Configuring Audio Settings

Topics:

- Automatic Gain Control
- Enable AEC for Headsets
- Noise Suppression
- Comfort Noise
- Audio Codecs

Configure modifications to the default audio configurations to optimize the audio quality of your phones.

Automatic Gain Control

Automatic gain control (AGC) boosts the volume of near-end conference participants. AGC is enabled by default to ensure far-end audio clarity.

Note: You can’t disable this feature. Changing the default settings may cause accessibility concerns for people who use audio augmentation assistive technology.

If you are running an application that also provides AGC through the software, Poly recommends that you disable the application AGC.

Enable AEC for Headsets

The default configuration enables acoustic echo cancellation (AEC) for both the handset and speakerphone. Enable AEC for connected Poly Bluetooth headsets to reduce echo during calls.

AEC includes the following features:

- Talk state detector: Determines whether the near-end user, far-end user, or both are speaking.
- Linear adaptive filter: Adaptively estimates the loudspeaker-to-microphone echo signal and subtracts that estimate from the microphone signal.
- Nonlinear processing: Suppresses any echo remaining after the linear adaptive filter.

Procedure

» Enable AEC for headsets.

```
voice.aec.bt.hd.enable="1"
```

Noise Suppression

Poly phones offer multiple options to suppress background noise during calls. Some options are integrated into the phone itself, but you can configure others.

Integrated noise suppression reduces background noise caused by items such as fans, projectors, and air conditioners.

Poly NoiseBlock

The default configuration enables Poly NoiseBlock on Poly phones.

For more information, see the parameter reference topic(s) in the *Poly CCX Parameter Reference Guide*.

Disable Poly NoiseBlock

Disable Poly Noiseblock and Poly NoiseBlockAI on your phones.

Poly NoiseBlock automatically mutes the microphone when a user stops speaking. It reduces interruptions caused by common office sounds (keyboard tapping, shuffling papers, etc.) and background chatter.

Procedure

» Disable Poly NoiseBlock.

```
voice.ns.hf.block="0"
```

Enable Poly NoiseBlockAI

Enable Poly NoiseBlockAI on your phones.

Poly NoiseBlockAI suppresses background noise while a call participant actively speaks. It also reduces interruptions caused by common office sounds (keyboard tapping, shuffling papers, etc.) and background chatter. Call recipients hear only the intended speaker's voice.

Note: You can't enable both Poly NoiseBlock and Poly NoiseBlockAI at the same time. The same parameter configures both modes.

Procedure

» Enable Poly NoiseBlockAI.

```
voice.ns.hf.block="2"
```

Acoustic Fence

Acoustic Fence technology suppresses background noise sent to the far end. This feature is particularly useful in call center environments where background noise can impact far-end audio quality.

Acoustic Fence works with the following devices:

- Phone handsets
- Wired headsets connected to the headset port

- USB headsets connected to the phone

Note: Acoustic Fence doesn't support Bluetooth headsets.

Enable Polycom Acoustic Fence for Handset Calls

Enable Polycom Acoustic Fence for handset calls to remove unwanted background noise from calls.

Important: Configuring the following parameter(s) causes the phone to reboot. Dependencies and overrides may affect other parameters.

Procedure

1. Enable noise suppression for handset calls.

   ```
   voice.ns.hs.enable="1"
   ```

2. Enable Polycom Acoustic Fence for handset calls.

   ```
   voice.ns.hs.enhanced="1"
   ```

3. Optional: Configure the Polycom Acoustic Fence threshold for handset calls.

   A lower number removes less background noise, while a higher number removes more background noise. The default value is 8.

   ```
   voice.ns.hs.nonStationaryThresh="<1 to 10>"
   ```

Enable Polycom Acoustic Fence for Headset Calls

Enable Polycom Acoustic Fence for headset calls to remove unwanted background noise from calls.

Procedure

1. Enable noise suppression for headset calls.

   ```
   voice.ns.hd.enable="1"
   ```

2. Enable Polycom Acoustic Fence for headset calls.

   ```
   voice.ns.hd.enhanced="1"
   ```

3. Configure the Polycom Acoustic Fence threshold for headset calls.

   A lower number removes less background noise while a higher number removes more background noise. The default value is 8.

   ```
   voice.ns.hd.nonStationaryThresh="<1 to 10>"
   ```

Add Acoustic Fence Options to the Local Interface

Add the Polycom Acoustic Fence menu items to the phone's **Basic** menu.

Important: Configuring the following parameter(s) causes the phone to reboot. Dependencies and overrides may affect other parameters.

Procedure

» Enable the **Acoustic Fence** menu item on the phone's local interface.

```
feature.acousticFenceUI.enabled="1"
```

Dynamically Deactivate Acoustic Fence in Full-Screen Mode

Enable the phone to dynamically deactivate Acoustic Fence when users change the view to full screen mode in a video call.

Enable this setting to optimize CCX 600 phone performance while using a Polycom EagleEye Mini USB camera with Acoustic Fence.

Procedure

» Enable the phone to dynamically deactivate Acoustic Fence when in full-screen mode.

```
video.disableAFOnFullScreen="1"
```

Configure VAD

Set the threshold for determining what is considered background noise using Voice activity detection (VAD).

Voice activity detection (VAD) conserves network bandwidth. VAD detects periods of silence in the transmit data path so the phone doesn't transmit unnecessary data packets for outgoing audio.

For compression algorithms without an inherent VAD function, such as G.711,the phone uses the codec-independent processing specified in RFC 3389.

G.711 Appendix II, in RFC 3389, defines the payload format for G.711 use in packet-based multimedia communication systems.

For more information about VAD, see Voice Activity Detection parameters in the *Parameter Reference Guide*.

Procedure

1. Enable VAD.

```
voice.vadEnable=”1"
```

2. Set the VAD threshold in decibels.

   The default value is 25. Sounds louder than the VAD threshold are considered voice. Sounds below the threshold are considered background and muted from the call.

```
voice.vadThresh=”<0 to 30>”
```

Comfort Noise

Comfort noise ensures a consistent background noise level to provide a natural call experience for speakerphone and handset calls.

Comfort noise is enabled by default on Poly phones, and the payload type is negotiated in the Session Description Protocol (SDP) with a default of 13 for 8 kHz codecs and 122 for 16 kHz codecs and higher.

Note: Comfort noise isn't related to the comfort noise packets the phone generates when you enable VAD.

Configure Comfort Noise for Speakerphone Calls

Add comfort noise to a hands-free call to ensure that the line isn't completely silent when callers aren't talking.

Important: Configuring the following parameter(s) causes the phone to reboot. Dependencies and overrides may affect other parameters.

Comfort noise provides a minimal level of audio on the line to ensure callers that the call is still connected. You can add and adjust the level of comfort noise for speakerphone and headset calls.

Procedure

1. Enable comfort noise for speakerphone calls.

   ```
   voice.cn.hf.enable="1"
   ```

2. Optional: Adjust the comfort noise level.

   The phone's default value of 30 is quite loud. Enter a higher number to reduce the comfort noise. A lower number increases the comfort noise.

   ```
   voice.cn.hf.attn="<0 to 90>"
   ```

Configure Comfort Noise for Handset Calls

Add comfort noise to a handset call to ensure that the line isn't completely silent when callers aren't talking.

Important: Configuring the following parameter(s) causes the phone to reboot. Dependencies and overrides may affect other parameters.

Procedure

1. Enable comfort noise for handset calls.

   ```
   voice.cn.hs.enable="1"
   ```

2. Optional: Adjust the comfort noise level.

The default value is 35.

```
voice.cn.hs.attn="<0 to 90>"
```

Audio Codecs

Configure the audio codecs for your phones.

This section provides basic information for configuring audio codecs. For more information on configuring audio codecs, see the parameter reference topic(s) in the *Poly CCX Parameter Reference Guide*.

Supported Audio Codec Specifications

Specifications for audio codecs supported on Poly phones.

Note: The network bandwidth necessary to send encoded voice is typically 5% to 10% higher than the encoded bit rate due to packetization overhead. For example, a G.722.1C call at 48 Kbps consumes about 100 Kbps of network bandwidth for both the receive and transmit signals (two-way audio).

Audio Codec Specifications

Device Support	Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
All systems	G.711 µ-law	RFC 1890	64 Kbps	80 Kbps	8 ksps	20 ms	3.5 kHz
All systems	G.711 a-law	RFC 1890	64 Kbps	80 Kbps	8 ksps	20 ms	3.5 kHz
All systems	G.711	RFC 1890	64 Kbps	80 Kbps	16 ksps	20 ms	7 kHz

Device Support	Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
All systems	G.722 Per RFC 3551. Even though the actual sampling rate for G.722 audio is 16,000 Hz (16 ksps), the RTP clock rate advertised for the G.722 payload format is 8,000 Hz because that value was erroneously assigned in RFC 1890 and must remain unchanged for backward compatibility.	RFC 3551	64 Kbps	80 Kbps	16 ksps	20 ms	7 kHz
All systems	G.722.1	RFC 3047	24 Kbps 32 Kbps	40 Kbps 48 Kbps	16 ksps	20 ms	7 kHz
All systems	G.722.1C	G7221C	224 Kbps 32 Kbps 48 Kbps	40 Kbps 48 Kbps 64 Kbps	32 ksps	20 ms	14 kHz
All systems	G.729AB	RFC 1890	8 Kbps	24 Kbps	8 ksps	20 ms	3.5 kHz
All systems	Opus	RFC 6716	8 to 24 Kbps	24 to 40 Kbps	8 ksps 16 ksps	20 ms	3.5 kHz 7 kHz
All systems	Lin16	RFC 1890	128 Kbps 256 Kbps 512 Kbps 705.6 Kbps 768 Kbps	132 Kbps 260 Kbps 516 Kbps 709.6 Kbps 772 Kbps	8 ksps 16 ksps 32 ksps 44.1 ksps 48 ksps	10 ms	3.5 kHz 7 kHz 14 kHz 20 kHz 22 kHz
All systems	Siren 7	SIREN7	16 Kbps 24 Kbps 32 Kbps	32 Kbps 40 Kbps 48 Kbps	16 ksps	20 ms	7 kHz

Device Support	Algorithm	Reference	Raw Bit Rate	Maximum IP Bit Rate	Sample Rate	Default Payload Size	Effective Audio Bandwidth
All systems	Siren14	SIREN14	24 Kbps 32 Kbps 48 Kbps	40 Kbps 48 Kbps 64 Kbps	32 ksps	20 ms	14 kHz
All systems	iLBC	RFC 3951	13.33 Kbps 15.2 Kbps	31.2 Kbps 24 Kbps	8 ksps	20 ms 30 ms	3.5 kHz
All systems	SILK	SILK	6 to 20 Kbps 7 to 25 Kbps 8 to 30 Kbps 12 to 40 Kbps	36 Kbps 41 Kbps 46 Kbps 56 Kbps	8 ksps 12 ksps 16 ksps 24 ksps	20 ms	3.5 kHz 5.2 kHz 7 kHz 11 kHz

Set Audio Codec Priority

Set the codec priority to improve consistency and reduce workload on the phones.

Note the following about audio codec priority:

- Permitted values to set audio codec priority are 1 to 35.
- 1 is the highest priority.
- A value of 0 or Null disables the codec.
- A change to the default value doesn't cause a phone to restart or reboot.

Note: The Opus codec isn't compatible with G.729 and iLBC. If you set Opus to the highest priority, G.729 and iLBC don't publish.

If you set G.729 and iLBC to the highest priority, Opus doesn't publish.

The phone doesn't answer calls using unsupported codecs. If the phone receives a call using an unsupported codec, the phone answers the call with the first supported codec priority.

The following values represent the configuration defaults. The default configuration sets the priority values from 1 to 8. All codecs **not** listed in the following table have a default priority value of 0 (disabled).

Audio Codec Priority Default Values

Parameter	Default Priority
`voice.codecPref.Siren22.64kbps`	1

Parameter	Default Priority
voice.codecPref.G7221_C.48kbps	2
voice.codecPref.Siren14.48kbps	3
voice.codecPref.G722	4
voice.codecPref.G7221.32kbps	5
voice.codecPref.G711_Mu	6
voice.codecPref.G711_A	7
voice.codecPref.G729_AB	8

Procedure

» Set audio codec priority.

```
voice.codecPref.AMRNB="<priority value>"
voice.codecPref.AMRWB="<priority value>"
voice.codecPref.G711_A="<priority value>"
voice.codecPref.G711_Mu="<priority value>"
voice.codecPref.G719.32kbps="<priority value>"
voice.codecPref.G719.48kbps="<priority value>"
voice.codecPref.G719.64kbps="<priority value>"
voice.codecPref.G722="<priority value>"
voice.codecPref.G7221.16kbps="<priority value>"
voice.codecPref.G7221.24kbps="<priority value>"
voice.codecPref.G7221.32kbps="<priority value>"
voice.codecPref.G7221_C.24kbps="<priority value>"
voice.codecPref.G7221_C.32kbps="<priority value>"
voice.codecPref.G7221_C.48kbps="<priority value>"
voice.codecPref.G729_AB="<priority value>"
voice.codecPref.iLBC.13_33kbps="<priority value>"
voice.codecPref.iLBC.15_2kbps="<priority value>"
voice.codecPref.Lin16.8ksps="<priority value>"
voice.codecPref.Lin16.16ksps="<priority value>"
voice.codecPref.Lin16.32ksps="<priority value>"
voice.codecPref.Lin16.44_1ksps="<priority value>"
voice.codecPref.Lin16.48ksps="<priority value>"
voice.codecPref.Opus="<priority value>"
voice.codecPref.SILK.8ksps="<priority value>"
voice.codecPref.SILK.12ksps="<priority value>"
voice.codecPref.SILK.16ksps="<priority value>"
voice.codecPref.SILK.24ksps="<priority value>"
voice.codecPref.Siren7.16kbps="<priority value>"
voice.codecPref.Siren7.24kbps="<priority value>"
voice.codecPref.Siren7.32kbps="<priority value>"
voice.codecPref.Siren14.24kbps="<priority value>"
voice.codecPref.Siren14.32kbps="<priority value>"
voice.codecPref.Siren14.48kbps="<priority value>"
voice.codecPref.Siren22.32kbps="<priority value>"
voice.codecPref.Siren22.48kbps="<priority value>"
voice.codecPref.Siren22.64kbps="<priority value>"
```

Configure the SILK Audio Codec

Configure the SILK audio codec settings.

Procedure

1. Set the maximum average encoder output bit rate in kpbs for the supported SILK sample rate. Replace *x* with the sample rate.

   ```
   voice.audioProfile.SILK.xksps.encMaxAvgBitrateKbps="<value>"
   ```

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

   The default is 2. The value range is 0 to 2.

   ```
   voice.audioProfile.SILK.encComplexity="<value>"
   ```

3. Optional: Enable inband forward error correction (FEC) in the SILK encoder.

   Note: When you enable this parameter, perceptually important speech information is sent twice: once in the normal bit stream and again at a lower bit rate in later packets. This results in an increased bit rate.

   ```
   voice.audioProfile.SILK.encInbandFECEnable="1"
   ```

4. Set the SILK encoder expected network packet loss percentage.

   The default is 0. The value range is 0 to 100.

   Note: Configuring this value enables less interframe dependency encoded into the bit stream. This results in increasingly larger bit rates but with an average bit rate less than that configured with `voice.audioProfile.SILK.*`.

   ```
   voice.audioProfile.SILK.encExpectedPktLossPercent="<value>"
   ```

5. Optional: Enable discontinuous transmission (DTX) in the SILK encoder.

   Note: DTX reduces the encoder bit rate to 0 bps during silence.

   ```
   voice.audioProfile.SILK.encDTXEnable="1"
   ```

Configure the Opus Audio Codec

Configure the Opus audio codec settings.

Procedure

1. Assign the Opus encoder’s application type.
   - VoIP (Default) - Process signal for improved speech intelligibility
   - Audio - Favors faithfulness to original input audio