to 0, and the server is confirmed to be a SIP server, then the value is assumed to be 1.

`voIpProt.server.x.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.

For more information, see *Technical Bulletin 5844: SIP Server Fallback Enhancements on Poly Phones*.

`voIpProt.server.x.registerRetry.baseTimeOut`

The base time period to wait before a registration retry. Used in conjunction with `voIpProt.server.x.registerRetry.maxTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.

If both parameters `voIpProt.server.x.registerRetry.baseTimeOut` and `reg.x.server.y.registerRetry.baseTimeOut` are set, the value of `reg.x.server.y.registerRetry.baseTimeOut` takes precedence.

60 - (default)

10 - 120

`voIpProt.server.x.registerRetry.maxTimeOut`

The maximum time period to wait before a registration retry. Used in conjunction with `voIpProt.server.x.registerRetry.maxTimeOut` to determine how long to wait. The algorithm is defined in RFC 5626.

If both parameters `voIpProt.server.x.registerRetry.maxTimeOut` and `reg.x.server.y.registerRetry.maxTimeOut` are set, the value of `reg.x.server.y.registerRetry.maxTimeOut` takes precedence.

60 - (default)

10 - 1800

`voIpProt.server.x.retryMaxCount`

The number of retries that will be attempted before moving to the next available server.

3 (default)

0 to 20 - If set to 0, 3 is used.

`voIpProt.server.x.retryTimeOut`

0 (default) - Use standard RFC 3261 signaling retry behavior.

0 to 65535 - The amount of time (in milliseconds) to wait between retries.

`voIpProt.server.x.specialInterop`

Enables server-specific features for all registrations.

Standard (default)

All other phones =

Standard

GENBAND

GENBAND-A2

ALU-CTS

DT

ocs2007r2

lcs2005

`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

`voIpProt.server.x.transport`

The transport method the phone uses to communicate with the SIP server.

Null or DNSnaptr (default) - If `voIpProt.server.x.address` is a hostname and `voIpProt.server.x.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 `voIpProt.server.x.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 will be used.

TLS - If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.

TCPOnly - Only TCP will be used.

`voIpProt.server.x.useOutboundProxy`

1 (default) - Enables to use the outbound proxy specified in `voIpProt.SIP.outboundProxy.address` for server x.

0 - Enables not to use the outbound proxy specified in `voIpProt.SIP.outboundProxy.address` for server x.

`voIpProt.SIP.acd.signalingMethod`

0 (default) - The 'SIP-B' signaling is supported. (This is the older ACD functionality.)

1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)

Change causes system to restart or reboot.

`voIpProt.SIP.acd.signalingMethod`

0 (default) - The 'SIP-B' signaling is supported. (This is the older ACD functionality.)

1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)

Change causes system to restart or reboot.

`voIpProt.SIP.alertInfo.x.class`

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

default (default)

See the list of ring classes in Ringtone Parameters.

`voIpProt.SIP.alertInfo.x.class`

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

default (default)

`voIpProt.SIP.alertInfo.x.class`

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

default (default)

See the list of ring classes in Ringtone Parameters.

`voIpProt.SIP.alertInfo.x.value`

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`

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

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

NULL (default)

voIpProt.SIP.allowTransferOnProceeding

1 (default) - Transfer during the proceeding state of a consultation call is enabled.

0 - Transfer during the proceeding state of a consultation call is enabled

2 - Phones will accept an INVITE with replaces for a dialog in early state. This is needed when using transfer on proceeding with a proxying call server such as openSIPS, reSIProcate or SipXecs.

voipProt.SIP.anat.enabled

Enables or disables Alternative Network Address Types (ANAT).

0 (default) - ANAT is disabled.

1 - ANAT is enabled.

voIpProt.SIP.authOptimizedInFailover

0 (default) - The first new SIP request is sent to the server with the highest priority in the server list when failover occurs.

1 - The first new SIP request is sent to the server that sent the proxy authentication request when failover occurs.

voIpProt.SIP.callee.SourcePreference

Set priority order to display the callee's identity for outgoing calls.

Null (default)

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

String

voIpProt.SIP.Caller.SourcePreference

Set priority order to display the caller's identity for incoming calls.

Null (default)

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

String

voIpProt.SIP.callinfo.precedence.overAlertinfo

0 (default) - Give priority to call-info header with answer-after string over alert-info feature is disabled.

1 - Give priority to call-info header with answer-after string over alert-info feature is enabled.

voIpProt.SIP.callinfo.precedence.overAlertinfo

0 (default) - The alert-info is given priority over call-info header.

1 - The call-info header with answer-after string is given priority over alert-info header.

`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

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

This parameter does not apply to shared lines.

`voIpProt.SIP.CID.sourcePreference`

Specify the priority order for the sources of caller ID information. The headers can be in any order.

Null (default) - Caller ID information comes from P-Asserted-Identity, Remote-Party-ID, and From in that order.

From,P-Asserted-Identity, Remote-Party-ID

P-Asserted-Identity,From,Remote-Party-ID

Supported Headers Default Order: P-Asserted-Identity,Remote-Party-ID,From

If `voIpProt.SIP.Caller.SourcePreference` or `voIpProt.SIP.Callee.SourcePreference` are configured then the order set by `voIpProt.SIP.CID.sourcePreference` is ignored.

`voIpProt.SIP.compliance.RFC3261.validate.contentLanguage`

1 (default) - Validation of the SIP header content language is enabled.

0 - Validation of the SIP header content language is disabled

`voIpProt.SIP.compliance.RFC3261.validate.contentLength`

1 (default) - Validation of the SIP header content length is enabled.

0 - Validation of the SIP header content length is disabled

`voIpProt.SIP.compliance.RFC3261.validate.uriScheme`

1 (default) - Validation of the SIP header URI scheme is enabled.

0 - Validation of the SIP header URI scheme is disabled

`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.

`voIpProt.SIP.conference.parallelRefer`

0 (default) - A parallel REFER is not sent to the call server.

1 - A parallel REFER is not sent to the call server.

Note: This parameter must be set for Siemens OpenScape Centralized Conferencing.

`voIpProt.SIP.connectionReuse.useAlias`

0 (default) - The alias parameter is not added to the via header

1 - The phone uses the connection reuse draft which introduces "alias".

`voIpProt.SIP.dialog.strictXLineID`

0 (default) - The phone will not look for x-line-id (call appearance index) in a SIP INVITE message.

1 - The phone will look for x-line-id (call appearance index) in a SIP INVITE message

`voIpProt.SIP.dialog.usePvalue`

0 (default) - Phone uses a `pval` field name in Dialog.

1 - Phone uses a `pvalue` field name in Dialog.

`voIpProt.SIP.dialog.useSDP`

0 (default) - A new dialog event package draft is used (no SDP in dialog body).

1 - Use this setting to send SDP in the dialog body for backwards compatibility

`voIpProt.SIP.dtmfViaSignaling.rfc2976`

Enable or disable DTMF relays for active SIP calls.

0 (default) - DTMF digit information is not sent

1 - DTMF digit information is sent in RFC2976 SIP INFO packets during a call.

Change causes system to restart or reboot.

`voIpProt.SIP.dtmfViaSignaling.rfc2976.nonLegacyEncoding`

Controls the behavior of the Star and Pound keys used for DTMF relays for active SIP calls.

0 (default) - The phone sends 10 when the Star key (*) is pressed and 11 when the Pound key (#) is pressed.

1 - The phone sends an asterisk (*) when the Star key is pressed and a hashtag (#) when the Pound key is pressed.

Change causes system to restart or reboot.

`voIpProt.SIP.enable`

A flag to determine if the SIP protocol is used for call routing, dial plan, DTMF, and URL dialing.

1 (default) - The SIP protocol is used.

0 - The SIP protocol is not used.

Change causes system to restart or reboot.

voIpProt.SIP.failoverOn503Response

A flag to determine whether or not to trigger a failover if the phone receives a 503 response. You must use a registration expiry of 66 seconds or greater for failover with a 503 response to work properly. This rule applies both to the phone configuration (`reg.x.server.y.expires` and `voIpProt.server.x.expires`) as well as the 200 OK register response from the server.

1 (default) - Enabled

0 - Disabled

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.

voIpProt.SIP.header.pEarlyMedia.support

0 (default) - The p-early-media header is not supported by the caller phone.

1 - The p-early-media header is supported by the caller phone.

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.

comma separated list

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`

voIpProt.SIP.header.warning.enable

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

1 - The warning header is displayed if received.

voIpProt.SIP.ignore.pEarlyMediaInactive

0 (default) – The phone does not ignore SIP messages received with “inactive” in the p-Early-Media header.

1 – The phone ignores SIP messages received with “inactive” in the p-Early-Media header on a non-active early dialog in case of forking and does not switch to a local ringback tone.

This parameter applies only when `voIpProt.SIP.header.pEarlyMedia.support` is enabled.

voIpProt.SIP.IM.autoAnswerDelay

The time interval from receipt of the instant message invitation to automatically accepting the invitation.

10 (default)

0 to 40

voIpProt.SIP.IMS.enable

This parameter applies to all registered or unregistered SIP lines on the phone.

0 (default) - The phone does not support IMS features introduced in UC Software 5.5.0.

1 - The phone supports IMS features introduced in UC Software 5.5.0.

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

voIpProt.SIP.keepalive.sessionTimers

0 (default) - The session timer is disabled.

1 - The session timer is enabled.

voIpProt.SIP.lineSeize.retries

Controls the number of times the phone will retry a notify when attempting to seize a line (BLA).

10 (default)

3 to 10

voIpProt.SIP.local.port

The local port for sending and receiving SIP signaling packets.

5060 - The value is used for the local port but is not advertised in the SIP signaling.

0 to 65535 - If set to 0, the 5060 value is used for the local port but is not advertised in the SIP signaling. For other values, that value is used for the local port and it is advertised in the SIP signaling

Change causes system to restart or reboot.

voIpProt.SIP.looseContact

0 (default) - The port parameter is added to the contact header in TLS case.

1 - The port parameter is not added to the contact header or SIP messages.

voIpProt.SIP.noContactHeaderIn200OKForNotify

0 (default) – Disabled

Phone sends contact header in 200 ok for NOTIFY.

1 – Enabled

Phone doesn't send contact header in 200 ok for NOTIFY.

`voIpProt.SIP.ms-forking`

This parameter applies when installing Microsoft Live Communications Server.

0 (default) - Support for MS-forking is disabled.

1 - Support for MS-forking is enabled.

Note: If any endpoint registered to the same account has MS-forking disabled, all other endpoints default back to non-forking mode. Windows Messenger does not use MS-forking so be aware of this behavior if one of the endpoints is using Windows Messenger.

`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

`voIpProt.SIP.newCallOnUnRegister`

1 (default) - The phone generates new Call-ID and From tag during re-registration.

0 - The phone does not generate new Call-ID and From tag during re-registration.

`voIpProt.SIP.outboundProxy.address`

The IP address or hostname of the SIP server to which the phone sends all requests.

Null (default)

IP address or hostname

`voIpProt.SIP.outboundProxy.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.

registration - The phone tries the primary server again when the registration renewal signaling begins.

`voIpProt.SIP.outboundProxy.failOver.failBack.timeout`

The time to wait (in seconds) before failback occurs (overrides `voIpProt.server.x.failOver.failBack.timeout`).

3600 (default) -If the fail back mode is set to Duration, the phone waits this long after connecting to the current working server before selecting the primary server again.

0, 60 to 65535 -If set to 0, the phone doesn't fail-back until a fail-over event occurs with the current server.

voIpProt.SIP.outboundProxy.failOver.failRegistrationOn

1 (default) - When set to 1, and the 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 `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.

Note: `voIpProt.SIP.outboundProxy.failOver.reRegisterOn` must be enabled.

voIpProt.SIP.outboundProxy.failOver.onlySignalWithRegistered

1 (default) - No signaling is accepted from or sent to a server that has failed until failback is attempted or failover occurs.

0 - Signaling is accepted from and sent to a server that has failed (even though failback hasn't been attempted or failover hasn't occurred). This parameter overrides `voIpProt.server.x.failOver.onlySignalWithRegistered`.

Note: `reRegisterOn` and `failRegistrationOn` parameters must be enabled.

voIpProt.SIP.outboundProxy.failOver.reRegisterOn

This parameter overrides the `voIpProt.server.x.failOver.reRegisterOn`.

0 (default) - The phone doesn't 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 the secondary server. If the registration succeeds signaling proceeds with the secondary server.

voIpProt.SIP.outboundProxy.port

The port of the SIP server to which the phone sends all requests.

0 (default)

0 to 65535

voIpProt.SIP.outboundProxy.transport

DNSnaptr (default) - If `reg.x.outboundProxy.address` is a hostname and `reg.x.outboundProxy.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.outboundProxy.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 will be used.

TLS - If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061.

TCPOnly - Only TCP will be used.

voIpProt.SIP.outboundProxy.transport

0 (default) - This feature is disabled.

1 - Enable line to SBC mapping and SBC list traversal.

`voIpProt.SIP.pingInterval`

The number in seconds to send PING message.

0 (default) - This feature is disabled.

0 to 3600 - This feature is enabled.

`voIpProt.SIP.pingMethod`

The ping method to be used.

PING (default)

OPTIONS

`voIpProt.SIP.presence.nortelShortMode`

This parameter is required when using the Presense feature with an Avaya or Ribbon Communications server.

0 (default)

1 - Different headers are sent in SUBSCRIBE when used feature with an Avaya or Ribbon Communications server. Support is indicated by adding a header `Accept-Encoding: x-nortel-short`. A PUBLISH is sent to indicate the status of the phone.

Change causes system to restart or reboot.

`voIpProt.SIP.regevent`

0 (default) - The phone is not subscribed to registration state change notifications for all phone lines.

1 - The phone is subscribed to registration state change notifications for all phone lines.

This parameter is overridden by the per-phone parameter reg.x.regevent.

`voIpProt.SIP.rejectNDUBInvite`

Specify whether or not the phone accepts a call for all registrations 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 for all line registrations.

1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code for all line registrations.

`voIpProt.SIP.renewSubscribeOnTLSRefresh`

1 (default) - For an as-feature-event, the SUBSCRIBE message is sent along with the RE-REGISTER when Transport Layer Security (TLS) breaks.

0 - The SUBSCRIBE and RE-REGISTER messages is sent at different times.

`voIpProt.SIP.rport`

0 (default) – The phone does not insert the rport parameter into the Via header of its requests.

1 – The phone inserts the rport parameter, as defined by RFC 3581, into the Via header of its requests.

`voIpProt.SIP.requestURI.E164.addGlobalPrefix`

0 (default) - ‘+’ global prefix is not added to the E.164 user parts in sip: URIs.

1 - ‘+’ global prefix is added to the E.164 user parts in sip: URIs.

`voIpProt.SIP.requestValidation.digest.realm`

Determines the string used for Realm.

PolycomSPIP (default)

string

Change causes system to restart or reboot.

`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.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

ALL - The phone controls all requests from unknown sources.

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.

`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 are validated.

A valid string - specified event is validated.

Change causes system to restart or reboot.

`voIpProt.SIP.RFC3261TimerI`

0 (default) - Timer I for reliable transport will be fired at five seconds. This parameter does not cause any change for unreliable transport.

1 - Timer I for reliable transport is fired at zero seconds.

`voIpProt.SIP.sendCompactHdrs`

0 (default) - SIP header names generated by the phone use the long form, for example `From`.

1 - SIP header names generated by the phone use the short form, for example `f`.

`voIpProt.SIP.serverFeatureControl.callRecording`

0 (default) - The BroadSoft BroadWorks v20 call recording feature for multiple phones is disabled.

1 - The BroadSoft BroadWorks v20 call recording feature for multiple phones is enabled.

`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.cf`

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

1 - Enable server-based call forwarding.

Change causes system to restart or reboot.

`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.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.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`.

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.

voIpProt.SIP.serverFeatureControl.missedCalls

0 (default) - Server-based missed calls is not enabled.

1 - Server-based missed calls is enabled. The call server has control of missed calls.

Change causes system to restart or reboot.

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.

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.

voIpProt.SIP.specialEvent.checkSync.alwaysReboot

0 (default) - The phone only reboots if necessary. Many configuration parameter changes can be applied dynamically without the need for a reboot.

1 - The phone always reboots when a NOTIFY message is received from the server with event equal to check-sync even if there has not been a change to software or configuration.

voIpProt.SIP.specialEvent.checkSync.downloadCallList

0 (default) - The phone does not download the call list for the logged-in user when a check sync event's NOTIFY message is received from the server.

1 - The phone downloads the call list for the logged-in user when a check sync event's NOTIFY message is received from the server.

`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.

`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.

`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.

`voIpProt.SIP.strictLineSeize`

0 (default) - Dial prompt is provided immediately when you attempt to seize a shared line without waiting for a successful OK from the call server.

1 - The phone is forced to wait for a 200 OK response when receiving a TRYING notify.

`voIpProt.SIP.strictReplacesHeader`

This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.

1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when `call.directedCallPickupMethod` is configured as native.

0 - Call pick-up requires a call id only.

`voIpProt.SIP.strictReplacesHeader`

This parameter applies only to directed call pick-up attempts initiated against monitored BLF resources.

1 (default) - The phone requires call-id, to-tag, and from-tag to perform a directed call-pickup when `call.directedCallPickupMethod` is configured as native.

0 - Call pick-up requires a call id only.

`voIpProt.SIP.strictUserValidation`

0 (default) - The phone is forced to match the user portion of signaling exactly.

1 - The phone uses the first registration if the user part doesn't match any registration.

voIpProt.SIP. supportFor100rel

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.supportFor199

Determine support for the 199 response code. For details on the 199 response code, see RFC 6228.

0 (Default) - The phone does not support the 199 response code.

1- The phone supports the 199 response code.

voIpProt.SIP.tcpFastFailover

0 (default) - A full 32 second RFC compliant timeout is used.

1 - A failover occurs based on the values of `reg.x.server.y.retryMaxCount` and `voIpProt.server.x.retryTimeOut`.

voIpProt.SIP.tcpFastFailover.timeout

2000 to 5000 - Define the time to wait before failing over to the next IP in the list of records resolved by the DNS server applicable only before the TCP connection establishment.

5000 (default).

voIpProt.SIP.tlsDsk.enable

0 (default) - TLS DSK is disabled.

1 - TLS DSK is enabled.

voIpProt.SIP.turnOffNonSecureTransport

0 (default) - Port 5060 is open for SIP messaging.

1 - Port 5060 is not open for SIP messaging.

Change causes system to restart or reboot.

voIpProt.SIP.use486forReject

0 (default) - The phone responds with 603.

1 - The phone responds with 486.

voIpProt.SIP.useContactInReferTo

0 (default) - The "To URI" is used in the REFER.

1 - The "Contact URI" is used in the REFER.

voIpProt.SIP.useLocalTargetUriforLegacyPickup

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 target URI in the XML dialog document is used and the current registrar's domain is appended to create the address for pickup or retrieval.

`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.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.useRFC3264HoldOnly`

0 (default) - When set to 0, and no media direction is specified, the phone enters backward compatibility mode when negotiating SDP and responds using the c=0.0.0.0 RFC 2543 signaling method.

1 - When set to 1, and no media direction is specified, the phone uses `sendrecv` compliant with RFC 3264 when negotiating SDP and generates responses containing RFC 3264-compliant media attributes for calls placed on and off hold by either end.

Note: `voIpProt.SIP.useSendonlyHold` applies only to calls on phones that originate the hold.

`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).

`voIpProt.SIP.ignoreEntityHost`

0 (default) – Doesn’t ignore the host part of the entity received in the XML body of NOTIFY for a dialog event.

1 - Ignores the host part of the entity received in the XML body of NOTIFY for a dialog event.

`voIpProt.SIP.forkedRespRecommendedCseq`

1 (default) - Generates the RFC compliance Cseq number.

0 - Generates the call specific CSeq number.

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 48

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 - 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.

Presence Status Parameters

Use the following parameters to enable Presence and display the **MyStatus** and **Buddies** soft keys on the phone.

`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.

`pres.idleSoftkeys`

1 (default) - The MyStat and Buddies presence idle soft keys display.

0 - The MyStat and Buddies presence idle soft keys do not display.

`pres.reg`

The valid line/registration number to use for presence. If the value is not a valid registration, this parameter is ignored.

1 (default)

1 - 34

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.

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

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`

STIR/SHAKEN Caller ID Validation Parameters

Use the following parameters to configure the STIR/SHAKEN caller ID validation.

`reg.x.SIP.stirshakenCallerVerification.enabled`

0 (default) - Disabled.

1 - Enables caller ID validation based on STIR/SHAKEN.

`reg.x.SIP.stirshaken.attestationName`

String - PAI header parameter name that's parsed for caller ID validation.

`verstat` (default)

0-64 characters

reg.x.SIP.stirshaken.attestationValue

A list of all the possible caller ID attestation values. Values are comma separated with no spaces.

TN-VALIDATION-PASSED,TN-VALIDATION-PASSED-A,TN-VALIDATION-PASSED-B,TN-VALIDATION-PASSED-C,NO-TN-VALIDATION,TN-VALIDATION-FAILED (default)

0-256 characters

reg.x.SIP.stirshaken.verstatPassed

A subset of the values listed in `reg.x.SIP.stirshaken.attestationValue` that pass validation.

TN-VALIDATION-PASSED,TN-VALIDATION-PASSED-A,TN-VALIDATION-PASSED-B (default)

0-256 characters

reg.x.SIP.stirshaken.verstatNotAvailable

A subset of the values listed in `reg.x.SIP.stirshaken.attestationValue` that don't need validation.

NO-TN-VALIDATION (default)

0-256 characters

reg.x.SIP.stirshaken.verstatFailed

A subset of the values listed in `reg.x.SIP.stirshaken.attestationValue` that fail validation.

TN-VALIDATION-PASSED-C,TN-VALIDATION-FAILED (default)

0-256 characters

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.

Certificate Parameters

Use the following parameters to configure TLS, LDAP, and online certificate status protocol parameters.

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 seven application profiles.
- Map profiles directly to the features that use certificates.

`device.sec.TLS.customCaCert1`

Specify a custom Platform CA 1 certificate.

Null (default)

String (maximum of 12288 characters)

`device.sec.TLS.customCaCert`

Specify a custom Platform CA 1 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.caCertList2`

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.cipherSuite2`

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.cipherSuiteDefault2`

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 for Platform Profile 1.

Builtin (default)

Builtin, Platform1, Platform2

`device.sec.TLS.profile.deviceCert2`

Specify which device certificates to use for Platform Profile 2.

Builtin (default)

Builtin, Platform1, Platform2

`sec.TLS.customCaCert.x`

The custom certificate to use with TLS Application Profiles.

Up to 7 (x=1 to 7) custom application certificates can be configured.

Null (default)

String

`sec.TLS.customDeviceKey.x`

The custom certificate to use with TLS Application Profiles.

Up to 7 (x=1 to 7) custom application certificates can be configured.

Null (default)

String

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

1 (default) - Enable Application CA 1 certificate for Application Profile x.

0 - Disable Application CA 1 certificate for Application Profile x.

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

1 (default) - Enable Application CA 2 certificate for Application Profile x.

0 - Disable Application CA 2 certificate for Application Profile x.

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

1 (default) - Enable Application CA 3 certificate for Application Profile x.

0 - Disable Application CA 3 certificate for Application Profile x.

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

1 (default) - Enable Application CA 4 certificate for Application Profile x.

0 - Disable Application CA 4 certificate for Application Profile x.

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

1 (default) - Enable Application CA 5 certificate for Application Profile x.

0 - Disable Application CA 5 certificate for Application Profile x.

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

1 (default) - Enable Application CA 6 certificate for Application Profile x.

0 - Disable Application CA 6 certificate for Application Profile x.

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

1 (default) - Enable Application CA 7 certificate for Application Profile x.

0 - Disable Application CA 7 certificate for Application Profile x.

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

Specifies whether the default certificate list is used for TLS Application Profile x (x=1 to 7).

1 (default) - Enable the default certificate list for Application Profile x.

0 - Disable the default certificate list for Application Profile x.

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

1 (default) - Enable Platform CA 1 certificate for Application Profile x.

0 - Disable Platform CA 1 certificate for Application Profile x.

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

1 (default) - Enable Platform CA 2 certificate for Application Profile x.

0 - Disable Platform CA 2 certificate for Application Profile x.

`sec.TLS.profile.x.cipherSuite`

Specifies the cipher suite for TLS Application Profile x (x=1 to 7).

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

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

`device.sec.TLS.profileSelection.syslog`

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

PlatformProfile1 (default)

PlatformProfile1 or PlatformProfile2

TLS Cipher Suite Parameters

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

`sec.TLS.cipherList`

String (1 - 1024 characters)

ALL:!aNULL:!eNULL:!DSS:!3DES:!CAMELLIA:!SEED:!ECDSA:!IDEA:!MEDIUM:!LOW:!EXP:!DH:!AECDH:!PSK:!SRP:!MD5:!RC4:@STRENGTH (default)

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 or TLS Application.

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

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.

Configuration Parameters

This section is a reference for configuration parameters available for PVOS features.

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`
  - `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`
  - `splash`

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

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

0-2400

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

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

0-2400

`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).

0-2400

`tone.chord.callProg.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.misc.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.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`

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

0=infinite

Positive integer

0-10000

`tone.chord.misc.x.onDur`

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

0=infinite

Positive integer

0-10000

`tone.chord.ringer.x.onDur`

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

0=infinite

Positive integer

0-10000

`tone.chord.callProg.x.offDur`

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

0=infinite

Positive integer

0-10000

`tone.chord.misc.x.offDur`

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

0=infinite

Positive integer

0-10000

`tone.chord.ringer.x.offDur`

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

0=infinite

Positive integer

0-10000

`tone.chord.callProg.x.repeat`

Number of times each ON/OFF cadence is repeated.

0=infinite

Positive integer

0-10000

`tone.chord.misc.x.repeat`

Number of times each ON/OFF cadence is repeated.

0=infinite

Positive integer

0-10000

`tone.chord.ringer.x.repeat`

Number of times each ON/OFF cadence is repeated.

0=infinite

Positive integer

0-10000

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).

1500 (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.extendedDiscovery`

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

0 (default)

0 - 3600

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

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.

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.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.

`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 received by the host on one leg to another.

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

`feature.photoIntegration.enable`

0 - Disable photo integration feature.

1 (default) - Enable photo integration feature.

`feature.restrictPerDataUploadMenu.enabled`

1 (default) - Displays the **Restrict Personal Data Upload** menu under **Basic** settings.

0 - Doesn't display the **Restrict Personal Data** menu under **Basic** settings.

`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 or unregistered lines, and unknown callers are identified on the display by their phone's IP address.

0 - URL/name dialing is not available.

`reg.x.urlDialing.enabled`

1 (default) - Enable dialing by URL for SIP line registration x.

0 - Disable dialing by URL for SIP line registration x.

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.

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

100 (default)

`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

200 (default)

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

Element	Definition	Permitted Values
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.	
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 or Unspecified
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

Element	Definition	Permitted Values
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
loc	Location of the speed dial	0 (primary), 1 (secondary)
tl	title of the contact	64 characters
em	email address of the contact	120 characters
ol	outgoing line to use to place a call to the contact	0 to 33 (0 = line registration 1. ..., 33 = line registration 34)

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.

Call Parameters

Poly phones support various call handling features, including automatically answering calls and missed calls tracking.

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.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 entering of the plus (+) symbol used to indicate an international call.

1 (default) - Enable the entering of the plus (+) symbol for international calls.

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`

This parameter is applicable only when call.internationalDialing.enabled="1".

Note: To enter a double asterisk (**) or double 0 (00), tap the asterisk (*) or 0 key once and wait for the key tap timer to expire to enter a second asterisk (*) or 0.

0 (default) - A quick double tap of * converts immediately to + to enter the international dialing prefix.

1 - A quick double tap of 0 converts immediately to + to enter the international dialing prefix.

`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.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.x.applyToCallListDial

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 (default) - 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 - 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 - The dial plan does not apply to forwarded calls for this line.

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

dialplan.x.applyToTelUriDial

0 - The dial plan is not applied to tel: URIs for this line.

1 (default) - The dial plan is applied to tel: URIs for this line.

Change causes system to restart or reboot.

dialplan.x.applyToUserDial

0 - The dial plan is not applied when the user presses the Dial soft key to send the dialed number when in idle state.

1 (default) - The dial plan is applied when the user presses the Dial soft key to send the dialed number when in idle state.

Change causes system to restart or reboot.

dialplan.x.applyToUserSend

0 - The dial plan is not applied when the user presses the Send soft key to send the dialed number.

1 (default) - The dial plan is applied when the user presses the Send soft key to send the dialed number.

Change causes system to restart or reboot.

`dialplan.x.conflictMatchHandling`

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

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.

0-100 seconds

Change causes system to restart or reboot.

`dialplan.x.digitmap`

string - max number of characters 2560

Change causes system to restart or reboot.

`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`

Specifies whether the trailing number is stripped from the digits sent out for the line.
0 - Disabled

1 (default) - Enabled

Change causes system to restart or reboot.

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

0 (default) - No alternate server is used when the emergency number set by dialplan.x.routing.emergency.y.value is dialed from this line x.

1 - The alternate server defined by dialplan.x.routing.emergency.1.address is used when the emergency number set by dialplan.x.routing.emergency.y.value is dialed from this line x.

2 - The alternate server defined by dialplan.x.routing.emergency.2.address is used when the emergency number set by dialplan.x.routing.emergency.y.value is dialed from this line x.

3 - The alternate server defined by dialplan.x.routing.emergency.3.address is used when the emergency number set by dialplan.x.routing.emergency.y.value is dialed from this line x.

x - 134

y and z = 1 to 3

Change causes system to restart or reboot.

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

Set the emergency number to dial from this line x. You can configure up to 3 numbers (1-3) for y.
Null (default)

string - max number of characters 64

Change causes system to restart or reboot.

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

Set the alternate server address to send emergency calls to for this line x.
Null (default)

string - max number of characters 256

Change causes system to restart or reboot.

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

Set the port number for the alternate server to send emergency calls to for this line x.
5060 (default)

1 to 65535

Change causes system to restart or reboot.

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

Set the transport for the alternate server to send emergency calls to for this line x.
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.period`

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.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.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.

Quick Setup 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.

REST API Parameter

Use the following parameter to enable the REST API.

`apps.restapi.enabled`

0 (default) - Disabled

1 - Enabled

SDP Parameters

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

`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.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 - The phone includes the SDP attribute "a=rtcp-fb" into offers during outbound SIP calls.

Session Timer Parameters

You can configure the phone to enable support for session timers in SIP signaling during calls.

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.

Software Upgrade Parameters

Specify the URL of a custom download server and the PVOS download server when you want the phone to check for software and perform software upgrades using the web configuration utility interface.

upgrade.custom.server.url

The URL of a custom download server when the phone is upgraded through the Web Configuration Utility, and the Poly Hosted Server option is selected as the Server Type.

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 1 to 3.

1 (Default) - Low

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.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.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.

`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.enable` 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.enable` 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.enable` 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

XMPP Parameters

Use the following parameters to set the XMPP feature 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

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.

Types of Device Parameters

The following parameters outline the three types of `<device/>` parameters, their permitted values, and the default value.

`device.set`

0 (default) - Don't use any `device.xxx` fields to set any parameters. Set this to 0 when you are not making changes to device parameters.

1 - Use the `device.xxx` fields that have `device.xxx.set="1"`. Set this to 1 when you are making changes to device parameters.

Change may cause system to restart or reboot.

`device.xxx`

Configuration parameter.

String

Change may cause system to restart or reboot.

`device.xxx.set`

0 (default) - Don't use the `device.xxx`value.

1 - Use the `device.xxx` value.

For example, if `device.net.ipAddress.set="1"`, then use the value set for `device.net.ipAddress`.

Change may cause system to restart or reboot.

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.

MSTeams - Sets the base profile to Microsoft Teams.

ZoomPhone - Sets the base profile to Zoom Phone.

8x8Work - Sets the base profile to 8x8 Work.

USBOptimized - Sets the base profile to USB Optimized.

Dialpad - Sets the base profile to Dialpad.

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.

160 (default)

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 (default) - 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"`.

129 (default)

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 (default) - 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.eapWorkaround`

`device.net.dot1x.eapWorkaround.set`

`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.extendedDiscovery`

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 HTTP/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. The phone will use the path specified in this parameter, if it is a non-NULL value, to look for the sip.ld software file. Otherwise, it will use the path specified in the APP_FILE_PATH attribute in the master configuration file.

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.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.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 device.sec.TLS.profile.caCertList2`

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 device.sec.TLS.profile.cipherSuite2

Enter the cipher suites to use for TLS Platform Profile 1 and TLS Platform Profile 2

String

device.sec.TLS.profile.cipherSuiteDefault1 device.sec.TLS.profile.cipherSuiteDefault2

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 device.sec.TLS.profile.deviceCert2

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 - Disables common name validation.

1 - Syslog 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 - Disables common name validation.

1 (default) - Verify the 802.1x server certificate for the `commonName/SubjectAltName` match with the server hostname that the phone is trying to connect.

`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)

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

Diagnostic and Status Parameters

Use the following parameters to configure diagnostic settings including factory resetting a phone.

Phone Boot Status Parameters

Use the following parameters to configure the phone boot status popup message.

`up.phoneBootStatusPopupEnabled`

1 (default) - The phone displays a popup message with phone status details after a restart or reboot.

0 - The phone does not display a popup message after a restart or reboot.

Phone Memory Alert Parameters

The following parameters configure the phone memory alert feature.

`up.sysFreeMemThresholdPercent`

Set the threshold of free memory, in percentage, below which the phone displays a warning message.

20 percent (default)

20 - 30 percent

`up.lowSysMemWarn.timeInMins`

Set the interval, in minutes, that the phone's free memory is checked.

0 (default)

0 - 1440 minutes

Remote Packet Capture Parameters

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

`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

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.

`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.

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.

Directories and Contacts Parameters

Use the directories and contacts parameters to configure contact directory, speed dial, and call log settings.

Corporate Directory Parameters

Use the parameters in the following table 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 Polycom Phones* at Poly Support.

`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 restart or reboot.

`dir.corp.allowCredentialsFromUI.enabled`

Enable users to enter LDAP credentials on the phone.

0 (default) – Users are not prompted to enter credentials on the phone when they access the Corporate Directory.

1 – Users are prompted to enter credentials on the phone when accessing the Corporate Directory for the first time.

Note: Users are only prompted to enter their credentials when credentials are not added through configuration or after a login failure.

`dir.corp.alt.port`

Set the port that connects to the server if a full URL is not provided.

0 (default)

Null

1 to 65535

`dir.corp.alt.protocol`

Set a directory protocol used to communicate to the corporate directory.

sopi (default)

UTF-8 encoding string

`dir.corp.alt.transport`

Choose a transport protocol used to communicate to the corporate directory.

TCP (default)

TLS

dir.corp.alt.user

Enter the user name used to authenticate to the GENBAND server.

Null (default)

UTF-8 encoding string

dir.corp.alt.viewPersistence

Determine if the results from the last address directory search displays on the phone.

0 (default)

1

dir.corp.attribute.x.addstar

Determine if the wild-card character, asterisk(*), is appended to the LDAP query field.

0

1 (default)

Change causes system 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 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 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 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)

1

Change causes system restart or reboot.

dir.corp.attribute.x.sticky

0 (default) - The filter string criteria for attribute x is reset after a reboot.

1 - The filter string criteria is retained through a 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.

Change causes system 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.

first_name

last_name (default)

phone_number

SIP_address

other

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.

Change causes system restart or reboot.

dir.corp.auth.useLoginCredentials

0 (default)

1

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): There is no timeout and automatic submit is disabled.

0 - 60 seconds

Change causes system restart or reboot.

dir.corp.backGroundSync

Determine if background downloading from the LDAP server is allowed.

0 (default)

1

Change causes system 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 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 restart or reboot.

dir.corp.bindOnInit

Determine if bind authentication is used on initialization.

1 (default)

0

Change causes system 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 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

0 to 255

dir.corp.filterPrefix

Enter the predefined filter string for search queries.

(objectclass=person) (default)

UTF-8 encoding string

Change causes system restart or reboot.

dir.corp.pageSize

Set the maximum number of entries requested from the corporate directory server with each query.

Change causes system 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`

Set 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.

0 (default)

1

Note: If you disable the feature after enabling it, then all the saved user credentials are deleted for all users.

`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 restart or reboot.

`dir.corp.querySupportedControlOnInit`

Determine if the phone makes an initial query to check the status of the server when booting up.

0

1 (default)

`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 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 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 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.ui.nameDisplay`

Defines the format in which LDAP query results display.

last_name_first_name (default) - LDAP query results display as **last_name, first_name, number**.

first_name_last_name - LDAP query results display as **first_name last_name, number**.

`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 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 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 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 Parameters

The following parameters configure the local contact directory.

`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'.

Change causes system to restart or reboot.

`dir.local.mode`

Determine if the phone prioritizes local end-user changes or allows the server-side MAC-directory file to override local edits.

devicePrioritized – Adding or editing contacts using the phone’s interface will store them as a local directory that is not uploaded to the phone’s provisioning server. This local copy is merged for display to the phone user with any directories received from the server. In cases where a contact has been edited, and the server later pushes a change (such as a contact’s name), the changes made by the phone user will be given the display priority.

Note: When `dir.local.mode="devicePrioritized"`, changing the value of `dir.local.devicePrioritized.deleteDirectory` deletes the contents of the phone's local directory.

serverPrioritized (default) – Changes to the contact directory made from the desk phone’s interface will be saved in the MAC-directory.xml file and uploaded to the phone’s provisioning server. In cases where the server does not accept the upload or does not process it to merge the change, the phone will keep the changes only until the server pushes its copy of the directory to the phone which will then result in the user's changes being deleted/overwritten.

`dir.local.devicePrioritized.deleteDirectory`

Enter any string value to remotely delete all device directory contacts. It is recommended to use a timestamp (DD-MM-YYYY-HH:MM:SS).

Note: Replacing Null with any value triggers the delete action.

Null (default)

String

`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.

`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.

Hardware and Accessories Parameters

Use the following parameters to configure power saving, headset, and speakerphone settings.

Headset and Speakerphone Parameters

You can use the parameters in the following list to enable and disable the headset or speakerphone and control other options for the headset and speakerphone.

`up.analogHeadsetOption`

Electronic Hookswitch (EHS) mode for the phone's analog headset jack.

2 (default) - Plantronics EHS-compatible headset is attached.

0 - No EHS-compatible headset is attached.

1 - Jabra EHS-compatible headset is attached.

3 - Sennheiser EHS-compatible headset is attached.

Change causes system to restart or reboot.

Power-Saving Parameters

Use the following parameters to configure power-saving features and feature options.

`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).

Microsoft Exchange Parameters

The following parameters configure Microsoft Exchange integration.

Call Log Synchronization Parameter

Use the following parameter to configure call logs.

feature.exchangeCallLog.enabled

1 (default) - The Exchange call log feature is enabled and the user call log history of Missed, Received, and outgoing calls can be retrieved on the phone.

You must also enable the parameter `feature.exchangeCalendar.enabled` to use the Exchange call log feature. If you disable `feature.exchangeCalendar.enabled` , also disable `feature.exchangeCallLog.enabled` to ensure call log functionality.

0 (default) - The Exchange call log feature is disabled and the user call logs history cannot be retrieved from the Exchange server.

Calendar Month View Parameters

The following parameters configure the month view.

calendar.monthView.enabled

0 (default) - Disables the **Month View** soft key.

1 - Enables the **Month View** soft key.

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.deleteUnlistedEvents

0 (default) - Remove events from the day view if they stop being reported by the server.

1 - Do not remove events from the day view if they stop being reported by the server.

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.parseOption`

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

Location (default)

All

LocationAndSubject

Description

Change causes a reboot.

`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.pollInterval`

The interval, in milliseconds, to poll the Exchange server for new meetings.

30000 (default)

4000 minimum

60000 maximum

`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`.

Generic Base Profile default is 0.

1 - Exchange autodiscovery is enabled and the phone automatically discovers the Exchange server using the email address or SIP URI 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.exchange2019.interop.enabled

0 (default) - Disabled

1 - The device sends a read notification for voicemail after playing to mark the voicemail has been read on the server.

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.oneTouchDirectory`

Generic Base Profile default is 0.

1 - The Skype for Business Search icon displays on the Home screen.

0 - The Skype for Business Search icon does not display on the Home screen.

Networking Parameters

Configure phone networking features using the following parameters.

3GPP Technical Specifications Parameters

Use the 3GPP parameters in the following list to configure IP Multimedia Subsystem (IMS) features.

`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), GENBAND, ALU-CTS, ocs2007r2, lcs2005

`voice.qualityMonitoring.processServiceRoute.enable`

0 (Default) - The VQMon messages generated by the phone do not contain service route information in SIP route headers.

1 - The VQMon messages generated by the phone contain service route information, if available, in SIP route headers.

Change causes system to restart or reboot.

`voIpProt.SIP.IMS.enable`

This parameter applies to all registered or unregistered SIP lines on the phone.

0 (Default) - The phone does not support IMS features introduced in UC Software 5.5.0.

1 - The phone supports IMS features introduced in UC Software 5.5.0.

`voIpProt.SIP.regevent`

0 (default) - The phone is not subscribed to registration state change notifications for all phone lines.

1 - The phone is subscribed to registration state change notifications for all phone lines.

This parameter is overridden by the per-phone parameter reg.x.regevent.

`voIpProt.SIP.rejectNDUBInvite`

Specify whether or not the phone accepts a call for all registrations 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 for all line registrations.

1 - If an NDUB event occurs, the phone rejects the call with a 603 Decline response code for all line registrations.

`voIpProt.SIP.supportFor199`

Determine support for the 199 response code. For details on the 199 response code, see RFC 6228.

0 (Default) - The phone does not support the 199 response code.

1- The phone supports the 199 response code.

Bluetooth Parameters

Use the parameters in the following list to configure Bluetooth.

`bluetooth.device.discoverable`

1 (default) - This device is discoverable for Bluetooth pairing.

0 - This device is not discoverable for Bluetooth pairing.

`bluetooth.device.name`

NULL (default)

UTF-8 string

Enter the name of the device that broadcasts over Bluetooth to other devices.

`bluetooth.discoverableTimeout`

Set the time in seconds after which other devices can discover this device over Bluetooth.

0 (default) - Other devices can always discover this device over Bluetooth.

0 - 3600 seconds

`bluetooth.pairedDeviceMemorySize`

Set the maximum number of devices that can be paired and cached as paired on the phone.

10 (default)

0 - 10

`bluetooth.radioOn`

0 - The Bluetooth radio transmitter/receiver is off.

1 (default) - The Bluetooth radio is on. You must turn on the Bluetooth radio to allow devices to connect over Bluetooth.

`feature.bluetooth.enabled`

For high security environments.

1 (default) – Enable Bluetooth and the Bluetooth phone screen icon.

0 - Disable Bluetooth and the Bluetooth phone screen icon.

Advice of Charge Parameters

The following parameters configure the Advice of Charge (AoC) feature.

Before configuring AoC parameters, you must set `voIpProt.SIP.IMS.enable to 1.`

`voIpProt.SIP.aoc.enable`

0 (Default) - The phone does not display call charge information.

1 - The phone displays call charge information.

`feature.adviceOfCharge.allowAudioNotification`

0 (Default) - There is no audio beep sound when the call charges information is updated on the phone display.

1 - The phone gives an audio beep when the call charges information is updated on the phone display.

DHCP IP Address Cache Configuration Parameters

Use the following parameters to configure DHCP IP address cache.

`device.net.cachedIPAddress`

0 (default) – IP addresses isn't cached.

1- If a DHCP response isn't received, the phone uses the last assigned IP address, provided one is cached already. A DHCP discover message is retried every `device.net.cachedIPAddressRetryTime` second.

`device.net.cachedIPAddressRetryTime`

Specify the time in seconds to send new DHCP to discover messages when using a cached IP address.

Note: This is only applicable when `device.net.cachedIPAddress` is enabled.

3600 (default)

300 - 7200

GZIP Encoding Parameter

Use the following parameter to configure GZIP Encoding to send notifications to the server.

`voIpProt.SIP.gzipEncoding.enable`

Enable or disable GZIP encoding.

0 - Disabled (Default)

1 - 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.

ICMP Parameters

You can configure Internet Control Message Protocol (ICMP) 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.

Map TR-104 Parameters to Poly Parameters

The data model TR-104 defines the TR-069 ACS parameter details.

The parameters listed as **Internal Value** are not directly mapped to a configuration parameter on the phone and the phone generates these values dynamically to provide to the ACS server.

The following table lists the TR-104 parameters and their corresponding Poly parameters.

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
DigitMap	`dialplan.digitmap`	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.SIP

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
RegistrarServer	`voIpProt.server.X.address`	Yes
RegistrarServerPort	`voIpProt.server.X.port`	Yes

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
OutboundProxy	voIpProt.SIP.outboundProxy.address	Yes
OutboundProxyPort	voIpProt.SIP.outboundProxy.port	Yes
RegisterExpires	voIpProt.server.X.expires	Yes
RegistersMinExpires	voIpProt.server.X.expires.overlap	Yes
RegisterRetryInterval	voIpProt.server.X.retryTimeOut	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.SIP.EventSubscribe.{i}

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
ExpireTime	voIpProt.server.X.subscribe.expires	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.H323

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.RTP

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
LocalPortMin	tcpIpApp.port.rtp.mediaPortRangeStart	Yes
LocalPortMax	tcpIpApp.port.rtp.mediaPortRangeEnd	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.RTP.SRTP

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
Enable	sec.srtp.enable	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.ButtonMap.Button.{i}

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
ButtonName	softkey.X.label	Yes
FacilityAction	softkey.X.action	Yes
UserAccess	softkey.X.enable	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.Line.{i}

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
DirectoryNumber	reg.X.address	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.Line.{i}.SIP

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
AuthUserName	`reg.X.auth.userId`	Yes
AuthPassword	`reg.X.auth.password`	Yes

TR-104 Parameters to Poly Parameters - VoiceService.{i}.VoiceProfile.{i}.Line.{i}.CallingFeatures

TR-104 ACS parameter names	CPE Parameter (Poly parameter names)	Writable
CallForwardUnconditionalEnable	`reg.X.fwdStatus`	Yes
CallForwardUnconditionalNumber	`reg.X.fwdContact`	Yes
CallForwardOnBusyEnable	`reg.X.fwd.busy.status`	Yes
CallForwardOnBusyNumber	`reg.X.fwd.busy.contact`	Yes
CallForwardOnNoAnswerEnable	`reg.X.fwd.noanswer.status`	Yes
CallForwardOnNoAnswerNumber	`reg.X.fwd.noanswer.contact`	Yes
CallForwardOnNoAnswerRingCount	`reg.X.fwd.noanswer.ringCount`	Yes
DoNotDisturbEnable	`divert.dnd.X.enabled`	Yes

Map TR-106 Parameters to Poly Parameters

The data model TR-106 defines the TR-069 ACS parameter details.

The parameters listed as **Internal Value** are not directly mapped to a configuration parameter on the phone, and the phone generates these values dynamically to provide to the ACS server.

The following table lists the TR-106 parameters and their corresponding Poly parameters.

TR-106 Parameters to Poly Parameters - Device.DeviceInfo

TR-106 ACS parameter names	Parameter (Poly parameter names)	Writable
Manufacturer	Internal Value	No
ManufacturerOUI	Internal Value	No
ModelName	Internal Value	No
ProductClass	Internal Value	No
SerialNumber	Internal Value	No
HardwareVersion	Internal Value	No
SoftwareVersion	Internal Value	No
UpTime	Internal Value	No

TR-106 Parameters to Poly Parameters - Device.ManagementServer

TR-106 ACS parameter names	Parameter (Poly parameter names)	Writable
URL	`device.tr069.acs.url`	Yes
Username	`device.tr069.acs.username`	Yes
Password	`device.tr069.acs.password`	Yes
PeriodicInformEnable	`device.tr069.periodicInform.enabled`	Yes
PeriodicInformInterval	`device.tr069.periodicInform.interval`	Yes
ConnectionRequestURL	Internal Value	No
ConnectionRequestUsername	`device.tr069.cpe.username`	Yes
ConnectionRequestPassword	`device.tr069.cpe.password`	Yes
UpgradesManaged	`device.tr069.upgradesManaged.enabled`	Yes
STUNServerAddress	`tcpIpApp.ice.stun.server`	Yes
STUNServerPort	`tcpIpApp.ice.stun.udpPort`	Yes
STUNUsername	`tcpIpApp.ice.username`	Yes
STUNPassword	`tcpIpApp.ice.password`	Yes

TR-106 Parameters to Poly Parameters - Device.LAN

TR-106 ACS parameter names	Parameter (Poly parameter names)	Writable
IPAddress	Internal Value	No
SubnetMask	Internal Value	No
DNSServers	Internal Value	No
MACAddress	Internal Value	No
MACAddressOverride	Internal Value	No

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.

`voIpProt.SIP.requestValidation.digest.realm`

Sets the realm for digest authentication for SIP request validation

PolycomSPIP (default)

string

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.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.

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.filterByIp`

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.filterByPort

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.forceSend

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.mediaPortRangeStart

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.

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

`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.

Up to 18 (x=1..18) can be configured for each DNS record type.

`dns.cache.A.x.address`

Specify the IP address that the hostname specified in dns.cache.A.x.name resolves to for DNS A records.
Null (default)

IP version 4 address

`dns.cache.A.x.name`

Specify the hostname for the DNS A record.
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.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.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

`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.

ICE Parameters

This section lists parameters that configure Interactive Connectivity Establishment (ICE).

`reg.1.ice.reInviteWithSelectedCandidate`

Determines whether the phone sends a re-INVITE with selected ICE candidates. The far end device determines the selected candidates based on STUN binding requests that include the USE-CANDIDATE attribute.

0 (default)- The phone doesn't send a re-INVITE.

1 - The phone sends a re-INVITE.

`tcpIpApp.ice.ConnCheckInetvalPairs`

Time interval in milliseconds to serialize first attempt of connectivity check of identified ICE candidate pairs per call.

25 (default)

10 - 100

`tcpIpApp.ice.ConnCheckInetvalRetries`

Time interval in milliseconds to serialize the retry attempts of connectivity check for identified pairs per call.

50 (default)

25 - 100

`tcpIpApp.ice.MaxRetries`

The maximum number of retry attempts performed on each ICE connectivity check pair identified in case of a request timeout or failure.

2 (default)

2 - 25

`tcpIpApp.ice.mode`

Disabled (default)

MSOCS

Standard

`tcpIpApp.ice.NetworkMode`

TCPUDP (default) – Gathers all the possible UDP and TCP ICE candidates.

TCPOnly – Gathers all the TCP candidates along with UDP host candidates.

UDPOnly - Gathers all the UDP candidates.

`tcpIpApp.ice.password`

Enter the password to authenticate to the TURN server.

NULL (default)

0 - 255

`tcpIpApp.ice.ReflexiveChecksRequired`

1 (default) - TCP and UDP reflexive candidates will be collected in candidate gathering process.

0 - TCP and UDP reflexive candidates will not be collected in candidate gathering process.

`tcpIpApp.ice.stun.server`

Enter the IP address of the STUN server.

NULL (default)

0 - 255

`tcpIpApp.ice.stun.udpPort`

The UDP port number of the STUN server.

3478 (default)

1 - 65535

`tcpIpApp.ice.tcp.enabled`

1 (default) - Enable TCP.

0 - Disable TCP.

`tcpIpApp.ice.turn.server`

Enter the IP address of the TURN server.

NULL (default)

0 - 255

`tcpIpApp.ice.turn.tcpPort`

443 (default)

1 - 65535

`tcpIpApp.ice.turn.udpPort`

The UDP port number of the TURN server.

443 (default)

1 - 65535

`tcpIpApp.ice.username`

Enter the user name to authenticate to the TURN server.

NULL (default)

0 - 255

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 (default) - Apply NAT configuration to both media and signaling per registration.

Disabled - 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.

`nat.refresh.interval`

Set the time interval for the phone to send STUN binding indications to keep the NAT port open and the phone reachable.

30 seconds (default) - The phone sends STUN binding indications for every 30 seconds to keep the NAT port open and the phone reachable.

0 seconds - Disable STUN Binding indication to refresh NAT bindings.

3600 seconds

`nat.device.pollInterval`

Set the time interval for the phone to send STUN binding request to the STUN server to detect whether NAT device is rebooted.

120 seconds (default) - The phone sends the STUN binding requests to the STUN server for every 120 seconds. If NAT IP address or the port details in the STUN binding response don’t match with the previous binding response, the phone automatically restarts.

0 - The phone doesn’t check whether NAT device is rebooted. If NAT device is rebooted and the NAT IP address or the port is changed, the phone doesn’t receive any incoming messages as the IP address and port details published in SIP register message don’t match. You need to restart the phone manually to make the changes effective. Poly recommends not to set the value as 0 seconds.

900 seconds

TR-069 Parameters

Poly provides parameters for the TR-104 and TR-106 data models that support provisioning of TR-069-enabled devices by an Auto-Configuration Server (ACS).

TR-104 is a parameter data model for VoIP-only devices, and TR-106 is a parameter data model for all TR-069-enabled devices.

`device.feature.tr069.enabled`

0 (default) - Disables TR-069 feature

1 - Enables TR-069 feature

`device.feature.tr069.enabled.set`

0 (default) - Disabled

1 - Enabled

`device.tr069.acs.password`

Sets the TR-069 ACS server password used to authenticate the phone.

`Null` (default)

String (256 maximum characters)

`device.tr069.acs.url`

Sets the URL for the TR-069 ACS server.

`Null` (default)

URL (256 maximum characters)

`device.tr069.acs.username`

Sets the TR-069 ACS server user name used to authenticate the phone.

`PlcmSpip` (default)

String (256 maximum characters)

`device.tr069.cpe.password`

Specifies the TR-069 CPE password, which authenticates a connection request from the ACS server.

`Null` (default)

String (256 maximum characters)

`device.tr069.cpe.username`

Specifies the TR-069 CPE user name, which authenticates a connection request from the ACS server.

`PlcmSpip` (default)

String (256 maximum characters)

`device.tr069.periodicInform.enabled`

Indicates whether the CPE must periodically send CPE information to ACS using the Inform method call.

0 (default) - Periodic Inform call is disabled.

1 - Periodic Inform call is enabled.

`device.tr069.periodicInform.interval`

Specifies the time interval in seconds in which the CPE must attempt to connect with the ACS to send CPE information if `device.tr069.periodicInform.enabled` ="1".

18000 (default)

0 to 36000

`device.tr069.upgradesManaged.enabled`

Indicates whether the ACS manages image upgrades for the phone or not.

0 (default) - The phone uses ACS or provisioning server for upgrade.

1 - The phone upgrades only from the ACS server.

`log.level.change.tr069`

Sets the log levels for the TR-069 feature.

4 (default)

0 - 6

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

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.

Wi-Fi Parameters

The parameters you configure depend on the security mode of your organization and whether or not you enable DHCP.

Poly phones support the following Wi-Fi security modes:

- WPA PSK
- WPA2 PSK
- WPA2 Enterprise

`device.wifi.country`

Enter the two-letter country code where you are connecting to a Wi-Fi network.

NULL (default)

Two-letter country code

`feature.wifiUserSettings.enabled`

1 (default) - Display Wi-Fi menu options on the phone menu.

0 - Wi-Fi menu options do not display on the phone menu.

`device.wifi.dhcpEnabled`

Enable or disable DHCP for Wi-Fi.

0 (default)

1

`device.wifi.enabled`

Enable or disable Wi-Fi.

0 (default)

1

`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.psk.keyType

Set the Pre-Shared Key (PSK) type

0 (default) - Passphrase.

1 - Hexadecimal key.

device.wifi.radio.enable2ghz

device.wifi.radio.enable5ghz

device.wifi.securityMode

Specify the wireless security mode.

NULL (default)

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.wpa2Ent.method

Set the Extensible Authentication Protocol (EAP) to use for 802.1X authentication.

NULL (default)

EAP-PEAPv0/MSCHAPv2

EAP-TLS

EAP-TTLS-MSCHAPv2

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.

`feature.wifi.usersettings.enable`

1 (default) – Wi-Fi options display in the Basic settings menu.

0 – Wi-Fi options do not display in the Basic settings menu.

`status.wifi.icon.enable`

1 (default) – Display the Wi-Fi icon on the status bar of the phone’s screen.

0 – Do not display the Wi-Fi icon on the status bar.

Phone Customization Parameters

Customize your phone using the following parameters.

Enhanced Feature Keys Parameters

The rules for configuring EFK for line keys, softkeys, and hard keys vary.

Note: You can include configuration file changes and enhanced feature key definitions in one configuration file. However, Poly recommends creating a new configuration file to make configuration changes.

Before configuring EFK, refer to Macro Definitions to become familiar with the macro language.

See the following list for the parameters you can configure and a brief explanation of how to use the contact directory to configure line keys.

`feature.enhancedFeatureKeys.enabled`

0 (default) - Disables the enhanced feature keys feature.

1 - Enables the enhanced feature keys feature.

`feature.EFKLineKey.enabled`

0 (default) – Does not allow configuring EFK as a line key.

1 - Allows configuring EFK as a line key.

Before you enable this parameter, set the parameter `feature.enhancedFeatureKeys.enabled` to 1.

`efk.efklist.x.action.string`

The action string contains a macro definition of the action that the feature key performs.

Null (default)

String (maximum of 64 characters)

If you enable EFK, this parameter must have a value (it cannot be Null).

For a list of macro definitions and example macro strings, see Macro Definitions.

Change causes system to restart or reboot.

`efk.efklist.x.label`

The text string used as a label on any user text entry screens during EFK operation.

Null (default) - Uses the Null string.

String (maximum of 64 characters)

If the label does not fit on the screen, the text shortens and appends with '…'.

Change causes system to restart or reboot.

`efk.efklist.x.mname`

The unique identifier used by the speed dial configuration to reference the enhanced feature key entry. Cannot start with a digit. Note that this parameter must have a value, it cannot be Null.

expanded_macro (default)

String (maximum of 64 characters)

Change causes system to restart or reboot.

`efk.efklist.x.status`

0 (default) - Disables the key x.

Null - Disables the key x.

1 - Enables the key x.

Change causes system to restart or reboot.

`efk.efklist.x.type`

Defines the SIP method.

Invite (default) - Performs the required action using the SIP INVITE method.

Null - default of INVITE is used.

This parameter is included for backwards compatibility. Do not use if possible. If `efk.x.action.string` contains types, this parameter is ignored.

Change causes system to restart or reboot.

`efk.efkprompt.x.label`

The prompt text on the user prompt screen.

Null (default) - No prompt displays.

String

If the label does not fit on the screen, the label shortens and '…' appends.

Change causes system to restart or reboot.

`efk.efkprompt.x.status`

This parameter must have a value. It cannot be Null.

0 (default) - Disables the EFK prompt.

1 - Enabled the EFK prompt.

If a macro attempts to use a prompt that is disabled or invalid, the macro execution fails.

Change causes system to restart or reboot.

`efk.efkprompt.x.type`

The type of characters entered by the user.

text (default) - The phone interprets characters as letters.

numeric - The phone interprets characters as numbers.

If Null, `numeric` is used. If this parameter has an invalid value, this prompt, and all parameters depending on this prompt, are invalid.

Note: A mix of `numeric` and `text` is not supported.

Change causes system to restart or reboot.

`efk.efkprompt.x.userfeedback`

The user input feedback method.

visible (default) - The text is visible.

masked - The text displays as asterisk characters (*). You can use this to mask password fields.

If this parameter has an invalid value it and all dependent parameters are invalid.

Change causes system to restart or reboot.

`efk.version`

The version of the EFK elements. This parameter is not required if there are no `efk.efklist` entries.

2 (default) - Supported version for SIP 3.1 and later.

1 - Supported version for or SIP 3.0.x or earlier.

Null - Disables the EFK feature.

Change causes system to restart or reboot.

`efk.softkey.alignleft`

Use this parameter to left-align softkeys and remove blank softkeys from the order.

0 (default)

1 - Left-aligns softkeys and removes blank softkeys from the order

Note: This parameter doesn't work with custom softkeys.

Change causes system to restart or reboot.

`efk.efklist.x.lineLabel`

Specifies EFK line key label.

ALL (default)

Change causes system to restart or reboot.

Flexible Line Keys Parameters

Line keys that you configure using this feature override the default line key assignments as well as any custom line key configurations you may have made.

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

`lineKey.reassignment.enabled`

Enable to specify at least two calls per line key.

0 (default) - Disabled

1 - Enabled

`lineKey.x.category`

Specify the line key category.

Line

BLF

EFK

SpeedDial

Presence

`lineKey.x.index`

Specify the line key number (dependent on category).

0 (default) - The index value for BLF or presence.

0- 9999

`up.staticBLF.FLKIndexRequired`

Enable to specify the BLF index for lineKey.x.index on BLF line keys. If set to 0, the phone determines where to place the BLF line keys.

0 (default) – Disabled

1 - Enabled

`up.EFK.FLKIndexRequired`

Enable to specify the EFK index for lineKey.x.index on EFK line keys. If set to 0, the phone determines where to place the EFK line keys.

0 (default) – Disabled

1 - Enabled

Microbrowser and Web Browser Parameters

You can configure the microbrowser and web browser to display a non-interactive web page on the phone's idle screen, and you can specify an interactive home web page that users can launch in a web browser.

The parameters listed below configure the home page, proxy, and size limits used by the microbrowser and browser when selected to provide services.

`apps.push.alertSound`

Enable for the phone to chime a sound when an alert is pushed.

0 (default) - Disabled

1 - Enabled

`apps.push.messageType`

Choose a priority level for push messages from the application server to the phone.

0 (None) - (default) - Discard push messages

1 (Normal) Allows only normal push messages

2 (Important) Allows only important push messages

3 (High) Allows only priority push messages

4 (Critical) Allows only critical push

5 (All) Allows all push messages

`apps.push.password`

The password to access the push server URL. To enable the push functionality, you must set values for the parameters apps.push.username and apps.push.password (not null).

NULL (default)

string

`apps.push.secureTunnelEnabled`

Enable to allow the connection to the web server to use a secure tunnel.

1 (default) - Enabled

0 - Disabled

`apps.push.secureTunnelPort`

Specify the port the phone uses to communicate to the web server when the secure tunnel is used.

443 (default)

1 - 65535

`apps.push.secureTunnelRequired`

Enable for communications to the web server require to require a secure tunnel.

1 (default) - Enabled

0 - Disabled

`apps.push.serverRootURL`

The URL of the application server you enter here is combined with the phone address and sent to the phone's browser. For example, if the application server root URL is `http://172.24.128.85:8080/sampleapps`

and the relative URL is / `examples/sample.html` , the URL sent to the microbrowser is `http://172.24.128.85:8080/sampleapps/examples/sample.html` . You can use HTTP or HTTPS.

NULL (default)

URL

`apps.push.username`

The user name to access the push server URL. To enable the push functionality, you must set values for the parameters `apps.push.username` and `apps.push.password` (not null).

NULL (default)

string

`apps.statePolling.password`

Enter the password that the phone requires to authenticate phone state polling.

NULL (default)

string

`apps.statePolling.responseMode`

1 (default) - Polled data you request is sent to a configured URL.

0 - Polled data is sent in the HTTP response.

`apps.statePolling.URL`

The URL to which the phone sends call processing state/device/network information. The protocol used can be either HTTP or HTTPS. Note: To enable state polling, the parameters `apps.statePolling.URL` , `apps.statePolling.username` , and `apps.statePolling.password` must be set to non-null values.

NULL (default)

string

`apps.statePolling.username`

Enter the user name that the phone requires to authenticate phone state polling.

NULL (default)

string

`apps.telNotification.appInitializationEvent`

0 (default) - No telephony notification event is sent.

1 - An XML telephony notification event is sent to report that the phone has completed initialization of its primary UC Software application. This event typically means that the phone is available and ready to receive network requests even if the phone user interface is not yet available.

`apps.telNotification.callStateChangeEvent`

0 (default) - Call state change notification is disabled.

1 - Call state notification is enabled.

`apps.telNotification.incomingEvent`

0 (default) - Incoming call notification is disabled.

1 - Incoming call notification is enabled.

`apps.telNotification.lineRegistrationEvent`

0 (default) - Line registration notification is disabled.

1 - Line registration notification is enabled.

`apps.telNotification.networkUpEvent`

0 (default) - No telephony notification event is sent.

1 – An XML telephony notification event is sent to report that the phone has received link up state from its LAN port and that an IP address was assigned.

`apps.telNotification.offhookEvent`

0 (default) - Disable off-hook notification.

1 - Enable off-hook notification.

`apps.telNotification.onhookEvent`

0 (default) - Disable on-hook notification.

1 - Enable on-hook notification.

`apps.telNotification.outgoingEvent`

0 (default) - Disable outgoing call notification.

1 - Enable outgoing call notification.

`apps.telNotification.taInitializationEvent`

0 (default) – No telephony notification event is sent.

1 - An XML telephony notification event is sent to report that the phone has started its test automation server and is ready to receive API commands.

`apps.telNotification.uiInitializationEvent`

0 (default) - No telephony notification event is sent.

1 - An XML telephony notification event is sent to report that the phone has completed start up of the phone user interface and is ready to receive physical key or touch inputs.

`apps.telNotification.URL`

The URL to which the phone sends notifications of specified events. You can use HTTP or HTTPS.

NULL (default)

string

`apps.telNotification.userLogInOutEvent`

Enable or disable the user login/logout notification.

0 (default) - Disabled

1 - Enabled

`apps.telNotification.x.URL`

Additional URLs to which the phone sends notifications of specified events, where x 1 to 9. You can use HTTP or HTTPS.

NULL (default)

string

`mb.main.home`

Specifies the URL of the microbrowser's home page. For example: `http://www.example.com/xhtml/frontpage/home` .

Null (default)

valid HTTP URL, String (maximum 255 characters)

`mb.main.idleTimeout`

Specifies the timeout in seconds for the interactive browser. If set to 0, there is no timeout.

40 (default)

0 - 600

`mb.main.loadWebImages`

Enable to allow images to load in the web browser.

1 (default) - Enabled

0 - Disabled

`mb.main.proxy`

Specifies the address of the HTTP proxy to be used by the microbrowser.

Null (port: 8080) (default)

domain name or IP address in the format <address>:<port>

`mb.main.reloadPage`

0 (default) - The microbrowser displays the content of the most recently viewed web page

1 - The microbrowser loads the URL configured in `mb.main.home` each time the browser is launched.

`mb.main.statusbar`

0 (default) - The status bar does not gets displayed.

1 - The status bar and status messages are displayed.

`mb.main.toolbar.autoHide.enabled`

1 (default) - The toolbar is not displayed.

0 - The toolbar displays continuously.

`mb.proxy`

Specify the Application browser home page, a proxy to use, and size limits.

Phone Keypad Parameters

You can configure phone keys in the following ways:

- Assign a function or feature to a key
- Turn a phone key into a speed dial
- Assign enhanced feature key (EFK) operations to a phone key
  For example, you can map a phone menu path to a single key press using a macro code. See Enhanced Feature Keys.
- Delete all functions and features from a phone key

Use the parameters in the following list to change the layout of your phone's keypad.

`key.x.function.prim`

Set the primary key function for key x.

Null (default)

String (maximum of 255 characters)

`key.x.subPoint.prim`

Set the secondary key function for key x.

Null (default)

String (maximum of 255 characters)

Feature Softkey Customization Parameters

You can enable or disable the default feature soft keys.

Note: The parameter `feature.enhancedFeatureKeys.enabled` must be enabled (set to 1) to use the Configurable softkey feature.

`softkey.feature.basicCallManagement.redundant`

Displays the Hold and Transfer softkeys.

1 (default) - Enabled

0 - Disabled

`softkey.feature.buddies`

Enable to display the Buddies softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.callers`

Enable to display the Callers softkey for all platforms.

1 - Enabled

0 (default) - Disabled

`softkey.feature.directories`

Enable to display the Directories (Dir) softkey.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`softkey.feature.doNotDisturb`

Enable or disable the Don’t Disturb (DND) softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.endcall`

Displays the End Call softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.forward`

Enable to display the Forward softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.join`

Enable to display the Join softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.menu`

Display the menu soft key/button on the idle screen.

0 (default) - Disabled

1 - Enabled

`softkey.feature.mystatus`

Enable to display the MyStatus softkey. The `pres.idleSoftKeys` parameter must be set to 1.

1 (default) - Enabled

0 - Disabled

`softkey.feature.newcall`

Enable to display the New Call softkey.

1 (default) - Enabled

0 - Disabled

`softkey.feature.redial`

Enable to display the Redial softkey. The parameter `feature.enhancedFeatureKeys.enabled` must be set to 1 first to configure this feature, and the parameter `efk.softkey.alignleft` must be set to 1 to move enabled softkeys into the positions of disabled softkeys.

1 - Enabled

0 (default) - Disabled

`softkey.feature.split`

Enable to display the Split softkey. The Split softkey allows you to split conference calls into individual calls.

1 (default) - Enabled

0 - Disabled

Disabling Default Soft Keys

You can disable the display of any of the following default soft key to make room for custom soft keys:

- New Call
- End Call
- Split
- Join
- Forward
- Directories
- MyStatus and buddies

- Hold, transfer, and conference

Example: Transfer Call to Broadsoft Voicemail

Use the following example configuration to automatically transfer an active call to a BroadSoft voicemail.

In this example, *55 is the star code for BroadSoft voicemail, and 8545 is the extension of the voicemail line the call transfers to. The exact star code to transfer the active call to voicemail depends on your call server.

Enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When you press the soft key—labeled VMail in this example—the call is placed on hold and automatically transferred to a BroadSoft voicemail.

Task

1 Update the configuration file as follows:

   - `softkey.1.label="VMail"`
   - `softkey.1.action="$FTransfer$$Cpause1$$FDialpadStar$$FDialpad5$$FDialpad5$ $FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$$FSoftKey1$"`
   - `softkey.1.enable="1"`
   - `softkey.1.use.active="1"`

2 Reboot the phone.

   When an incoming call connects and becomes active, the VMail soft key displays.

Example: Send-to-Voicemail Prompt

Use the following example to enable users to enter a voicemail extension to transfer an active call to BroadSoft voicemail.

In this example, *55 is the star code used for BroadSoft voicemail. The exact star code to transfer the active call to voicemail depends on your call server.

Enabling the parameter `softkey.1.use.active` causes the soft key to display when a call becomes active on the line. When a user presses the soft key, the call is placed on hold and a field prompts the user to enter the extension of a voicemail line to transfer the call to. The `efk.prompt*` parameters control the numeric prompt field users enter the extension into.

Note that this example works only on line 1 of the phone.

Task

1 Update the configuration file as follows:

   - `softkey.1.label="VMail"`
   - `softkey.1.action="^*55$P1N10$$Tinvite$"`
   - `softkey.1.enable="1"`
   - `softkey.1.use.active="1"`
   - `efk.efkprompt.1.label="Voice Mail"`
   - `efk.efkprompt.1.status="1"`
   - `efk.efkprompt.1.type="numeric"`

2 Reboot the phone.

   When an incoming call connects and becomes active, the VMail soft key displays.

3 Press the **VMail** soft key.

   A field displays prompting you to enter an extension.

4 Enter the extension you want to transfer the call to.

5 Press the **Enter** soft key.

Example: Speed Dial Soft Key with a Pause

Use the following example to configure a soft key to automatically dial a number with a pause in the dialing sequence.

In this example, use `$CpauseX$` where `X` is the number of seconds to pause—7 in this example. Adding this pause function enables users to automatically dial into a conference ID that requires an entry code after the conference call is connected.

Task

» Update the configuration file as follows:

  - `softkey.1.label="VMail"`
  - `softkey.1.action="$S1$$Tinvite$$Cwc$$Cpause7$$FDialpad8$$FDialpad5$$FDialpad4$ $FDialpad5$"`
  - `softkey.1.enable="1"`
  - `softkey.1.use.idle="1"`
  - `feature.enhancedFeatureKeys.enabled="1"`

The values for this example are explained as follows:

- `$S1$`— Speed dial line 1
- `$S1$$Tinvite$$` —The phone sends an invite to $S1$
- `$Cwc$` —The phone waits for the call to connect
- `$Cpause7$` —The phone waits for 7 seconds before dialing the remaining numbers
- `$FDialpad8$$FDialpad5$$FDialpad4$$FDialpad5$` —The phone enters the entry code 8545.

Example: Directory-Linked Speed Dial Soft Key with a Pause

Use the following example to add a speed dial line key linked to a directory file with a pause in the dialing sequence.

Task

1 Update the configuration file as follows:

  - `feature.enhancedFeatureKeys.enabled="1"`
  - `efk.efklist.1.action.string="501$Tinvite$$Cwc$$Cpause7$1234#$Tdtmf$"`
    `efk.efklist.1.label="number"`
  - `efk.efklist.1.mname="number"`
  - `efk.efklist.1.status="1"`

2 In a contact directory file or speed dial file (000000000000-directory.xml or <MACaddress>-directory.xml), add the following:

  - `<fn>Call Number</fn>`
  - `<ct>!number</ct>`
  - `<sd>99</sd>`

The following values are included in the action string: `<ct>"501$Tinvite$$Cwc$$Cpause7$1234#$Tdtmf$":`

- `501$Tinvite$`—Dial 501
- `$Cwc$`—Wait for the call to connect
- `$Cpause7$`—A seven second pause
- `1234#$Tdtmf$`—Send 1234 dual-tone multi-frequency

The following EFK commands are linked to the directory file:

- The parameter `efk.efklist.1.mname="number"`is linked to the speed dial contact `<ct>!number</ct>` of the directory file
- Use `<fn>Call Number</fn>` to define the name that displays on the key
- Use `<sd>99</sd>` to identify which directory entry to link to the key

> **Note:**
> For more example configurations, see the two following documents at Poly Support:

- *Using Enhanced Feature Keys and Configurable Soft Keys on Polycom Phones: Technical Bulletin 42250*
- *Using Enhanced Feature Keys (EFK) Macros to Change Soft Key Functions on Polycom Community: Feature Profile 42250*

Softkey Parameters

You can create up to 10 custom soft keys.

If you configure more soft keys than what can fit on the phone's screen, a More soft key displays. Users can use the More soft key to display any additional soft keys available.

If you want the phone to display both default and custom soft keys, you can configure them in any order. However, the order in which soft keys display depends on the phone's menu level and call state. If you have configured custom soft keys to display with the default soft keys, the order of the soft keys may change.

Note: The Hold, Transfer, and Conference soft keys are grouped together to avoid usability issues. You may experience errors if you try to insert a soft key between these three grouped soft keys.

The following list includes the parameters for configuring soft keys. Note that this feature is part of enhanced feature keys (EFK), and you must enable the EFK parameters to configure soft keys. See the Enhanced Feature Keys section for details about configuring soft keys and line keys.

`feature.enhancedFeatureKeys.enabled`

0 (default) - Disables the enhanced feature keys feature.

1 - Enables the enhanced feature keys feature.

`softkey.x.action`

Controls the action or function for the custom soft key x.

Null (default)

macro action string, 2048 characters

This value uses the same macro action string syntax as an Enhanced Feature Key.

`softkey.x.enable`

0 (default) - The x soft key is disabled.

1 - The x soft key is enabled.

`softkey.x.insert`

0 (default) - The phone places the soft key in the first available position.

0 to 10 - The phone places the soft key in the corresponding position and moves the following soft keys by one position to the right.

For example, if the soft key is set to 3, the soft key is displayed in the third position from the left. If the soft key already exists in the third position, it is moved to fourth position and the following soft keys are moved to right by one space.

If `softkey.x.precede` is configured, this value is ignored. If the insert location is greater than the number of soft keys, the key is positioned last after the other soft keys.

`softkey.x.label`

The text displayed on the soft key label. If Null, the label is determined as follows:

- If the soft key performs an Enhanced Feature Key macro action, the label of the macro defined using `efk.efklist` is used.

- If the soft key calls a speed dial, the label of the speed dial contact is used.
- If the soft key performs chained actions, the label of the first action is used.
- If the soft key label is Null and none of the preceding criteria are matched, the label is blank.

Null (default)

String

Note: The maximum number of characters for this parameter value is 15; 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 used. Parameter values that exceed the phone's maximum display length are truncated by ellipses (…). The phone truncates the beginning of numerical labels (for example, …4567) and truncates the end of alphabetical labels (for example, Abcd…).

`softkey.x.precede`

0 (default) - The phone locates the soft key in the first available position from left.

1 - The phone locates the soft key before the default soft key position.

`softkey.x.use`

Specify which call states the soft key displays in.

`softkey.x.use.active`

0 (default) - Does not display the soft key x during an active call.

1 - Displays the soft key x during an active call.

`softkey.x.use.alerting`

0 (default) - Does not display the soft key x in an alerting state.

1 - Displays the soft key x in an alerting state.

`softkey.x.use.dialtone`

0 (default) - Does not display the soft key in the dial tone state.

1 - Displays the soft key x in the dial tone state.

`softkey.x.use.hold`

0 (default) - Does not display the soft key x in the hold state.

1 - Displays the soft key x in the hold state.

`softkey.x.use.idle`

0 (default) - Does not display the soft key x in the idle state.

1 - Displays the soft key x in the idle state.

`softkey.x.use.park`

0 (default) - Does not display the soft key x in the parked state.

1 - Displays the soft key x in the parked state.

`softkey.x.use.proceeding`

0 (default) - Does not display the soft key x in the proceeding state.

1 - Displays the soft key x in the proceeding state.

`softkey.x.use.setup`

0 (default) - Does not display the soft key x in the setup state.

1 - Displays the soft key x in the setup state.

`up.displayConferenceSoftkeyOnTransfer`

1 (default) - Displays the **Conference** softkey on the phone.

0 - Hides the **Conference** softkey on the phone.

`up.hotelingSignInMenu.displayModeSoftkey`

1 (default) - Displays the **Mode** softkey in the **Hoteling** menu on the phone.

0 - Hides the **Mode** softkey in the **Hoteling** menu on the phone.

Phone Display Option Parameters

Use the following parameters to configure phone display options including time and date, phone languages, line labels, and LED behaviors.

Active Call Screen Parameters

Use the parameters in the following list to set the default screen that displays when the phone is in call.

`up.LineViewCallStatus.enabled`

0 (default) - In an active call, the active call screen displays. Any incoming or outgoing call triggers the display of the active call screen.

1 - During an incoming call and in an active call, the line view displays and call details display on the status ribbon.

`up.LineViewCallStatus.timeout`

Specify the timeout period in seconds after which the phones go back to the Line Screen when the user goes to the Active Call Screen from the Line View.

10 (default)

2 - 10

Capture Your Device's Current Screen Parameters

Use the following parameters to get a screen capture of the current screen on your device.

`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 User Interface Color Parameters

You can configure custom colors for the user interface using the parameters in the following list.

`ui.home.background`

Set the color of the background of the Home screen.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.menu.background`

Set the background color of the Menu screen.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.menu.item.background`

Set the color of the background for menu items.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.menu.item.text.color`

Set the color of the text for menu items.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.menu.title.background`

Set the background color for the title of the Menu.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.softkey.background`

Set the background color for softkeys.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.softkey.text.color`

Set the color of the text for softkeys.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.statusBar.background`

Set the background color for the Status bar.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

`ui.statusBar.text.color`

Set the color of the text that displays on the Status bar.

Null (default)

#RRGGBB color codes

Change causes system to restart or reboot.

Digital Picture Frame Parameters

The parameters you can configure are included in the following list.

`feature.pictureFrame.enabled`

Enable or disable the digital picture frame.

1 (default) - Enabled

0 - Disabled

Change causes system to restart or reboot.

`up.pictureFrame.folder`

Path name for images.

NULL (Default) - Images stored in the root folder on the USB flash drive are displayed.

string - 0 to 40 characters

Example: If images are stored in the /images/phone folder on the USB flash drive, set this parameter to `images/phone` .

`up.pictureFrame.timePerImage`

Specify the number of seconds to display each picture frame image before moving to the next picture.

5 (Default)

3-300

Digital Phone Label Parameters

You can create a short personal message to display in the status bar on the phone's screen.

`lcl.status.LineInfoAtTop`

Enable or disable the text set in `lcl.status.LineInfoAtTopText` to display on the phone screen

0 (default) - Disabled

1 - Enabled

`lcl.status.LineInfoAtTopText`

Provides the text be displayed on the phones screen. Up to 14 digits is allowed. The use of characters is permitted but might lead to truncation.

Null (default)

string

Note: You must enable `lcl.status.LineInfoAtTop` to configure this parameter.

Graphic Display Background Parameters

The configured background image displays across the entire phone screen, and the time, date, line and key labels display over the background.

If you want the background image to display more visibly from behind line key labels, use `up.transparentLines` to render line key labels transparent.

Use the parameters in the following list to configure graphic display background.

`bg.background.enabled`

Enable or disable the ability for users to set a custom background image on the phone screen. If enabled, options for customization are available on the phone screen and in the Web Configuration Utility for users.

0 - Disabled

1 (default) - Enabled

`bg.color.bm.x.name`

Specify the name of the phone screen background image file including extension with a URL or file path of a BMP or JPEG image.

Note: If the file is missing or unavailable, the built-in default solid pattern is displayed.

Hide MAC Address Parameters

The following list includes parameters that configure the display of MAC address.

`device.mac.hide`

0 (default) - MAC address displays.

1 - MAC address is hidden.

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.

LED Pattern Parameters

The LED pattern parameters listed in the following list configure the pattern state, color, and duration of the LED indicators and the pattern types on Poly devices.

For each parameter, specify x, y, and a permitted value:

- Specify an LED pattern using the LED pattern parameters.
- For x, specify an LED pattern type.
- For y, specify the step in the LED pattern with a number between 1-20.

Use the parameters in the following list to set the pattern state, color, and duration of the LED indicators.

`ind.pattern.x.step.y.state`

Depending on the pattern type, the default value varies.

0 - Turn off the LED indicator.

1 - Turn on the LED indicator.

`ind.pattern.x.step.y.color`

Specify the color of the LED indicator. Depending on the pattern type, the default value varies.

Red

Green

Yellow

`ind.pattern.x.step.y.duration`

Specify the duration of the pattern in milliseconds. Depending on the pattern type, the default value varies.

0 (default)

0 - 32767

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 - 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 - Displays date above time.

0 (default) - Displays date below time.

Note: Overrides the `lcl.datetime.date.dateTop` parameter.

`lcl.ml.lang.clock.x.format`

"D,dM" (The default value for each language varies.)

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 - Displays the day and month in long format (Friday/November).

0 (default) - 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)

Comma-separated string of language indexes.

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

Off-Hook Phone Screen Parameters

Use the following parameters to enable and configure the phone's off-hook behavior.

`up.OffHookLineView.enabled`

Enable or disable the Lines view as the default screen to display when the phone is off-hook.

0 (Default) - The dialer screen is the default view when the phone is off-hook.

1 - The Lines view is the default view when the phone is off-hook.

`up.OffHookLineView.inCallActions.enabled`

Enable or disable the Lines view as the default screen to display when you modify an active call using a transfer, consult, or conference.

0 (Default) - The dialer screen is the default view when you modify an active call.

1 - The Lines view is the default view when you modify an active call.

`up.offHookSpeedDialShortcut.enable`

1 (default) - Displays the speed dial shortcut for one or two digits followed by # in the off-hook state.

0 – Does not display the speed dial shortcut for one or two digits followed by # in the off-hook state.

Pagination Configuration Parameters

Use the following parameter to configure pagination on the phone.

`up.Pagination.enabled`

Enable the pagination feature.

0 (default) - Disabled.

1 - Enable.

`up.smartPagination.enabled`

Enable smart pagination to skip empty pages.

0 (default) - Disabled.

1 - Enable.

Requires `up.Pagination.enabled` to be enabled.

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

Home Screen Parameters

Use the parameters in the following list to configure the phone's **Home** screen.

`feature.preferredHomeScreen.enabled`

Enable whether the preferred home screen can be set.
1 (default) - The preferred home screen can be set.

0 - The preferred home screen can't be set, and the phone uses the default screen.

`feature.preferredHomeScreen`

Set the preferred home screen.
default (default) - The home screen displays the current time, date, and icons for configured phone features.

line - The **Lines** page displays as the phone's home screen.

meeting - The **Meetings** page displays as the phone's home screen.

`homeScreen.application.enable`

1 (default) - Enable display of the **Applications** function on the main menu.

0 - Disable display of the **Applications** function on the main menu.

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,Md" (default)

String

`lcl.datetime.date.longFormat`

1 - Displays the day and month in long format (Friday/November).

0 (default) - Displays the day and month in abbreviated format (Fri/Nov).

`lcl.datetime.time.24HourClock`

1 - Displays the time in 24-hour clock mode.

0 (default) - 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

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)
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

Permitted Value	Time Zone Description
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
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

Permitted Value	Time Zone Description
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
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

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.

Security Parameters

The following parameters are for security.

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.

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 **Advanced** 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*.

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`

-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.

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.

`feature.usb.host.enabled`

Disable unused USB ports to increase device security.
0 - Disable the USB ports.

1 (default) - Enable the USB ports.

`feature.usb.host.massStorage`

Enable or disable USB mass storage devices.
0 - Disable USB mass storage devices.

1 (default) - Enable USB mass storage devices.

`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.

`call.autoAnswerMenu.enable`

1 (default) - Enables the phone's Autoanswer menu.

0 - Disables the phone's Autoanswer menu.

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.

`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.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.

Web Configuration Utility Lock Parameters

Use the following parameters to configure how the Web Configuration Utility will behave after failed login attempts.

`httpd.cfg.lockWebUI.enable`

1 (default) - Enable the Web Configuration Login Lock feature.

0 - Disable the Web Configuration Login Lock feature.

`httpd.cfg.lockWebUI.lockOutDuration`

60 seconds (default) - The period of time the user is locked out of the Web Configuration Utility. The user can try logging in again after this time.

60 - 300 seconds

The lock-out timer starts after the maximum number of unsuccessful attempts within the duration you configure. After the lock-out time has expired, the timers and the number of incorrect attempts resets to 60 seconds.

httpd.cfg.lockWebUI.noOfInvalidAttempts

5 (default) - After five failed login attempts, the user is locked out of the Web Configuration Utility.

Specify the maximum number of failed login attempts after which the user is locked out of the Web Configuration Utility.

3 - 20 seconds

httpd.cfg.lockWebUI.noOfInvalidAttemptsDuration

60 seconds (default) - After a user reaches the maximum failed login attempts within 60 seconds, the user is locked out of the Web Configuration Utility.

After a user reaches the maximum failed login attempts within this time duration, the user is locked out of the Web Configuration Utility. The user can try logging in again after the lock-out duration set by `httpd.cfg.lockWebUI.lockOutDuration` .

60 - 300 seconds

The timer starts again after the first incorrect password attempt.

Web Configuration Utility Security Banner Parameters

The following list includes the parameters of the web user interface for security banner parameters.

feature.webSecurityBanner.enabled

0 (default) - No security banner message displays on the phone's web user interface.

1 - A security banner with the configured message displays phone's web user interface. Use feature.webSecurityBanner.msg to configure the message.

feature.webSecurityBanner.msg

Customize the text in security banner.

"This is default text for the security log-on banner" (default) - This text displays because the security log-on banner has been enabled and the custom text to be displayed in the security log-on banner has not been configured.

2000 characters (maximum)

Web Proxy Parameters

Use the following parameters to configure web proxy on your phone.

feature.wpad.enabled

0 (default) - The phone doesn't authenticate with a web proxy server.

1 - The phone authenticates with the web proxy server defined by `feature.wpad.proxy`.

feature.wpad.proxy

Defines the web proxy server address.

Null (default)

0 to 255 characters

`feature.wpad.basicAuth.enabled`

0 (default) - Basic web proxy authentication is disabled.

1 - Basic web proxy authentication is enabled. The phone provides a username and password when authenticating with a web proxy server. Set the user name with `feature.wpad.proxy.username` and the password with `feature.wpad.proxy.password`.

Note: As of PVOS 8.0.0, the following parameter is deprecated:

```
feature.wpad.basicAuth.enabled
```

`feature.wpad.proxy.username`

Configures the username for basic web proxy authentication.

Null (default)

0 to 255 characters

`feature.wpad.proxy.password`

Configures the password for basic web proxy authentication.

Null (default)

0 to 255 characters

Shared Lines Parameters

Use the following parameters to configure shared lines features on your phones.

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.

`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.

Intercom Calls Parameters

Use the parameters in the table to configure the behavior of the calling and answering phone.

feature.intercom.enable

0 (default) - Disable the Intercom feature.

1 - Enable the Intercom feature.

softkey.feature.intercom

1 (default) - Enables the Intercom soft key.

0 - Disables the Intercom soft key.

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

voIpProt.SIP.intercom.alertInfo.encapsulateWithAngleBrackets

Encapsulate Alert-Info header in angular brackets ("<" and ">") for improved processing of intercom calls.

0 (default) - The value of `voIpProt.SIP.intercom.alertInfo` is used verbatim.

For example:

```
Alert-Info: intercom
```

1 - The value of `voIpProt.SIP.intercom.alertInfo` is encapsulated with angle brackets.

For example:

```
Alert-Info: <intercom>
```

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.

`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.

Push-to-Talk Parameters

Administrators must enable group paging and PTT before users can subscribe to a PTT channel.

PTT works in conjunction with group paging, and you can enable PTT or group paging, or enable both to operate simultaneously.

`ptt.pttMode.enable`

Enable or disabled push-to-talk.

0 (default) - Disabled

1 - Enabled

`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.allowOffHookPages`

Enable to allow PTT messages to play on the phone while it is in an active call.

0 (default) - Disabled. The user must accept incoming PTT messages to play out.

1 - Enabled

`ptt.callWaiting.enable`

Enable to allow call waiting when incoming PTT calls come through on active audio channels.

0 (default) - Disabled

1 - Enabled

`ptt.channel.x.allowReceive`

Enable channel x to receive PTT calls.

1 (default) - Enabled

0 - Disabled

`ptt.channel.x.allowTransmit`

Enable outgoing PTT calls on channel x.

1 (default) - Enabled

0 - Disabled

`ptt.channel.x.available`

1 (default) - Channel x is available.

0 - Channel x is not available.

`ptt.channel.x.label`

Specify a label for channel x.

Null (default)

string

`ptt.channel.x.subscribed`

`ptt.channel.1.subscribed` through `ptt.channel.25.subscribed` are available.

0 (default) - The PTT is not subscribed for channel x.

1 - The PTT is subscribed for channel x.

`ptt.codec`

Specify codec to use for PTT.

G.722 (default)

G.711Mu

G.726QI

G.722

`ptt.defaultChannel`

Specify the default channel number used for PTT transmissions.

1 (default)

1 - 25

`ptt.emergencyChannel`

Specify the channel to use for emergency PTT transmissions.

25 (default)1 - 25

`ptt.emergencyChannel.volume`

Set the emergency page audio volume relative to the maximum speakerphone volume of the phone. Positive values are louder than the maximum and negative values are quieter. The gain to use for emergency page/PTT is the maximum termination gain plus this parameter. Note: To enter a negative number, press the * key first.

-10 (default)

-57 - 0

`ptt.port`

Specifies the port values to send and receive audio.

5001 (default)

0 to 65535

`ptt.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.payloadSize`

Specify the payload size for PTT transmissions.

20 (default)

10

30

40

50

60

70

80

`ptt.priorityChannel`

Specify the channel number to use for priority PTT transmissions.

24 (default)

1 - 25

`ptt.volume`

Controls the volume level for pages without changing the volume level for incoming calls.

-20 (default)

-57 to 0

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 for SIP line registration x.

0 - Enables call diversion features on shared lines for SIP line registration x.

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.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.filterReflectedBlaDialogs`

1 (default) - bridged line appearance NOTIFY messages are ignored.

0 - bridged line appearance NOTIFY messages is not ignored

`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.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.server.y.expires.lineSeize`

Requested line-seize subscription period.

30 - (default)

0 to 65535

`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.

SIP-B Automatic Call Distribution Parameters

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

`feature.acdLoginLogout.enabled`

Enable or disable the ACD login/logout feature.

0 (default) - Disabled

1 - Enabled

Change causes system to restart or reboot.

`reg.x.acd-login-logout`

0 (default) - The ACD feature is disabled for registration.

1 - ACD feature is enabled for registration.

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.acd-agent-available`

0 (default) - The ACD feature is disabled for registration.

1 - ACD feature is enabled for registration.

If both ACD login/logout and agent available are set to 1 for registration x, the ACD feature is enabled for that registration.

`voIpProt.SIP.acd.signalingMethod`

0 (default) - The 'SIP-B' signaling is supported. (This is the older ACD functionality.)

1 - The feature synchronization signaling is supported. (This is the new ACD functionality.)

Change causes system to restart or reboot.

`acd.simplifiedAgentStateControl`

0 (default) - Displays menu items.

1 - Hides ASignIN and associated soft keys. Also hides menu items under **Menu > Settings > Feature > ACD**.

System Log Parameters

Use the following parameters to configure phone event logging parameters.

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.

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`.

The logging level set here determines the lowest severity log level that can be logged for each sub-system component.

`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)

`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.

Logging Render Parameters

The following list includes parameters for configuring logging features.

`log.render.file`

When you enable this option, the phone first writes log files directly into its flash memory. The contents of the flash memory then upload to a provisioning server after a predetermined period of time or when the flash memory becomes full.

1 (default) - The phone uploads the log file content to the server.

0 - The phone prevents uploading the log file content to the server.

Note: Poly recommends that you prevent the ability to upload log files only when necessary to reduce data traffic when the phone starts or reboots.

`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.realtime`

Poly recommends that you do not change this value.

1 (default) - Enable

0 - Disable

`log.render.stdout`

Poly recommends that you do not change this value.

0 - Disable

1 (default) - Enable

`log.render.type`

Refer to the Event Timestamp Formats table for timestamp type.

2 (default)

0 = HMSms (Number of seconds since boot up)

1 = YMDHm (Year Month Day Hour minute)

2 = MDHms (Month Day Hour millisecond)

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

Scheduled Logging Parameter

Scheduled logging can help you monitor and troubleshoot phone issues.

Use the parameters in this list to configure scheduled logging.

You must contact Poly Customer Support to obtain the template file `techsupport.cfg` containing parameters that configure scheduled logging.

`log.sched.x.name`

Configure the number of debug commands you want to schedule an output for. You can configure 1-10 debug commands per phone. Set the number of debug commands as x.

If x = 1, the default command name is 'showCpuLoad'.

9 (default)

If x = 2, the default command name is 'showBatteryStat'.

22 (default)

3 - 10 = No default value

The following are permitted values:

NULL

memShow

checkStack

cameraLogShow

ls

ifShow

ifShowVerbose

showProcesses

showCpuUsage

showCpuLoad

ethBufPoolShow

sysPoolShow

netPoolShow

netRxShow

endErrShow

routeShow

netCCB

arpShow

fsShow

ipStatShow

udpStatShow

sipPrt

showBatteryStat

If you encounter any camera related issue, set the `log.sched.x.name` value to cameraLogShow where x = 1 or 2 and set `log.level.change.slog=2`.

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.xxx`

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**.

4 (default)

0 - 6

Possible values for xxx are acom, ares, app1, 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, 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.

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 Service Provider Parameters

Use the following parameters to configures features for third-party service providers.

Broadsoft Parameters

Use the following parameters to configure your phones with the Broadsoft call features.

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.

BroadWorks Call Decline Parameter

Use the parameter below to enable users to reject calls on a shared line.

`call.shared.reject`

For shared line calls on the BroadWorks server.

0 (default) - The Reject soft key does not display.

1 - The phone displays a Reject soft key to reject an incoming call to a shared line.

Flexible Seating Parameters

Use the following parameters to configure Flexible Seating.

`hoteling.reg`

1 (default) - Specifies the phone line on the host phone which hosts the guest line.

`hotelingMode.type`

-1 (Default): The parameter does not exist on the BroadSoft server.

0 - Both Flexible Seating and Hoteling are disabled on the BroadSoft Device Management Server (DMS).

1 - Hoteling is enabled

2 - Flexible Seating is enabled but guest is not logged in.

3 - Flexible Seating location is enabled and guest is logged in.

Note: This parameter overrides `voIpProt.SIP.specialEvent.checkSync.downloadDirectory` when set to 2 or 3.

Executive-Assistant Parameters

Use the following list of configuration parameters to enable and configure the Executive-Assistant feature.

In the BroadWorks Web Portal, you must enable the Executive Service for private and shared executive lines, and the Executive-Assistant Service for private and shared assistant lines.

The BroadWorks server allows the following configuration options: Executive private line, Executive-Assistant Service line, and a shared alias line. Administrators can set up executive and assistant lines in the following scenarios:

- A private executive line with an assistant with a private line
- Shared executive line with an assistant with a private line
- Shared executive line with a shared line alias on the assistant's phone
  - The shared line must be created as a shared location of a line with the Executive Service on the BroadWorks server.
  - In this option, the main line registration is a private line for the assistant, and the secondary registration is a shared line for the executive.

`feature.BSExecutiveAssistant.enabled`

0 (default) - Disables the BroadSoft Executive-Assistant feature.

1 - Enables the BroadSoft Executive-Assistant feature.

`feature.BSExecutiveAssistant.regIndex`

The registered line assigned to the executive or assistant for the BroadSoft Executive-Assistant feature.

1 (default) to 255 - The registered line for the Executive or Assistant.

`feature.BSExecutiveAssistant.userRole`

ExecutiveRole (default) - Sets the registered line as an Executive line.

AssistantRole - Sets the registered line as an Assistant line.

Note: A phone can only have a line set as an Executive or an Assistant; an Executive and an Assistant line can't be on the same phone.

`feature.BSExecutiveAssistant.SimplifiedAssistant.enabled`

0 (default) - Displays the Pick Call and Barge-in soft keys in the Assistants menu on the phone.

1 - Removes the Pick Call and Barge-in soft keys from the Assistants menu on the phone.

`feature.BSExecutiveAssistant.SimplifiedExec.enabled`

0 (default) - Displays the Pick Call and Barge-in soft keys in the Executive menu on the phone.

1 - Removes the Pick Call and Barge-in soft keys from the Executive menu on the phone.

Enhanced Call Park Parameters

The following list includes the configuration parameters you can use to enable and configure this feature.

`reg.x.enhancedCallPark.enabled`

0 (default) - To disable the BroadWorks Enhanced Call Park feature.

1 - To enable the BroadWorks Enhanced Call Park feature.

`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

`feature.enhancedCallPark.allowAudioNotification`

0 (default) - Disables the audio notifications for parked calls on private and shared lines.

1 - Enables the audio notifications for parked calls on private and shared lines.

`call.parkedCallRetrieveString`

The star code that initiates retrieval of a parked call.

Null (default)

Permitted values are star codes.

BroadSoft Directory Parameters

To perform a search and to view contacts on the BroadSoft directories, configure the directories.

You can configure this feature using the parameters in the following list.

`feature.broadsoftGroupDir.enabled`

0 (default) - Disables Group Directory.

1 - Enables Group Directory.

`feature.broadsoftdir.enabled`

0 (default) - Disables Enterprise Directory.

1 - Enables Enterprise Directory.

Change causes system to restart or reboot.

`feature.broadsoftPersonalDir.enabled`

0 (default) - Disables Personal Directory.

1 - Enables Personal Directory.

Enterprise Directory Search Parameters

Use the following parameter to configure the Enterprise Directory Search feature.

`feature.broadsoftdir.showDefaultSearch`

0 (default) - No contacts are displayed when the search box field is empty.

1 - Enables the user to view the initial list of contacts for an empty search box.

BroadSoft Server-Based Call Logs Parameters

Use the following parameter to enable the BroadSoft server-based call logs feature.

`feature.broadsoft.callLogs`

Disabled (default) - Disable the BroadSoft server call logs feature.

Basic - Enable the BroadSoft server call logs feature.

BroadSoft Server-Based Redial Parameter

Use the following parameter to configure this feature.

`feature.broadsoft.basicCallLogs.redial.enabled`

0 (default) - Disables the option to redial the last number.

1 - Enables the phone to redial the last number.

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

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

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.

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 Server-based Call Waiting Parameter

Use the parameter below to configure server-based call waiting alerts.

`feature.broadsoft.xsi.callWaiting.enabled`

0 (default) - Disable incoming calls during an active call.

1 - Enable incoming calls during an active call.

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

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.

Hoteling Parameters

To enable Hoteling, you must configure Poly phones with the BroadSoft BroadWorks R17 platform.

You cannot use Hoteling in conjunction with the feature-synchronized automatic call distribution (ACD) feature and you must disable all ACD parameters to use the Hoteling feature. If both features are enabled at the same time, ACD take precedence and the Hoteling GuestIn/GuestOut soft keys do not display.

Use the parameters in the following list to configure Hoteling.

`feature.hoteling.enabled`

0 (default) - Enable Hoteling.

1 - Disable Hoteling.

`hoteling.reg`

Specify the line registration to use for Hoteling. You must disable the Automatic Call Distribution (ACD) feature and all ACD parameters to use Hoteling.

1 (default)

1 - 34

ACD Agent Availability Parameters

Use the parameters in this list to configure the ACD agent availability feature.

`feature.acdServiceControlUri.enabled`

Enable to display the **Trace**, **Emergency**, and **Disp Code** softkeys.

You must also enable the `feature.enhancedFeatureKeys.enabled` parameter to enable this parameter.

0 (default) - Disabled

1 - Enabled

`feature.acdLoginLogout.enabled`

Enable the ACD login/logout feature.

0 (default) - Disabled

1 - Enabled

`feature.acdPremiumUnavailability.enabled`

Enable the premium ACD unavailability feature.

0 (default) - Disabled

1 - Enabled

`voIpProt.SIP.acd.signalingMethod`

0 (default) - Support SIP-B signaling.

1 - Support the synchronization signaling feature.

`acd.UnavailableMacroReasonCodeMenu.enabled`

Enable to display the unavailable reason code menu for unavailable macros.

You must enable this parameter if you disable the `acd.defaultUnavailReasonCode.enabled` parameter.

0 (default) – Disabled

1 - Enabled

`feature.showRejectSoftKey.enable`

Disable to not display the **Reject** softkey for an incoming call.

0 - Disabled

1 (default) - Enabled

`reg.x.showRejectSoftKey`

Disable to not display the **Reject** softkey for an incoming call on the configured registered line.

0 - Disabled

1 (default) - Enabled

Note: If you configure both the `reg.x.ShowRejectSoftKey` parameter and the `feature.showRejectSoftKey.enable` parameter, then the value for `reg.x.ShowRejectSoftKey` takes precedence.

`acd.defaultUnavailReasonCode.enabled`

Disable to not display the reason code **None** in the unavailable reason code menu.

0 - Disabled

1 (default) - Enabled

Note: If you disable the `acd.defaultUnavailReasonCode.enabled` parameter, then you can't select the first item in the **Unavailable Reason Code** menu.

`voIpProt.SIP.copyUnknownHeaders`

Specify the comma separated header names.

Default ()

String - The total number of headers is 15, and maximum number of characters is 256.

Example: `voIpProt.SIP.copyUnknownHeaders="User-to-User,x_TFN,PraestoSF-ID"`

`acd.X.unavailreason.active`

Disable to make unavailable reason *X* not selectable.

X can be any number from 1 to 100. This number defines the order in which the reason is listed, from 1 to 100.

0 - Disabled

1 (default) - Enabled

`acd.X.unavailreason.codeValue`

String limit for the code value that displays in the list for unavailable reason *X*. There is no default for this parameter.

1 to 255 characters

`acd.X.unavailreason.codeName`

String limit for the unavailable reason description for unavailable reason *X*. There is no default for this parameter.

1 to 255 characters

`acd.X.unavailreason.isVisible`

Disable to not show the unavailable reason *X* to the user.

0 - Disabled

1 (default) - Enabled

Ribbon Communication Parameters

Use the following parameters to configure your phones with Ribbon Communications call features.

MADN-SCA Parameters

The following list includes all parameters available for configuring MADN-SCA and feature options.

Note:
If you configure the line-specific parameter `reg.x.server.y.address` , you must also configure values in the line-specific parameter `reg.x.server.y.specialInterop` .

If you configure the global parameter `voIpProt.server.x.address` , you must also configure values in the global parameter `voIpProt.server.x.specialInterop` .

For all deployments, including Ribbon Communications, line-specific configuration parameters override global configuration parameters. If you set values in both line-specific and global parameters, line-specific parameters are applied and global parameters are not applied.

`reg.x.address`

The user part (for example, 1002) or the user and the host part (for example, `1002@polycom.com`) of the registration SIP URI.

Null (default)

string address

`reg.x.server.y.specialInterop`

Specify the server-specific feature set for the line registration.

Standard (Default)

GENBAND

ALU-CTS

ocs2007r2

lcs2005

`voIpProt.server.x.specialInterop`

Enables server-specific features for all registrations.

Standard (default)

All other phones = Standard, GENBAND, GENBAND-A2, ALU-CTS, DT, ocs2007r2, lcs2005

`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.bargeInEnabled`

0 (default) - barge-in is disabled for line x.

1 - barge-in is enabled (remote users of shared call appearances can interrupt or barge in to active calls).

`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 `call.callsPerLineKey` .

24 (default)

1-24

`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.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.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.auth.domain`

The domain of the authorization server that is used to check the user names and passwords.

Null (default)string

`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).

Global Address Book Parameters

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

`feature.corporateDirectory.alt.enabled`

0 (default) - Disables the global address book service.

1 - Enables the global address book service.

`dir.corp.alt.address`

Enter the URL address of the GAB service provided by the server.

Null (default)

Hostname

FQDN

dir.corp.alt.port

Set the port that connects to the server if a full URL is not provided.

0 (default)

Null

1 to 65535

dir.corp.alt.user

Enter the user name used to authenticate to the Ribbon Communications server.

Null (default)

UTF-8 encoding string

dir.corp.alt.viewPersistence

Determine if the results from the last address directory search displays on the phone.

0 (default) - Disabled

1 - Enabled

dir.corp.alt.attribute.x.filter

Enter a filter to use to set a predefined search string through configuration files.

Null (default)

UTF-8 encoding string

dir.corp.alt.attribute.x.sticky

0 (default) – the filter string criteria for attribute x is reset after a reboot.

1 – the filter string criteria is retained through a 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.

dir.corp.alt.attribute.x.label

Enter a label to identify a user.

Null (default)

UTF-8 encoding string

dir.corp.alt.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

`dir.corp.alt.attribute.x.type`

Define how x is interpreted by the phone. Entries can have multiple parameters of the same type.

first_name

last_name (default)

phone_number

SIP_address

Other – for display purposes only.

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.

`dir.local.serverFeatureControl.method`

Specifies a method for synchronizing the directory and server.

None (default)

GENBANDSOPI - Enables the GENBANDSOPI protocol on the phone to get the personnel address book service from the Ribbon Communications server.

Personal Address Book Parameters

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

Note that when you enable server control, five telephone number fields per contact are available.

`feature.corporateDirectory.alt.enabled`

0 (default) - Disables the global address book service.

1 - Enables the global address book service.

`dir.local.serverFeatureControl.method`

Specifies a method for synchronizing the directory and server.

None (default)

GENBANDSOPI - Enables the GENBANDSOPI protocol on the phone to get the personnel address book service from the Ribbon Communications server.

`dir.local.serverFeatureControl.reg`

Specifies the phone line to enable the personal address book feature on.

1 (default)

1 -34

`dir.genband.local.contacts.maxSize`

Specify the maximum number of contacts available in the Ribbon Communications personnel address book contact directory.

100 (default)

1 - 100

Enhanced 911 (E.911) Location Parameters for Ribbon Communications

Use the parameter below to configure this feature.

Emergency Instant Message Parameters

Use the following parameters to configure emergency messages on phones registered with Ribbon Communications.

`feature.instantMessaging.displayTimeout`

Specify the time in minutes instant messages display.

Messages display until one of the following occurs:

- Timeout
- Another instant message is received
- A pop-up message displays
- The phone receives an incoming call
- The user presses any key or message on the phone

1 minute (default)

1 – 60 minutes

`feature.instantMessaging.ring`

instantMessage (default) – The phone plays a configured tone when an emergency instant message is received.

Silent – No tone is played.

`feature.instantMessaging.enabled`

0 (default) – The phone does not display emergency instant messages.

1 - Received emergency instant messages display on the phone.

PVOS Parameters

Use the following parameters to configure user-controlled software update settings.

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.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 Profile Parameters

Use the following parameter to configure user profiles for user phone access and settings.

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.

Video Parameters

Use the following parameters to configure video settings for phones that support video conferencing.

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.

`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.

I-Frames

When video streams initialize, devices transmit video packets called I-frames (reference frames) that contain information to display a complete image.

The phones send smaller and less complete frames, known as P-frames, to consume less bandwidth. Due to packet loss, jitter, or corruption, phones occasionally must make multiple requests for a complete I-frame to reset the full frame. Devices can then revert to P-frame updates.

You can set parameters to control an I-frame request. The following table indicates parameter dependencies and messaging behavior when setting an I-frame request method.

I-Frame Parameter Dependencies

video.forceRtcpVideoCodecControl	video.dynamicControlMethod	voIpProt.SDP.offer.rtcpVideoCodecControl	Behavior when requesting video I-frame updates
0	0 (n/a)	0	The phone sends only SIP INFO messages. No RTCP-FB is offered in SDP.
0	1 (n/a)	0	The phone sends only SIP INFO messages. No RTCP-FB is offered in SDP.
0	0 (n/a)	1	RTCP-FB is offered in SDP. If SDP responses don’t contain the required RTCP-FB attribute, then the phone uses only SIP INFO requests.
0	1 (N/A)	1	RTCP-FB is offered in SDP. If SDP responses don’t contain the required RTCP-FB attribute, then the phone uses only SIP INFO requests.
1	0	0	The SDP attribute `a=rtcp-fb` isn’t included in SDP offers. The phone attempts both RTCP-FB and SIP INFO messages.
1	1	0	The SDP attribute `a=rtcp-fb` isn’t included in SDP offers. The phone attempts both RTCP-FB and SIP INFO messages. If the phone receives no RTCP-FB messages, the phone sends only SIP INFO messages. If the phone receives no response for SIP INFO messages, then the phone attempts both RTCP-FB and SIP INFO messages again.
1	0	1	RTCP-FB is offered in SDP. Even if the SDP response doesn’t include an accepted `a=rtcp-fb` attribute, the phone sends both RTCP-FB and SIP INFO messages.
1	1	1	RTCP-FB is offered in SDP. Even if the SDP response doesn’t include an accepted `a=rtcp-fb` attribute, the phone initially sends both RTCP-FB and SIP INFO messages. If the phone doesn’t receive a RTCP-FB response, the phone sends only SIP INFO messages.

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.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 CCX 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 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 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.

512 (default) - The overlay does not time out.

128 2048

`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.

768 (default)

128 – 2048

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