iLBC Support for SIP and H.323

The internet Low Bitrate Codec (iLBC) is a standard, high-complexity speech codec suitable for robust voice communication over IP. The iLBC has built-in error correction functionality that helps the codec perform in networks with high-packet loss. This codec is supported on both Session Initiation Protocol (SIP) and H.323.

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.

Prerequisites for iLBC Support for SIP and H.323

Cisco Unified Border Element

- Cisco IOS Release 12.2(11)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 2.5 or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions for iLBC Support for SIP and H.323

The iLBC Support for SIP and H.323 feature is supported on the following:

- IP-to-IP gateways with no transcoding and conferencing
- All c5510 DSP-based platforms

Information About iLBC Support for SIP and H.323

The internet Low Bit Rate Codec (iLBC) is designed for narrow band speech and results in a payload bit rate of 13.33 kbits per second for 30-millisecond (ms) frames and 15.20 kbits per second for 20 ms frames.

When the codec operates at block lengths of 20 ms, it produces 304 bits per block, which is packetized as defined in RFC 3952. Similarly, for block lengths of 30 ms it produces 400 bits per block, which is packetized as defined in RFC 3952.

The iLBC has built-in error correction functionality to provide better performance in networks with higher packet loss.

How to Configure an iLBC Codec

Configuring an iLBC Codec on a Dial Peer

The iLBC is intended for packet-based communication. Perform the following steps to configure the iLBC codec on a dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **rtp payload-type cisco-codec-ilbc** [*number*
5. **codec ilbc** [**mode** *frame_size* [**bytes** *payload_size*]]
6. **exit**

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 configuration mode for the VoIP dial peer designated by *tag*.

Step 4	**rtp payload-type cisco-codec-ilbc** [*number*
	
	Example:
	
	`Device(config-dial-peer)# rtp payload-type cisco-codec-ilbc 100`
		Identifies the payload type of a Real-Time Transport Protocol (RTP) packet. Keyword and argument are as follows:
		
		• **cisco-codec-ilbc** [*number*]--Payload type is for internet Low Bit Rate Codec (iLBC). Range: 96 to 127. Default: 116.
		
		Note Do not use the following numbers because they have preassigned values: 96, 97, 100, 117, 121 to 123, and 125 to 127. If you use these values, the command will fail. You must first reassign the value in use to a different unassigned number, for example:
		
		`rtp payload-type nse 105`
		`rtp payload-type cisco-codec-ilbc 100`

Step 5	**codec ilbc** [**mode** *frame_size* [**bytes** *payload_size*]] **Example:** `Device(config-dial-peer)# codec ilbc mode 30 bytes 200`	Specifies the voice coder rate of speech for a dial peer. Keywords and arguments are as follows: • **mode** *frame_size* --The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20. • **bytes** *payload_size* --Number of bytes in an RTP packet. For mode 20, valid values are 38 (default), 76, 114, 152, 190, and 228. For mode 30, valid values are 50(default), 100, 150, and 200.
Step 6	**exit** **Example:** `Device(config-dial-peer)# exit`	Exits the current mode.

Configuring an iLBC Codec in the Voice Class

When using multiple codecs, you must create a voice class in which you define a selection order for codecs; then, you can apply the voice class to VoIP dial peers. The **voice class codec** global configuration command allows you to define the voice class that contains the codec selection order. Then, use the **voice-class codec** dial-peer configuration command to apply the class to individual dial peers.

To configure an iLBC in the voice class for multiple-codec selection order, perform the following steps.

You can configure more than one voice class codec list for your network. Configure the codec lists and apply them to one or more dial peers based on which codecs (and the order) you want supported for the dial peers. Define a selection order if you want more than one codec supported for a given dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class codec** *tag*
4. **codec preference** *value* **ilbc** [**mode** *frame_size*] [**bytes** *payload_size*]
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **voice-class codec** *tag*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enters 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 99`	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class. The argument is as follows: • *tag* --Unique identifier on the router. Range is 1 to 10000.

	Command or Action	Purpose
Step 4	**codec preference** *value* **ilbc** [**mode** *frame_size*] [**bytes** *payload_size*] **Example:** `Device(config-voice-class)# codec preference 1 ilbc 30 200`	Specifies a list of preferred codecs to use on a dial peer. Keywords and arguments are as follows: • *value* --Order of preference, with 1 being the most preferred and 14 being the least preferred. • **mode** *frame_size* --The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20. • **bytes** *payload_size* --Number of bytes in an RTP packet. For mode 20, valid values are 38 (default), 76, 114, 152, 190, and 228. For mode 30, valid values are 50(default), 100, 150, and 200.
Step 5	**exit** **Example:** `Device(config-voice-class)# exit`	Exits the current mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 16 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 99`	Assigns a previously configured codec selection preference list (the codec voice class that you defined in step 3) to the specified VoIP dial peer. **Note** The **voice-class codec**command in dial-peer configuration mode contains a hyphen. The **voice class** command in global configuration mode does not contain a hyphen.
Step 8	**exit** **Example:** `Device(config-dial-peer)# exit`	Exits the current mode.

Verifying iLBC Support for SIP and H.323

You can use the following commands to check iLBC status:

- **show voice call summary**
- **show voice call status**

- **show voice dsmp stream**
- **show call active voice**
- **show call history voice**
- **show voice dsp and its extensions**
- **show dial-peer voice**
- **show voice dsp channel operational-status**

Feature Information for iLBC Support for SIP and H.323

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.

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Table 1: Feature Information for iLBC Support for SIP and H.323

Feature Name	Releases	Feature Information
iLBC Support for SIP and H.323	12.2(11)T 12.2(15)T	The iLBC is a standard, high-complexity speech codec suitable for robust voice communication over IP. The iLBC has built-in error correction functionality that helps the codec perform in networks with high-packet loss. This codec is supported on both Session Initiation Protocol (SIP) and H.323. The following commands were introduced or modified: **codec ilbc**, **codec preference**, and **rtp payload-type**.

Feature Name	Releases	Feature Information
iLBC Support for SIP and H.323	Cisco IOS XE Release 2.5	The iLBC is a standard, high-complexity speech codec suitable for robust voice communication over IP. The iLBC has built-in error correction functionality that helps the codec perform in networks with high-packet loss. This codec is supported on both Session Initiation Protocol (SIP) and H.323. The following commands were introduced or modified: **codec ilbc**, **codec preference**, and **rtp payload-type**.