Codec Preference Lists

Overview

This chapter describes how to negotiate an audio codec from a list of codec associated with a preference. This chapter also describes how to disable codec filtering by configuring Cisco Unified Border Element (CUBE) to send an outgoing offer with all configured audio codecs in the list assuming that the dspfarm supports all these codecs.

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

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 Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element

Feature Name	Releases	Feature Information
Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the CUBE	Baseline Functionality	The following command was introduced or modified: **voice-class codec (dial peer).**

Codecs Configured Using Preference Lists

The list of codecs supported in a SIP-to-SIP call can be configured with a Codec Preference List on each SIP call leg with following properties:

- Incoming and outgoing dial-peers can be configured with different preference lists.
- Midcall codec changes for supplementary services are supported with preference lists. Transcoder resources are dynamically inserted or deleted when there is a codec or RTP-NTE to in-band DTMF interworking required.
- Reinvite-based supplementary services that are invoked from the Cisco Unified Communications Manager (CUCM), like call hold, call resume, Music On Hold (MOH), call transfer, and call forward are supported with preference lists.
- T.38 fax and fax passthrough switchover with preference lists are supported.
- Reinvite-based call hold and call resume for the Secure Real-Time Transfer protocol (SRTP) and Real-Time Transport Protocol (RTP) interworking on CUBE is supported with preference lists.
- High availability is supported for calls that use codecs with preference lists. But calls requiring the transcoder to be invoked are not checkpointed. During midcall renegotiation, if the call releases the transcoder, then the call is checkpointed.

Restrictions

For All Calls (SIP-to-SIP calls)

- Video codecs are not supported with preference lists.
- Multiple audio streams are not supported.
- Codec re-packetization feature is not supported when preference lists are configured.

Note Codec preference in the voice class codec on the outgoing call leg is not followed when the same codecs are available in the respective incoming invite with SDP with different codec preference. Cube prioritizes and follows the incoming invite with SDP codec preference when compared to the voice class codec preference on the outgoing dial-peer leg.

Configure Audio Codecs Using a Codec Voice Class and Preference Lists

Preferences can be used to determine which codecs will be selected over others.

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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class codec** *tag*
4. Do 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. **exit**
6. **dial-peer voice** *number* **voip**
7. **voice-class codec** *tag*
8. **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	Do the following for each audio codec you want to configure in the voice class: • **codec preference** *value codec-type*[**profile** *profile-tag*]	Configure a codec within the voice class and specifies a preference for the codec. This becomes part of a preference list

	Command or Action	Purpose
	• **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*]	
Step 5	**exit** **Example:** `Device(config-class)# exit`	Exits the current mode. • Enter your password if prompted.
Step 6	**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 7	**voice-class codec** *tag* **Example:** `Device(config-dial-peer)# voice-class codec 10`	Applies the previously configured voice class and associated codecs to a dial peer.
Step 8	**end** **Example:** `Device(config-dial-peer)# end`	Returns to privileged EXEC mode.

Disable Codec Filtering

CUBE is configured to filter common codecs for the subsets, by default. The filtered codecs are sent in the outgoing offer. You can configure the CUBE to offer all the codecs configured on an outbound leg instead of offering only the filtered codecs.

Note This configuration is applicable only for early offer calls from the CUBE. For delayed offer calls, by default all codecs are offered irrespective of this configuration.

Perform this task to disable codec filtering and allow all the codecs configured on an outbound leg.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class codec** *tag* **offer-all**
5. **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** *tag* **voip** **Example:** `Device(config)# dial-peer voice 10 voip`	Enters dial peer voice configuration mode.
Step 4	**voice-class codec** *tag* **offer-all** **Example:** `Device(config-dial-peer)# voice-class codec 10 offer-all`	Adds all the configured voice class codec to the outgoing offer from the CUBE.
Step 5	**end** **Example:** `Device(config-dial-peer)# end`	Exits the dial peer voice configuration mode.

Troubleshoot Negotiation of an Audio Codec from a List of Codecs

Use the following commands to debug any errors that you may encounter when you configure the Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the CUBE feature:

- **debug ccsip all**
- **debug voip ccapi input**
- **debug voip rtp session**

For DSP-related debugs, use the following commands:

- **debug voip dsmp all**
- **debug voip dsmp rtp both payload all**
- **debug voip ipipgw**

Verify Negotiation of an Audio Codec from a List of Codecs

Perform this task to display information to verify Negotiation of an Audio Codec from a List of Codecs on Each Leg of a SIP-to-SIP Call on the Cisco Unified Border Element configuration. These **show** commands need not be entered in any specific order.

SUMMARY STEPS

1. **enable**
2. **show call active voice brief**
3. **show voip rtp connections**
4. **show dspfarm dsp active**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

Step 2 **show call active voice brief**

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

Example:

```
Device# show call active voice brief
<ID>: <CallID> <start>ms.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
 media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
 long duration call detected:<y/n> long duration call duration :<sec> timestamp:<time>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
   last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  <codec> (payload size)
 Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm
  MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
         speeds(bps): local <rx>/<tx> remote <rx>/<tx>
 Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
 bw: <req>/<act> codec: <audio>/<video>
  tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
 rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4
1243 : 11 971490ms.1 +-1 pid:1 Answer 1230000 connecting
 dur 00:00:00 tx:415/66400 rx:17/2561
 IP 192.0.2.1:19304 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
```

```
 long duration call detected:n long duration call duration:n/a timestamp:n/a
1243 : 12 971500ms.1 +-1 pid:2 Originate 3210000 connected
 dur 00:00:00 tx:5/10 rx:4/8
 IP 9.44.26.4:16512 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729br8 TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
0    : 13 971560ms.1 +0 pid:0 Originate  connecting
 dur 00:00:08 tx:415/66400 rx:17/2561
 IP 192.0.2.2:2000 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
0    : 15 971570ms.1 +0 pid:0 Originate  connecting
 dur 00:00:08 tx:5/10 rx:3/6
 IP 192.0.2.3:2000 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729br8 TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 2
Multicast call-legs: 0
Total call-legs: 4
```

Step 3 **show voip rtp connections**

Displays Real-Time Transport Protocol (RTP) connections.

Example:

```
Device# show voip rtp connections
VoIP RTP active connections :
No. CallId     dstCallId  LocalRTP RmtRTP     LocalIP                                RemoteIP
1     11         12         16662    19304    192.0.2.1
192.0.2.2
2     12         11         17404    16512    192.0.2.2
192.0.2.3
3     13         14         18422    2000     192.0.2.4
9.44.26.3
4     15         14         16576    2000     192.0.2.6
192.0.2.5
Found 4 active RTP connections
```

Step 4 **show dspfarm dsp active**

Displays active DSP information about the DSP farm service.

Example:

```
Device# show dspfarm dsp active
SLOT DSP VERSION  STATUS CHNL USE   TYPE    RSC_ID BRIDGE_ID PKTS_TXED PKTS_RXED
0    1   27.0.201 UP     1    USED  xcode   1      0x9         5         8
0    1   27.0.201 UP     1    USED  xcode   1      0x8         2558      17
Total number of DSPFARM DSP channel(s) 1
```