Main page: Cisco Unified MeetingPlace, Release 7.0

Up one level: Reference Information

Use this page to modify audio and video settings. To find this page, click **System Configuration > Call Configuration > Media Parameters**.

Table: Field Reference: Media Parameters Page

Field	Description
Media Parameters	
Audio RTP starting port	The Media Server assigns RTP[1]/UDP[2] ports starting from the specified value up to that value plus 1024. Modify this field only if required to conform with local firewall rules. This field applies only to audio ports. Video ports always start at 20000 and are not configurable. Default 16384
TTL	Time-to-live value in the IP header of transmitted voice packets. Determines how many hops an IP packet can travel through the network before it is discarded. Recommendation: Set the value at least high enough to match the number of router hops between Cisco Unified MeetingPlace and the furthest user endpoint. Using a relatively low number can help reduce the quantity of stray packets on the network. Default: 64
QoS[3]	
Audio media IPv4 Video media IPv4 Signaling IPv4	Differentiated Services (DiffServ) code point (DSCP) settings that determine the QoS for the audio and video media signaling, as defined in RFC 2475. Recommendation: Keep the default value. The other values are available for the rare instances when the network requires a different DSCP setting. For more information, see the "Network Infrastructure" chapter of the *Cisco Unified Communications Solution Reference Network Design (SRND)* that applies to your version of Cisco Unified Communications Manager at http://www.cisco.com/go/designzone. Defaults:

	• Audio media: EF DSCP (101110) • Video media: AF41 DSCP (100010) • Signaling: CS3 (precedence 3) DSCP (011000)
Echo Canceller[4]	
Window size (milliseconds)	Range of echo return delay that the LEC will attempt to cancel. Default: 128
Enable non-linear processing (NLP)	NLP removes the small amount of residual uncanceled echo that inevitably passes through the echo canceller and may be useful for removing residual echo from acoustic or low-bandwidth voice codec (for example, ITU-T G.729) endpoints. Set this field to No: • If you do not want to suppress the residual echo. • If you notice subtle voice quality issues, such as variations in background noise levels while NLP is enabled. Default: Yes
Enable comfort noise in NLP	To help make the overall background noise level continuous, the NLP generates comfort noise. Set this field to No if you prefer silence instead of comfort noise whenever NLP is actively removing residual echo. Note, however, that disabling comfort noise may result in undesirable variations of background noise levels between silence and noise. Default: Yes
Enable LEC w. G.729?	Whether to enable LEC when G.729 is in use. Restrictions: • You must set this field to No if you select the higher capacity option in the Global audio mode field on the Meeting Configuration Page. Otherwise, the G.729 codec will be disabled. • Changes to this field take effect only after restarting the system.[5] This field was introduced in Release 7.0.2. Default: Yes
Minimum echo return loss (ERL) (dB)	A lower ERL setting may help the LEC cancel loud echoes, but it increases the risk of distortion caused by clipping or squelching of the signal. Default: 6

Bulk delay (milliseconds)	This value is added to the Window size (milliseconds), so that the cancelled echo return delays will range from Bulk delay (milliseconds) to Bulk delay (milliseconds) + Window size (milliseconds). This allows the LEC to work on echoes that are outside the normal range in exchange for not canceling short-return-delay echoes. Default: 0
Gain Control[6]	
Enable automatic gain control (AGC)	AGC causes Cisco Unified MeetingPlace to dynamically adjust the input gain so the average energy matches a specific level. This is useful when various phones, or people in a conference room, produce different volume levels. Nevertheless, AGC can be problematic in cases where noise may be mistaken for voice. When AGC is disabled, the specified Fixed gain (dB) is applied to all inputs. Restriction: This field is not supported in Release 7.0.1. Default: No
AGC target level (dBm)	The target energy level for the AGC algorithm is applied to all inputs. Make this number less negative to increase the average volume level. The default value of -18 is a typical level for telephony circuits. Restrictions: • This field applies only when the Enable automatic gain control (AGC) field is set to Yes. • This field is not supported in Release 7.0.1. Default: -18
Fixed gain (dB)	The fixed input gain is applied to all inputs. Use positive numbers to increase the volume, and use negative numbers to decrease the volume. The default value of 0 leaves the input level alone. Restriction: This field applies only when the Enable automatic gain control (AGC) field is set to No. Default: 0
Digits[7]	
Enable RFC 2833 detection	There are three DTMF methods: 1. RFC-2833, which is negotiated and can be disabled 2. KPML, which is negotiated and cannot be disabled

	3. In-band DTMF tones, which is not negotiated but can be disabled (see "Enable in-band DTMF detection" setting) RFC 2833 is a standard mechanism for transmitting keypad digits in-band in VoIP media packets. It is commonly used as an adjunct to SIP signaling. Most calls will negotiate either RFC 2833 (in band) or KPML[8] (out of band) depending on the capabilities of the user endpoint. If both RFC-2833 and KPML are negotiated (implying that RFC-2833 was enabled), Cisco Unified MeetingPlace will listen for RFC-2833 and not KPML. You can force the use of KPML by disabling RFC-2833 if you are trying to validate KPML. Otherwise, disabling RFC-2833 is typically not necessary as most calls will not notice a difference. If you do notice a difference it may be due to Cisco Unified Communications Manager inserting a MTP to translate RFC-2833 to KPML. This happens if a trunk or endpoint does not support out-of-band signaling. Depending on the setup, MTP insertion may result in loss of video or, if you run out of MTP resources, call failure. Default: Yes
RFC2833 payload type	*Not supported.* Appears only in Release 7.0.1.
Enable in-band DTMF detection	Whether to turn on the signal processing which looks for in-band acoustic DTMF[9] tones in the input audio media stream. Note that DTMF works well only with the G.711 codec. Recommendation: Enter Yes to support terminals that lack another signaling mechanism, including RFC 2833, KPML, or H.245. Enter No if you find that Cisco Unified MeetingPlace responds to voices as if they were keypad inputs (talk off). Default: Yes
Jitter Buffer	
Maximum size (milliseconds) Minimum size (milliseconds)	Maximum and minimum lengths of time, in milliseconds, that the jitter buffer holds voice packets. A large jitter buffer helps the system accurately reassemble the media stream, but it adds to perceived latency. Jitter refers to the variation in the delay of received packets. When voice data is sent across the network, the packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, the delay between each received packet can vary instead of remaining constant. Some packets may even arrive out of order or not arrive at all. A higher Maximum size (milliseconds) helps the system adapt to poor conditions. A lower value may be better for interactive conversations, where an occasional dropped packet may be preferable to long latency.

	The Minimum size (milliseconds) is the starting jitter buffer size. The closer this value is to the typical jitter on the network, the quicker the system adapts, but this adds directly to latency. Maximum size (milliseconds) default: 250 Minimum size (milliseconds) default: 30
Miscellaneous	
Maximum conference speakers	Maximum number of input lines that will be simultaneously mixed together in a meeting. A small value (2) reduces the background noise and echo, which is best for lecture-style meetings. A large value (4) is best for more interactive meetings. Default: 4

Footnotes:

1. RTP = Real-Time Transport Protocol

2. UDP = User Datagram Protocol

3. QoS = Quality of Service

4. The echo canceller parameters control the Line Echo Canceller (LEC), which reduces audible echo in meetings.

5. A system restart terminates all existing call connections. Proceed only during a scheduled maintenance period or during a period of extremely low usage. To restart the system, enter sudo mpx_sys restart in the CLI. For information about logging into the CLI, see Using the Command-Line Interface (CLI) in Cisco Unified MeetingPlace.

NOTE: When you restart the Web Server, all manual changes made to the registry are lost.

6. The gain control parameters apply a fixed or adaptive gain to all audio inputs.

7. The digits parameters control how keypad inputs are received.

8. KPML = Key Press Markup Language

9. DTMF = Dual Tone Multi-Frequency

Related Topics

- Configuring Parameters that Affect Sound and Video Quality