Audio Configuration

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Configure Different Audio Volume

You can configure the volume settings in the phone web interface.

You can also configure the parameters in the phone configuration file with XML (cfg.xml) code. To configure each parameter, see the syntax of the string in the **Parameters for Audio Volume** table in Parameters for Audio Volume, on page 1.

Before you begin

Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **User**.

Step 2 In the **Audio Volume** section, configure the volume level for audio parameters as described in the **Parameters for Audio Volume** table in Parameters for Audio Volume, on page 1.

Step 3 Click **Submit All Changes**.

Parameters for Audio Volume

The following two tables describes the acoustic and audio settings.

The following table defines the function and usage of Audio Volume parameters in the Audio Volume section under the User tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 1: Parameters for Audio Volume

Parameter	Description
Ringer Volume	Sets the default volume for the ringer. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Ringer_Volume ua="rw">8</Ringer_Volume>` • In the phone web page, enter a valid value as the ringer volume. Allowed values: an integer ranging between 0 and 15 Default: 9
Speaker Volume	Sets the default volume for the speakerphone. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Speaker_Volume ua="rw">11</Speaker_Volume>` • In the phone web page, enter a valid value as the speaker volume. Allowed values: an integer ranging between 0 and 15 Default: 11
Handset Volume	Sets the default volume for the handset. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Handset_Volume ua="rw">9</Handset_Volume>` • In the phone web page, enter a valid value as the handset volume. Allowed values: an integer ranging between 0 and 15 Default: 10

Configure the Voice Codecs

A codec resource is considered allocated if it has been included in the SDP codec list of an active call, even though it eventually might not be chosen for the connection. Negotiation of the optimal voice codec sometimes depends on the ability of the Cisco IP Phone to match a codec name with the far-end device or gateway codec

name. The phone allows the network administrator to individually name the various codecs that are supported such that the correct codec successfully negotiates with the far-end equipment.

The Cisco IP Phone supports voice codec priority. You can select up to three preferred codecs. The administrator can select the low-bit-rate codec that is used for each line. G.711a and G.711u are always enabled.

You can also configure the parameters in the phone configuration file with XML(cfg.xml) code. To configure each parameter, see the syntax of the string in Audio Codec Parameters, on page 3.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where n is an extension number.

Step 2 In the **Audio Configuration** section, configure the parameters as defined in the Audio Codec Parameters, on page 3 table.

Step 3 Click **Submit All Changes**.

Audio Codec Parameters

The following table defines the function and usage of the voice codec parameters in the **Audio Configuration** section under the **Voice** > **Ext (n)** tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file (cfg.xml) with XML code to configure a parameter.

Table 2: Audio Codec Parameters

Parameter	Description
Preferred Codec	Preferred codec for all calls. The actual codec used in a call still depends on the outcome of the codec negotiation protocol. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Preferred_Codec_1_ ua="rw">G711u</Preferred_Codec_1_>` • In the phone web interface, select your preferred codec from the list. Allowed values: G711u|G711a|G729a|G722|G722.2|iLBC|OPUS Default: G711u

Parameter	Description
Use Pref Codec Only	Select **No** to use any code. Select **Yes** to use only the preferred codes. When you select Yes, calls fail if the far end does not support the preferred codecs. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Use_Pref_Codec_Only_1_ ua="rw">No</Use_Pref_Codec_Only_1_>` • In the phone web interface, set this field to Yes or No as needed. Allowed values: Yes|No Default: No
Second Preferred Codec	Codec to use if the codec specified in **Preferred Codec** fails. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Second_Preferred_Codec_1_ ua="rw">Unspecified</Second_Preferred_Codec_1_>` • In the phone web interface, select your preferred codec from the list. Allowed values: Unspecified|G711u|G711a|G729a|G722|G722.2|iLBC|OPUS Default: Unspecified
Third Preferred Codec	Codec to use if the codecs specified in **Preferred Codec** and **Second Preferred Codec** fail. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Third_Preferred_Codec_1_ ua="rw">Unspecified</Third_Preferred_Codec_1_>` • In the phone web interface, select your preferred codec from the list. Allowed values: Unspecified|G711u|G711a|G729a|G722|G722.2|iLBC|OPUS Default: Unspecified

Parameter	Description

G711u Enable
G711a Enable
G729a Enable
G722 Enable
G722.2 Enable
iLBC Enable
	Enables the use of a specific codec.
	Perform one of the following:
	• In the phone configuration file with XML(cfg.xml), enter a string in this format:
	`<G711u_Enable_1_ ua="rw">Yes</G711u_Enable_1_>`
	`<G711a_Enable_1_ ua="rw">Yes</G711a_Enable_1_>`
	`<G729a_Enable_1_ ua="rw">Yes</G729a_Enable_1_>`
	`<G722_Enable_1_ ua="rw">Yes</G722_Enable_1_>`
	`<G722_Enable_1_ ua="rw">Yes</G722_Enable_1_>`
	`<G722.2_Enable_1_ ua="rw">No</G722.2_Enable_1_>`
	`<iLBC_Enable_1_ ua="rw">No</iLBC_Enable_1_>`
	`<OPUS_Enable_1_ ua="rw">Yes</OPUS_Enable_1_>`
	• In the phone web interface, set the corresponding field to **Yes** to enable the use of a specific codec, or **No** to disable it.
	Note The transmit rate for the G.729a codec is at 8 kbps.

Silence Supp Enable	Enables or disables silence suppression. When set **Yes**, silent audio frames are not transmitted. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Silence_Supp_Enable_1_ ua="rw">No</Silence_Supp_Enable_1_>` • In the phone web interface, set this field to **Yes** to enable silence suppression, or **No** to disable it. Allowed values: Yes|No Default: No

Parameter	Description
DTMF Tx Method	The method for transmitting DTMF signals to the far end. The options are: • AVT—Audio video transport. Sends DTMF as AVT events. • InBand—Sends DTMF by using the audio path. • Auto—Uses InBand or AVT based on the outcome of codec negotiation. • INFO—Uses the SIP INFO method. • InBand+INFO—Uses both the audio path and the SIP INFO method. • AVT+INFO—Uses both the AVT and the SIP INFO method. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<DTMF_Tx_Method_1_ ua="rw">Auto</DTMF_Tx_Method_1_>` • In the phone web interface, select your preferred transmitting method from the list. Default: Auto
Codec Negotiation	When set to **Default**, the phone responds to an Invite with a 200 OK response advertising the preferred codec only. When set to **List All**, the phone responds listing all the codecs that the phone supports. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Codec_Negotiation_1_ ua="na">Default</Codec_Negotiation_1_>` • In the phone web interface, select the desired option from the list. Allowed values: Default|List All Default: Default
Encryption Method	Encryption method to be used during secured call. Options are AES 128 and AES 256 GCM Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Encryption_Method_1_ ua="na">AES 128</Encryption_Method_1_>` • In the phone web interface, select your preferred encryption method from the list. Allowed values: AES 128 |AES 256 GCM Default: AES 128.

Voice Quality Reporting

You can capture voice quality metrics for Voice over Internet Protocol (VoIP) sessions with a Session Initiation Protocol (SIP) event package. Voice call quality information derived from RTP and call information from SIP is conveyed from a User Agent (UA) in a session (reporter) to a third party (collector).

The Cisco IP phone uses User Datagram Protocol (UDP) to send a SIP PUBLISH message to a collector server.

Supported Scenarios for Voice Quality Reporting

Currently, only the basic call scenario supports voice quality reporting. A basic call can be a peer to peer incoming or outgoing call. The phone supports periodic SIP publish message.

Mean Opinion Scores and Codecs

The voice quality metrics use Mean Opinion Score (MOS) to rate the quality. A MOS rating of 1 is the lowest quality; a MOS rating of 5 is the highest quality. The following table gives a description of some of the codecs and MOS scores. The phone supports all codecs. For all codecs, the phone sends the SIP Publish message.

Codec	Complexity and Description	MOS	Minimum Call Duration for Valid MOS Value
G.711 (A-law and u-law)	Very low complexity. Supports uncompressed 64 kbps digitized voice transmission at one to ten 5 ms voice frames-per-packet. This codec provides the highest voice quality and uses the most bandwidth of any of the available codecs.	A minimum value of 4.1 indicates good voice quality.	10 seconds
G.729A	Low to medium complexity.	A minimum value of 3.5 indicates good voice quality.	30 seconds
G.729AB	Contains the same reduced complexity modifications present in the G.729A.	A minimum value of 3.5 indicates good voice quality.	30 seconds

Configure Voice Quality Reporting

You can generate a voice quality report for each extension on the phone. The parameters for the Voice Quality Metrics (VQM) SIP Publish Message help you to:

- Generate voice quality reports.
- Name your reports.
- Determine when your phone sends SIP Publish messages.

You can also configure the parameters in the phone configuration file with XML(cfg.xml) code. See VQM SIP Publish Message Parameters, on page 8

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where (n) is the extension number.

Step 2 In **SIP Settings**, enter a value for the **Voice Quality Report Address** parameter. You can enter either a domain name or an IP address.

You can also add a port number along with the domain name or an IP address for this parameter. If you do not enter a port number, the value of the **SIP UDP Port** (5060) is used by default. If the collector server URL parameter is blank, a SIP PUBLISH message is not sent out.

Step 3 Enter your report name for the **Voice Quality Report Group** parameter.

Your report name can't begin with a hyphen (-), semicolon (;), or a space.

Step 4 Enter an interval, in seconds, for the **Voice Quality Report Interval** parameter. Example: `20` for 20-second interval reporting.

Step 5 Click **Submit All Changes**.

VQM SIP Publish Message Parameters

The following table defines the Voice Quality Metrics (VQM) SIP Publish Message parameters in the **Sip Settings** section under **Voice** > **Ext(n)** tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file (cfg.xml) with XML code to configure a parameter.

Table 3: VQM SIP Publish Message Parameters

Parameter Name	Description
Voice Quality Report Address	Allows you to enter one of the following options: • Domain Name • IP address • The SIP UDP port number along with the domain name In the phone XML configuration file (cfg.xml), enter a string in this format: `<Voice_Quality_Report_Address_1_ ua="na">fake_vq_collector</Voice_Quality_Report_Address_1_>` Default parameter = empty (no report) Default SIP UDP Port = 5060

Parameter Name	Description
Voice Quality Report Group	Allows you to enter a voice quality report name. Your report name cannot begin with a: • hyphen (-) • semicolon (;) • space In the phone XML configuration file (cfg.xml), enter a string in this format: `<Voice_Quality_Report_Group_1_ ua="na">test-group-1</Voice_Quality_Report_Group_1_>` Default parameter = empty (The report will use the canonical name in the form of **`identifier@ipAddress`**.)
Voice Quality Report Interval	Allows you to determine when the phones send SIP Publish messages. If you have properly configured the **Voice Quality Report Address**, the SIP Publish messages can be sent: • When the call has ended or is placed on hold. • Periodically, when you enter an interval in seconds for this parameter. Example: **`20`** for 20-second intervals. In the phone XML configuration file (cfg.xml), enter a string in this format: `<VQ_Report_Interval_1_ ua="na">20</VQ_Report_Interval_1_>` Default parameter = 0 (no periodic SIP Publish Message)