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).

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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 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:
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**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td></td>
<td>Device&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td>dial-peer voice tag voip</td>
<td>Enters dial peer voice configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Device(config)# dial-peer voice 10 voip</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
<td></td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>voice-class codec tag [offer-all]</td>
<td>Adds all the configured voice class codec to the outgoing offer from the Cisco UBE.</td>
<td></td>
</tr>
</tbody>
</table>

**Example:**

```plaintext
Device(config-dial-peer)# voice-class codec 10 offer-all
```

<table>
<thead>
<tr>
<th><strong>Step 5</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>end</td>
<td>Exits the dial peer voice configuration mode.</td>
</tr>
</tbody>
</table>

**Example:**

```plaintext
Device(config-dial-peer)# end
```

---

**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 ccap 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
   speed(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>
x: <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 cntrl rcvd:<n> timestamp:<n>
   long duration call detected:<n> long duration call duration:<n> timestamp:<n>
1243 : 12 971500ms.1 +-1 originate 3210000 connected
dur 00:00:00 tx:5/10 rx:4/8
IP 9.44.26.1:16512 SRTP: off rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g729br8 TextRelay: off
media inactive detected:<n> media cntrl rcvd:<n> timestamp:<n>
   long duration call detected:<n> long duration call duration:<n> timestamp:<n>
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 cntrl rcvd:<n> timestamp:<n>
   long duration call detected:<n> long duration call duration:<n> timestamp:<n>
0 : 15 971570ms.1 +0 pid:0 originate connecting
Negotiation of an Audio Codec from a List of Codecs

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 1 11 12 16662 19304 192.0.2.1 192.0.2.1
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 0 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

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>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</td>
<td>15.1(2)T</td>
<td>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: <code>voice-class codec (dial peer)</code>.</td>
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<tr>
<td>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</td>
<td>Cisco IOS XE Release 3.8S</td>
<td>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: <code>voice-class codec (dial peer)</code>.</td>
</tr>
<tr>
<td>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</td>
<td>15.3(2)T</td>
<td>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.</td>
</tr>
</tbody>
</table>