Introduction to Codecs

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Overview

A codec is a device or software capable of encoding or decoding a digital data stream or signal. Audio codecs can encode or decode a digital data stream of audio. Video codecs encode or decode digital video streams.

This chapter describes the basics of encoding digital voice samples using codecs and how to configure them.

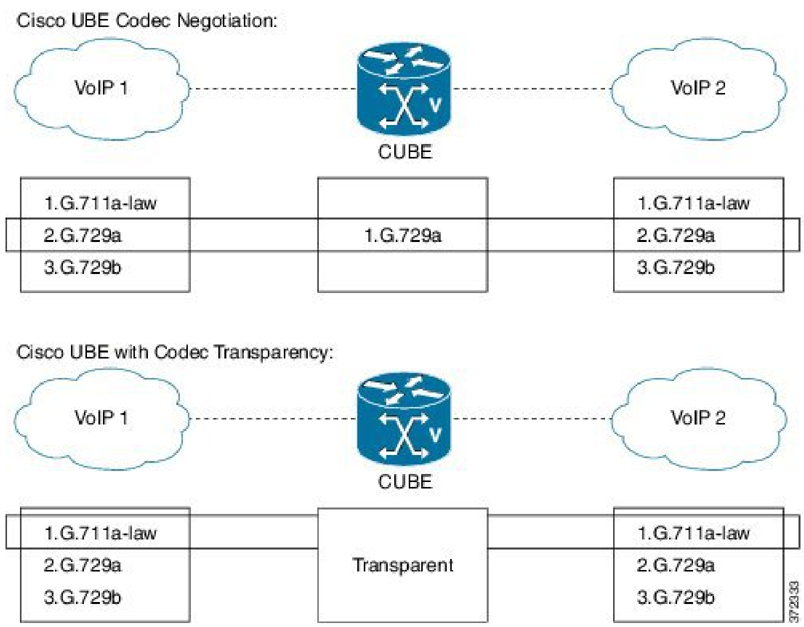
Cisco Unified Border Element (CUBE) uses codecs to compress digital voice samples to reduce bandwidth usage per call.

Configuring codecs allows the CUBE to act as a demarcation point on a VoIP network and allows calls on a specific dial peer to be established only if the desired codec criteria are satisfied. Also, preferences can be used to determine which codecs are selected over others.

If codec filtering is not required, CUBE also supports transparent codec negotiations. The codec filtering enables negotiations between endpoints with CUBE leaving the codec information untouched.

The following illustrations show how codec negotiation is performed on CUBE. Two VoIP clouds must be interconnected. In this scenario, both VoIP 1 and VoIP 2 networks have G.711 a-law that is configured as the preferred codec.

Figure 1: Codec Negotiation on CUBE



In the first example, the CUBE router is configured to use the G.729a codec. This can be done by using the appropriate codec command on both VoIP dial peers. When a call is set up, CUBE accepts only G.729a calls, thus influencing the codec negotiation.

In the second example, the CUBE dial peers are configured with a transparent codec and this leaves the codec information that is contained within the call signaling untouched. Because both VoIP 1 and VoIP 2 have G.711 a-law as their first choice, the resulting call is a G.711 a-law call.

Note H.323 protocol is no longer supported from Cisco IOS XE Bengaluru 17.6.1a onwards. Consider using SIP for multimedia applications.

Restrictions

While using the voice-class codec transparent, only the offer is passed transparently (without filtering). Codec filtering is carried out on SDP content in ANSWER messages. Only the first codec in the offer list is included in the outbound message.

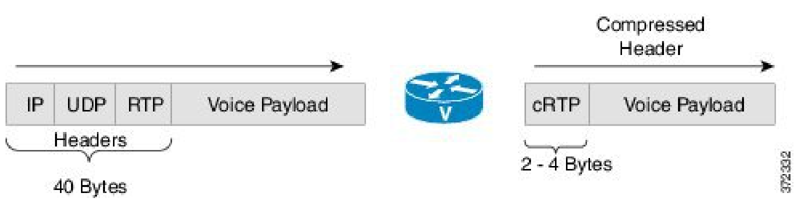
Note You can use 'pass-thru content sdp,' if you do not want to involve CUBE in the codec negotiation.

Media Transmission

When a VoIP call is established, the digitized audio and video samples must be transmitted. These samples are often called the media payload. The following media protocols specify the standards related to transmission of media payload in VoIP networks:

- Real-Time Transport Protocol (RTP)—RTP is a Layer 4 protocol that is encapsulated inside UDP segments. RTP carries the actual media voice samples in a call.
- Real-Time Control Protocol (RTCP)—RTCP is a companion protocol to RTP. Both RTP and RTCP operate at Layer 4 and are encapsulated in UDP. RTP and RTCP typically use UDP ports 16384 to 32767, though these ranges may vary according to hardware platform. RTP uses the even port numbers in that range, whereas RTCP uses the odd port numbers. While RTP is responsible for carrying the media payload, RTcP carries information about the RTP stream such as latency, jitter, packets, and octets sent and received.
- Compressed RTP (CRTP)—One of the challenges with RTP is its overhead. Specifically, the combined IP, UDP, and RTP headers are approximately 40 bytes in size, whereas a common voice payload size on a VoIP network is only 20 bytes, which includes 20 ms of voice by default. In that case, the header is twice the size of the payload. cRTP is used for RTP header compression and can reduce the 40-byte header to 2 or 4 bytes in size (depending on whether UDP checksums are in use), as shown in the figure below.

Figure 2: Compressed RTP



- Secure RTP (SRTP)—To help prevent an attacker from intercepting and decoding or possibly manipulating media packets, SRTP encrypts RTP packets. In addition, SRTP provides message authentication, integrity checking, and protection against replay attacks.

  VPN technology is used to protect traffic between sites. Sending SRTP via a VPN tunnel results in the encryption of traffic that is already secure, adding significant overhead and bandwidth needs. It is recommended that SRTP is used for media traffic, and that is excluded from VPN tunnels. By encrypting media directly on communicating endpoints, it is possible to realise a more secure solution, that requires less bandwidth and infrastructure investment.

Voice Activity Detection

Voice Activity Detection (VAD) is a technology that works with the human nature of voice conversations, mainly that one person listens while the other talks. VAD classifies traffic as speech, unknown, and silence. Speech and unknown payloads are transported, but silence is dropped. This accounts for approximately 30 percent savings in bandwidth over time.

VAD can significantly reduce the amount of bandwidth that is required by a media stream. However, VAD has a few negative attributes that must be considered. Because no packets are sent during silence, the listener can get the impression that the talker has been disconnected. Another characteristic is that it takes a moment for VAD to recognize the speech as having started again, and as a result, the first part of the sentence can be clipped. This can be annoying to the listening party. Music on Hold (MoH) and fax can also cause VAD to become ineffective because the media stream is constant.

VAD is enabled by default in CUBE dial peers as long as the codec selected supports it. VAD can be disabled at the VoIP dial peer using the **no vad** command. Some codecs, such as G.729b and G.729ab, support Comfort Noise Generation (CNG). When VAD is enabled, white noise is played to the listener during times when no packets are received. This leads the listener to believe that call is still connected. Cisco IP Phones and most gateways support CNG.

G.729 Annex-B and G.723.1 Annex-A include an integrated VAD function, but otherwise performs the same as G.729 and G.723.1, respectively.

Note VAD to NO-VAD calls is not supported by Cisco Unified Border Element (CUBE). This confirms that CUBE cannot generate comfort noise to fill periods of silence.

VoIP Bandwidth Requirements

The amount of bandwidth that is required varies by the codec and the transmission media. Two events require bandwidth. The media stream itself requires bandwidth between 17–106 kbps depending on codec, header compression, and Layer 2 and 3 headers. In addition, call signaling must be taken into account. While bandwidth required by call signaling is much smaller, problems occur when this traffic is dropped in congested networks.

Table 1: Protocol Header Size Assumptions

Protocol	Header size
IP	20 bytes
UDP	8 bytes
RTP	12 bytes
cRTP	Reduces size of IP, UDP, RTP to 2 or 4 bytes

The table below gives calculations for the default voice payload sizes in Cisco Unified Communications Manager or CUBE. For additional calculations, including different voice payload sizes and other protocols, use the TAC Voice Bandwidth Codec Calculator (registered customers only). For an explanation of each of the column headings, see the table below.

Table 2: Codec and Bandwidth Information

Codec & Bit Rate (kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Payload Size (ms) Packets Per Second (PPS)	Bandwidth Ethernet (kbps)
G.711 (64 kbps)	80	10	4.1	160	20	50	87.2
G.729 (8 kbps)	10	10	3.92	20	20	50	31.2

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G.723.1 (6.3 kbps)	24	30	3.9	24	30	33.3	21.9
G.723.1 (5.3 kbps)	20	30	3.8	20	30	33.3	20.8
G.726 (32 kbps)	20	5	3.85	80	20	50	55.2
G.726 (24 kbps)	15	5		60	20	50	47.2
G.728 (16 kbps)	10	5	3.61	60	30	33.3	31.5
G722_64k(64 kbps)	80	10	4.13	160	20	50	87.2
ilbc_mode_20(15.2 kbps)	38	20	3.95	38	20	50	38.4
ilbc_mode_30(13.33 kbps)	50	30	3.88	50	30	33.3	28.8
OPUS							

Table 3: Explanation of Terms

Codec Bit Rate (kbps)	Based on the codec, this is the number of bits per second that must be transmitted to deliver a voice call. (codec bit rate = codec sample size / codec sample interval).
Codec Sample Size (Bytes)	Size (Bytes) Based on the codec, this is the number of bytes captured by the digital signal processor (DSP) at each codec sample interval. For example, the G.729 coder operates on sample intervals of 10 ms, corresponding to 10 bytes (80 bits) per sample at a bit rate of 8 kbps. (codec bit rate = codec sample size / codec sample interval).
Codec Sample Interval (ms)	This is the sample interval at which the codec operates. For example, the G.729 coder operates on sample intervals of 10 ms, corresponding to 10 bytes (80 bits) per sample at a bit rate of 8 kbps. (codec bit rate = codec sample size / codec sample interval).
MOS	MOS is a system of grading the voice quality of telephone connections. With MOS, a wide range of listeners judge the quality of a voice sample on a scale of one (bad) to five (excellent). The scores are averaged to provide the MOS for the codec.
Voice Payload Size (Bytes)	The voice payload size represents the number of bytes (or bits) that are filled into a packet. The voice payload size must be a multiple of the codec sample size. For example, G.729 packets can use 10, 20, 30, 40, 50, or 60 bytes of voice payload size.

Voice Payload Size (ms)	Payload Size (ms) The voice payload size can also be represented in terms of the codec samples. For example, a G.729 voice payload size of 20 ms (two 10 ms codec samples) represents a voice payload of 20 bytes [(20 bytes * 8) / (20 ms) = 8 kbps]
PPS	PPS represents the number of packets that need to be transmitted every second in order to deliver the codec bit rate. For example, for a G.729 call with voice payload size per packet of 20 bytes (160 bits), 50 packets must be transmitted every second [50 pps = (8 kbps) / (160 bits per packet)]

Supported Audio and Video Codecs

CUBE is required to support the codec used between endpoints. g729r8 is supported by default. All other codecs have to be configured. The following codecs are supported:

Table 4: Audio Codecs Supported on CUBE

Codec Keyword	**Codec**
aacld	AACLD 90000 bps
clear-channel	Clear Channel 64000 bps (No voice capabilities: data transport only)
g711alaw	G.711 A Law 64000 bps
g711ulaw	G.711 u Law 64000 bps
g722-64	G722-64K 64000 bps
g723ar53	G.723.1 ANNEX-A 5300 bps (contains built-in VAD that cannot be disabled)
g723ar63	G.723.1 ANNEX-A 6300 bps (contains built-in VAD that cannot be disabled)
g723r53	G.723.1 5300 bps
g723r63	G.723.1 6300 bps
g726r16	G.726 16000 bps
g726r24	G.726 24000 bps
g726r32	G.726 32000 bps
g728	G.728 16000 bps
g729br8	G.729 ANNEX-B 8000 bps (contains built-in VAD that cannot be disabled)

Codec Keyword	Codec
g729r8	G.729 8000 bps
gsmamr-nb	GSM AMR-NB 4750 to 12200 bps (contains built-in VAD that cannot be disabled)
ilbc	iLBC 13330 or 15200 bps
isac	iSAC 10 to 32 kbps (variable bit-rate)
mp4a-latm	MP4A-LATM upto 128 kbps
opus	Opus upto 510 kbps
transparent	Transparent; uses the endpoint codec

Table 5: Video Codecs Supported on CUBE

Codec Keyword	Codec
h261	Video Codec H261
h263	Video Codec H263
h263+	Video Codec H263+
h264	Video Codec H264
mpeg4	Video Codec MPEG-4 ISO/IES 14496-2

Configure Codecs

Configure Voice Class Codec and Preference Lists

Preferences determine which codecs are selected over others.

A codec voice class is a construct within which a codec preference order is defined. A codec voice class can then be applied to a dial peer, which then follows its preference order.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class codec** *tag*
4. Add the following for each audio codec you want to configure in the voice class:
   - **codec preference** *value codec-type*[**profile** *profile-tag*]
   - **codec preference** *value codec-type*[**bytes** *payload-size* **fixed-bytes**]

- **codec preference** *value* **isac** [**mode {adaptive | independent}** [**bit-rate** *value* **framesize { 30 | 60 } [fixed]**]]
- **codec preference** *value* **ilbc** [**mode** *frame-size* [**bytes** *payload-size*]]
- **codec preference** *value* **mp4-latm** [**profile** *tag*]

5. Add the following to configure a video codec:
   - **video codec** *codec*
6. **exit**
7. **dial-peer voice** *number* **voip**
8. **voice-class codec** *tag* **offer-all**
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device> configure terminal`	Enters global configuration mode.
Step 3	**voice class codec** *tag* **Example:** `Device(config)# voice class codec 10`	Enters voice-class configuration mode for the specified codec voice class.
Step 4	Add the following for each audio codec you want to configure in the voice class: • **codec preference** *value* *codec-type*[**profile** *profile-tag*] • **codec preference** *value* *codec-type*[**bytes** *payload-size* **fixed-bytes**] • **codec preference** *value* **isac** [**mode {adaptive | independent}** [**bit-rate** *value* **framesize { 30 | 60 } [fixed]**]] • **codec preference** *value* **ilbc** [**mode** *frame-size* [**bytes** *payload-size*]] • **codec preference** *value* **mp4-latm** [**profile** *tag*]	Configure a codec within the voice class and specify a preference for the codec. This becomes part of a preference list.
Step 5	Add the following to configure a video codec: • **video codec** *codec* **Example:** `video codec h261`	Configures a video codec within the voice class.
Step 6	**exit**	Exits the current mode.

	Command or Action	Purpose
	Example: `Device(config-class)# exit`	• Enter your password if prompted.
Step 7	**dial-peer voice** *number* **voip** **Example:** `Device(config)# dial-peer voice 1 voip`	Enters dial peer configuration mode for the specified VoIP dial peer.
Step 8	**voice-class codec** *tag* **offer-all** **Example:** `Device(config-dial-peer)# voice-class codec 10`	Applies the previously configured voice class and associated codecs to a dial peer. • The **offer-all** keyword allows the device to offer all codecs configured in a codec voice class.
Step 9	**end** **Example:** `Device(config-dial-peer)# end`	Returns to privileged EXEC mode.

Configure Audio and Video Codecs at the Dial Peer Level

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. Enter one of the following to configure audio codec:
   - **codec** *codec* [**bytes** *payload-size* **fixed-bytes**]
   - **codec isac** [**mode {adaptive | independent}** [**bit-rate** *value* **framesize { 30 | 60 } [fixed]**]]
   - **codec ilbc** [**mode** *frame-size* [**bytes** *payload-size*]]
   - **codec mp4-latm** [**profile** *tag*]
   - **codec opus** [**profile** *tag*]
5. Add the following to configure a video codec:
   - **video codec** *codec*
6. (Optional) Enter one of the following to configure RTP payload type:
   - **rtp payload-type cisco-codec-isac** *number*
   - **rtp payload-type cisco-codec-ilbc** *number*
   - **rtp payload-type cisco-codec-video-h263+** *number*
   - **rtp payload-type cisco-codec-video-h264** *number*
   - **rtp payload-type opus** *number*
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device> configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *number* **voip** **Example:** `Device(config)# dial-peer voice 1 voip`	Enters dial peer configuration mode for the specified VoIP dial peer.

Step 4	Enter one of the following to configure audio codec:
	• **codec** *codec* [**bytes** *payload-size* **fixed-bytes**]
	• **codec isac** [**mode {adaptive | independent}** [**bit-rate** *value* **framesize { 30 | 60 } [fixed]**]]
	• **codec ilbc** [**mode** *frame-size* [**bytes** *payload-size*]]
	• **codec mp4-latm** [**profile** *tag*]
	• **codec opus** [**profile** *tag*]
	Example:
	For g711alaw Codec
	`Device(config-dial-peer)# codec g711alaw`
	Example:
	For ISAC Codec
	`Device(config-dial-peer)# codec isac mode independent`
		Configures an audio codec at the dial peer level.
		• By default, g729r8, 20-byte payload is configured.

Step 5	Add the following to configure a video codec: • **video codec** *codec* **Example:** For h2621 video codec `Device(config-dial-peer)# video codec h261`	Configures a video codec at the dial peer level.
Step 6	(Optional) Enter one of the following to configure RTP payload type: • **rtp payload-type cisco-codec-isac** *number* • **rtp payload-type cisco-codec-ilbc** *number* • **rtp payload-type cisco-codec-video-h263+** *number* • **rtp payload-type cisco-codec-video-h264** *number* • **rtp payload-type opus** *number* **Example:**	Configures the RTP payload type.

	Command or Action	Purpose
	`Device(config-dial-peer)# rtp payload-type opus 114`	
Step 7	**end** **Example:** `Device(config-dial-peer)# end`	Returns to privileged EXEC mode.

Verify an Audio Call

SUMMARY STEPS

1. **show call active voice [compact]**
2. **show call active voice [brief]**

DETAILED STEPS

Step 1 **show call active voice [compact]**

Displays a compact version of call information for voice calls in progress.

Example:

```
Router# show call active voice compact

 <callID>  A/O FAX T<sec> Codec       type        Peer Address       IP R<ip>:<udp>
Total call-legs: 2
        23 ANS     T3     mp4a-latm   VOIP        Psipp          9.45.33.11:57210
        24 ORG     T3     mp4a-latm   VOIP        P123           9.45.33.11:57210
```

Example:

```
Router# show call active voice compact

<callID>   A/O FAX T<sec> Codec    type    Peer Address    IP R<ip>:<udp>
Total call-legs: 2
58 ANS     T11          g711ulaw   VOIP     Psipp 2001:......:230A:6080
59 ORG     T11          g711ulaw   VOIP     P5000110011       10.13.37.150:6090
```

Step 2 **show call active voice [brief]**

Displays a brief summary of call information for voice calls in progress.

Configuration Examples for Codecs

Example: Configuring a Codec at Dial-Peer Level

```
Device(config)# dial-peer voice 5550199 voip
Device(config-dial-peer)# incoming called-number 5550199
```

```
Device(config-dial-peer)# codec g711ulaw
Device(config-dial-peer)# end
```

Example: Configuring a Codec Preference List and Applying it to a Dial Peer

```
Device(config)# voice class codec 100
Device(config-class)# codec preference 1 g711ulaw
Device(config-class)# exit
Device(config)# dial-peer voice 10 voip
Device(config-dial-peer)# voice-class codec 100
Device(config-dial-peer)# end
```

Example: Configuring a Codec Profile, Codec Preference List and Applying it to a Dial Peer for Opus Codec

```
router(config)#codec profile 79 opus
router(conf-codec-profile)#fmtp "fmtp:114 maxplaybackrate=16000; sprop-maxcapturerate=16000;
 maxaveragebitrate=20000; stereo=1; sprop-stereo=0; useinbandfec=0; usedtx=0"
router(conf-codec-profile)#exit

router(config)#voice class codec 80
router(config-class)#codec preference 1 opus profile 79
router(config-class)#exit

router(config)#dial-peer voice 604 voip
router(config-dial-peer)#rtp payload-type opus 126
router(config-dial-peer)#voice-class codec 80 offer-all
router(config-dial-peer)#exit
```