

Codec Support and Restrictions

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Overview

This chapter provides advanced information about the support of and restrictions for using certain codecs with CUBE. For basic information on how to configure codecs, refer to the Overview section.

Feature Information

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for Codec Support

Feature Name	Releases	Feature Information
Opus Codec Transcoding	Cisco IOS XE Bengaluru 17.6.1a	Transcoding support was introduced for Opus codec.

Feature Name	Releases	Feature Information
Opus Codec Negotiation	Cisco IOS XE Amsterdam 17.3.1	Opus audio codec support on CUBE was introduced. The following commands were introduced or modified as part of Opus codec feature: • **codec opus** [**profile** *tag*] • **codec profile** *tag profile* • **codec preference** *value codec-type* [**profile** *profile-tag*] • **rtp payload-type opus** *payload-type-number*
Secure Communications Interoperability Protocol (SCIP) support in CUBE (This feature is available in **'preview'** mode)	Cisco IOS XE 17.16.1a	This feature introduces SCIP audio and video codec support for secure communication on CUBE. The following commands were modified as part of the SCIP codec feature: • **video codec** *codec-type* • **codec preference** *value codec-type* • **rtp payload-type codec-audio-scip** *payload-type-number* • **rtp payload-type codec-video-scip** *payload-type-number*

Note **Preview Features Disclaimer:** In Cisco IOS XE 17.16.1a release, the Secure Communications Interoperability Protocol (SCIP) feature is available in 'preview' mode as it includes limited functionality or incomplete software dependencies. Cisco reserves the right to disable preview features at any time without notice. Cisco Technical Support provides reasonable effort support for features in preview mode. There is no Service Level Objective (SLO) in response times for features in preview mode; response times may be slow.

OPUS Codec

The Opus interactive speech and audio codec is designed to handle a wide range of interactive audio applications. This includes Voice over IP, video conferencing, and in-game chat. The OPUS codec also supports live, distributed music performances. It scales from low bitrate narrowband speech at 6 kbps to very high-quality stereo music at 510 kbps. Opus uses both Linear Prediction (LP) and Modified Discrete Cosine Transform (MDCT) algorithms to achieve good compression of both speech and music.

Note Opus codec is only supported for SIP-SIP call scenarios. It is not supported for TDM-SIP or Analog-SIP flows.

Opus Codec Configuration

- You can configure Opus codec handling with the following dial peer and voice class codec commands:
  - **codec opus** [**profile** *tag*] —The CLI command under **dial-peer** configuration mode is enhanced:

    ```
    router(config)#dial-peer voice 3002 voip
    router(config-dial-peer)#codec opus profile 2
    ```

  - **codec profile** *tag profile* —The CLI command that is configured under global configuration mode is enhanced to configure **opus** as a supported codec:

    ```
    router(config)#codec profile 2 opus
    router(conf-codec-profile)#fmtp "fmtp:114 maxplaybackrate=16000;
    sprop-maxcapturerate=16000; maxaveragebitrate=20000; stereo=1; sprop-stereo=0;
    useinbandfec=0; usedtx=0; cbr=0"
    router(conf-codec-profile)#exit
    ```

  - **codec preference** *value codec-type* [**profile** *profile-tag*]—The CLI command that is configured under voice class configuration mode is enhanced to configure **opus** as a preferred codec on the dial-peer:

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

  - **rtp payload-type** [**opus** *number*] —The CLI command that is configured under dial-peer configuration mode is enhanced to configure **opus** as a supported payload type:

    ```
    router(config)#dial-peer voice 604 voip
    router(config-dial-peer)#rtp payload-type opus 126
    ```

    - The default payload-type for **opus** is set to 114.
    - The default payload-type for **cisco-codec-aacld** is set to 112.

  - The CLI command **show call active voice** [**brief** | **compact**] is modified to include details on Opus codec in the command output.
  - The CLI command **show sip-ua calls** is modified to include details on Opus codec in the command output.

- Opus codec profile configuration is optional and can be used to define the initial offer in delayed offer to early offer call flows. Else, DSP-specific Opus parameters are used if the DSPFARM profile is configured with Opus codec.
- Transcoding between Opus and other codecs is available with PVDM4 DSP cards from Cisco IOS XE Bengaluru 17.6.1a onwards. Calls that require Opus transcoding are dropped by earlier releases.
- Opus Codec is supported for both secure and nonsecure calls (RTP-to-RTP, SRTP-to-SRTP, SRTP-to-RTP, and RTP-to-SRTP).
- Opus supports several clock rates. Only the highest clock rate of 48000 is advertised in the SDP. The following is a sample configuration of SDP for Opus codec:

  ```
  m=audio 16000 RTP/AVP 114       a=rtpmap:114 opus/48000/2
  ```

- Opus codec defines the optional media format (fmtp) parameters in a call. CUBE passes through the optional **fmtp** parameters from one side to the other if an Opus codec is configured on both sides of the call.

- Opus codec defines the following fmtp parameters:
  - **maxaveragebitrate**
  - **maxplaybackrate**
  - **stereo**
  - **useinbandfec**
  - **usedtx**
  - **sprop-maxcapturerate**
  - **sprop-stereo**
  - **cbr**
- Dynamic payload interworking is enabled by default on CUBE. Configure **asymmetric payload** [**full**] if CUBE is not required to handle payload interworking. In this scenario, interworking is handled at the endpoints handling the media.
- For interworking scenarios that support Opus codec, SDP Offer-Answer by CUBE must include default fmtp parameters that are required by DSP. The Offer-Answer includes the DSP-specific default parameters only if DSPFARM is configured with Opus codec.

  Example:

```
m=audio 4002 RTP/AVP 114

a=rtpmap:114 opus/48000/2

a=fmtp:114  maxaveragebitrate=32000

a=maxptime:20
```

Restrictions

- The Opus codec option is not available with the Extended Media Forking (XMF) API.
- CUBE doesn't support processing of multiple fmtp lines. If the received SDP has multiple fmtp lines, then only the first fmtp line is passed in the outbound INVITE.
- CUBE DSP doesn't support multiframes OPUS RTP packets for transcoding.

ISAC Codec Support on CUBE

The iSAC codec is an adaptive VoIP codec specially designed to deliver wideband sound quality in both low- and high-bit rate applications. The iSAC codec automatically adjusts the bit-rate for the best quality or a fixed bit rate can be used if the network characteristics are known. This codec is designed for wideband VoIP communications. The iSAC codec offers better quality with reduced bandwidth for sideband applications.

Restrictions

Low complexity is not supported for the iSAC codec.

AAC-LD MP4A-LATM Codec Support

As part of this feature, CUBE supports the following:

- Accept and send MP4A-LATM codec and corresponding FMTP profiles
- Configure MP4A-LATM under dial-peer or under voice-class codec as preferred codec
- Pass across real-time transport protocol (RTP) media for MP4A-LATM codec without any interworking
- Offer pre-configured FMTP profile for MP4A-LATM for DO-EO (Delayed-Offer to Early-Offer) calls
- Offer more than one FMTP profile (each with different payload type number) as mentioned by the offering endpoint, so that the answering endpoint can choose the best option.
- Offer only one instance of MP4A-LATM if media forking is applicable. The offered instance is the first one received in the offer.
- Calculate bandwidth for MP4A-LATM on the basis of either “b=TIAS” attribute or “bitrate” parameter in the FMTP attribute. If none of them are present in the session description protocol (SDP), the default maximum bandwidth, that is, 128 Kbps will be used for calculation.
- The following CUBE features are supported with the MP4A-LATM codec:
    - Basic call (audio and video) flow-around and flow-through (FA and FT).
    - Voice Class Codec support in CUBE with codec filtering
    - SRTP and SRTCP passthrough for SIP-to-SIP calls
    - Supplementary services
    - RSVP
    - Dynamic payload type interworking for DTMF and codec packets for SIP-to-SIP calls
    - Media Anti-Trombone with SIP signaling control on CUBE
    - Support for SIP UPDATE message per RFC 3311
    - RTP Media Loopback
    - Media forking for IP based calls using Zephyr recording server
    - CUBE Mid-call Re-INVITE consumption
    - Signaling forking (Fastweb multile SIP Early Dialog Support, FA and FT)
    - Maximum bandwidth-based CAC
    - Media Policing
    - Box-to-Box High Availability (B2B HA)
    - Inbox High Availability (Inbox HA)

Restrictions for AAC-LD MP4A-LATM Codec Support

CUBE does not support the following:

- Codec transcoding between MP4A-LATM and other codecs
- Dual-tone Multifrequency (DTMF) interworking with MP4A-LATM codec
- Non-SIP-SIP, that is, SIP to other service provider interface (SPI) interworking with MP4A-LATM codec

(Preview) SCIP Codec



Important In Cisco IOS XE 17.16.1a release, the Secure Communications Interoperability Protocol (SCIP) feature is available in 'preview' mode as it includes limited functionality or incomplete software dependencies. Cisco reserves the right to disable preview features at any time without notice. Cisco Technical Support provides reasonable effort support for features in preview mode. There is no Service Level Objective (SLO) in response times for features in preview mode; response times may be slow.

Secure Communication Interoperability Protocol (SCIP) is an application layer protocol that provides end-to-end capability exchange, packetization/de-packetization of media, reliable transport, and payload encryption.

It is an international standard for secure signaling message exchange between end devices to establish a secure traffic session.

SCIP handles packetization/de-packetization of the encrypted media and acts as a tunneling protocol. The CUBE receives encrypted SCIP payloads and transparently transmits them to the other end of the call using RTP.

SCIP Codec Configuration

- You can configure SCIP codec handling with the following dial peer and voice class codec commands:
  - **codec** *codec*—The CLI command configured in dial peer configuration mode is enhanced to support **scip** audio codec.

```
Router(config)# dial-peer voice 10 voip
Router(config-dial-peer)# codec scip
```

  - **video codec** *codec*—The CLI command configured in dial peer or in voice class configuration mode is enhanced to support **scip** video codec.

```
Router(config)# dial-peer voice 10 voip
Router(config-dial-peer)# video codec scip
```

```
Router(config)# voice class codec 10
Router(config-class)# video codec scip
```

  - **codec preference** *value codec-type*—The CLI command that is configured in voice class configuration mode is enhanced to support **scip** as a preferred codec on the dial-peer:

```
Router(config)#voice class codec 10
Router(config-class)#codec preference 1 scip
```

- **rtp payload-type** [**codec-audio-scip** *number*] —The CLI command that is configured in dial-peer configuration mode is enhanced to support **scip** audio payload type. The default payload-type for **codec-audio-scip** is 109:

```
Router(config)#dial-peer voice 60 voip
Router(config-dial-peer)#rtp payload-type codec-audio-scip 105
```

- **rtp payload-type** [**codec-video-scip** *number*] —The CLI command that is configured in dial-peer configuration mode is enhanced to support **scip** video payload type. The default payload-type for **codec-video-scip** is 110:

```
Router(config)#dial-peer voice 60 voip
Router(config-dial-peer)#rtp payload-type codec-video-scip 106
```

- The CLI command **show dial-peer voice** *tag* is modified to display RTP-payload with SCIP audio and video codec configuration details.
- The CLI command **show call active voice [brief | compact]** is modified to display SCIP codec audio details in the command output.
- The CLI command **show call active video [brief | compact]** is modified to display SCIP codec video details in the command output.
- The CLI command **show sip-ua calls** is modified to display Scip codec details in the command output.

- The media name in the SDP is audio or video and the encoding name is scip. The following is a sample configuration of SDP for Scip codec:

  An example for a sample SDP audio and video scip:

```
m=audio 10000 RTP/AVP 96
a=rtpmap:96 scip/8000

m=video 10002 RTP/AVP 96
a=rtpmap:96 scip/90000
```

- To maintain the same payload type with SCIP codec in outbound call, it is recommended to configure **asymmetric payload [full]** in global or dial-peer configuration mode.
- To pass-through the exact received RTCP content towards outbound direction, it is recommended to configure **rtcp all-pass-through** in global configuration mode.
- Disable Voice Activity Detection (VAD) while configuring in dial-peer and global configuration modes. By default, VAD is enabled.

  The following example shows how to configure VAD for a Voice over IP (VoIP) dial peer in global configuration mode:

```
dial-peer voice 200 voip
 vad
```

- Do not configure DTMF in dial-peer or in global configuration mode.

Limitations

CUBE does not support the following:

- Bandwidth-based Call Admission Control (CAC) calculation for SCIP audio and video codec.
- Transcoding
- DTMF interworking
- Multiple audio m-lines
- Application or image streams with SCIP codec.

Syslog Messages

From Cisco IOS XE 17.16.1a, the system generates syslog messages for the following scenarios:

Transcoding failure for SCIP calls

When an incoming offer message has only SCIP codec and the outbound dial-peer is configured with a different codec, such a call fails as the transcoder is not supported. Generates the following syslog message:

SIP-3-TRANSCODER_FAILURE: Failed to attach transcoder for call-id = 3

SCIP application m-line rejection

CUBE rejects SDP with SCIP application m-line, as application m-line for payload is not supported, generates the following syslog message:

SIP-3-APPLICATION_MEDIA_FAILURE: Unsupported application m-line for payload = scip, call-id = 32

Note CUBE supports only application m-line with H224 payload.

SIP (SCIP) to TDM call failures

SIP (SCIP codec) to TDM call fails during offer answer negotiation. Generates the following syslog message:

SIP-3-SIP_TDM_FAILURE: Unsupported codec: scip for TDM to SIP Calls, Call-id: 521

DTMF Interworking of Inband

SCIP (Inband DTMF) interworking with SCIP (Out-of band DTMF) is not supported, generates a syslog with the following error string:

SIP-3-DTMF_INTERWORKING_FAILURE: DTMF interworking failure for dtmf1 = inband-voice, dtmf2 = rtp-nte, call-id = 1