Negotiation of an Audio Codec from a List of Codecs

The 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 supports negotiation of an audio codec using the Voice Class Codec and Codec Transparent infrastructure on the Cisco Unified Border Element (Cisco UBE).

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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.

Benefits

Following are the benefits of the 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:

- You can configure dissimilar Voice Class Codec configurations on the incoming and outgoing dial peers.

- Both normal transcoding and high-density transcoding are supported with the Voice Class Codec configuration.
- Mid-call codec changes for supplementary services are supported with the Voice Class Codec configuration. Transcoder resources are dynamically inserted or deleted when required.
- Reinvite-based supplementary services 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 the Voice Class Codec configuration.
- T.38 fax and fax passthrough switchover with Voice Class Codec configuration are supported.
- Reinvite-based call hold and call resume for Secure Real-Time Transfer protocol (SRTP) and Real-Time Transport Protocol (RTP) interworking on Cisco UBE are supported with the Voice Class Codec configuration.
- High availability support for calls that use Voice Class Codec, but calls that require transcoder to be invoked are not checkpointed. During mid-call renegotiation, if the call releases the transcoder, then the call is checkpointed.

Prerequisites for Negotiation of an Audio Codec from a List of Codecs

To the configure 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 you must know the following:

- Transcoding configuration on the Cisco UBE.
- The digital signal processor (DSP) requirements to support the transcoding feature on the Cisco UBE.
- The existing Voice Class Codec configuration on the dial peers.

Cisco Unified Border Element

- Cisco IOS Release 15.1(2)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.8S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions for Negotiation of an Audio Codec from a List of Codecs

The 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 has the following limitations:

- Mid-call insertion or deletion of the transcoder with voice class codec for H323-H323 and H323-SIP is not supported.
- Voice class codec is not supported for video calls.

Disabling Codec Filtering

Cisco UBE is configured to filter common codecs for the subsets, by default. The filtered codecs are sent in the outgoing offer. You can configure the Cisco UBE 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 Cisco UBE. 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.

	Command or Action	Purpose
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 Cisco UBE.
Step 5	**end** **Example:** `Device(config-dial-peer)# end`	Exits the dial peer voice configuration mode.

Troubleshooting 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 Cisco Unified Border Element feature:

- **debug ccsip all**
- **debug voip ccapi input**
- **debug sccp messages**
- **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**

Verifying 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 sccp connections**
5. **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 sccp connections**

Displays information about the connections controlled by the Skinny Client Control Protocol (SCCP) transcoding and conferencing applications.

Example:

```
Device# show sccp connections
sess_id    conn_id      stype mode     codec   sport rport ripaddr
5          5            xcode sendrecv g729b   16576 2000  192.0.2.3
5          6            xcode sendrecv g711u   18422 2000  192.0.2.4
Total number of active session(s) 1, and connection(s) 2
```

Step 5 **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
```

Feature Information for Negotiation of an Audio Codec from a List of Codecs

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 Cisco Unified Border Element	15.1(2)T	The 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 supports negotiation of an audio codec using the Voice Class Codec and Codec Transparent infrastructure on the Cisco UBE. The following command was introduced or modified: **voice-class codec (dial peer).**
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	Cisco IOS XE Release 3.8S	The 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 supports negotiation of an audio codec using the Voice Class Codec and Codec Transparent infrastructure on the Cisco UBE. The following command was introduced or modified: **voice-class codec (dial peer)**.
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.	15.3(2)T	This feature provides high availability support 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 under the Voice Class Codec.