G.722-64 and iLBC Codec Support on Cisco Unified Border Elements, DSP Farms, and Voice Gateways

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The G.722-64 and iLBC codecs are supported for Cisco Unified Border Elements (Cisco UBEs), DSP farms, and voice gateways. Conferencing and universal transcoding are supported on both codecs.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. To reach links to specific feature documentation in this module and to see a list of the releases in which each feature is supported, use the "Feature Information for G.722-64 and iLBC Codec Support on Cisco UBEs, DSP Farms, and Voice Gateways" section on page 18.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

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How to Configure G.722-64 and iLBC Codecs for Voice Gateways

The G.722-64 and iLBC codecs can be used to enable conferencing and transcoding on Cisco IOS voice gateways in a Cisco Unified Communications Manager network. Digital signal processor (DSP) farms provide conferencing and transcoding services using DSP resources on high-density digital voice/fax network modules.

To configure conferencing and transcoding for voice gateway routers, see the "Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter of the *Cisco CallManager and Cisco IOS Interoperability Guide*.

For more information on configuring iLBC codecs for H.323 and SIP, see the "Dial Peer Overview" chapter and "Dial Peer Features and Configuration" chapter in *Dial Peer Configuration on Voice Gateway Routers*

The following changes apply to this chapter:

Codecs

End-user devices must be equipped with one of the following codecs:

Codec	Packetization Periods for Transcoding (ms)
G.711 a-law, G.711 u-law, G.722-64	10, 20, or 30
G.729, G.729A, G.729B, G.729AB	10, 20, 30, 40, 50, or 60
iLBC	20 or 30

Conferencing and Transcoding Session Capacities

Each DSP is individually configurable to support either conferencing or transcoding and standard voice termination. The total number of conferencing, transcoding, and voice termination sessions is limited by the capacity of the entire system, which includes the DSPs, hardware platform, physical voice interface, and Cisco Unified Communications Manager.

Table 1 and Table 2 list the maximum number of conference calls and transcoding sessions that DSPs can handle, in theory. Actual capacity may be less based on the total system design.

Table 1 DSP Theoretical Session Capacities

Application	NM-HD-1V/2V (1 DSP)	NM-HD-2VE (3 DSPs)	NM-HDV2 (16 DSPs)	2801/2811 (2 PVDM2-64)	2821/2851 (3 PVDM2-64)	3825, 3845 (4 PVDM2-64)
Conferencing						
G.711	8 sessions (64 conferees)	24 sessions (192 conferees)	50 sessions (400 conferees)	50 sessions (400 conferees)	50 sessions (400 conferees)	50 sessions (400 conferees)
G.722-64	2 sessions (16 conferees)	6 sessions (48 conferees)	32 sessions (256 conferees)	16 sessions (128 conferees)	24 sessions (192 conferees)	32 sessions (256 conferees)
G.729	2 sessions (16 conferees)	6 sessions (48 conferees)	32 sessions (256 conferees)	16 sessions (128 conferees)	24 sessions (192 conferees)	32 sessions (256 conferees)
iLBC	1 session (8 conferees)	3 sessions (24 conferees)	16 sessions (128 conferees)	8 sessions (64 conferees)	12 sessions (96 conferees)	16 sessions (128 conferees)

Table 1 DSP Theoretical Session Capacities (continued)

Application	NM-HD-1V/2V (1 DSP)	NM-HD-2VE (3 DSPs)	NM-HDV2 (16 DSPs)	2801/2811 (2 PVDM2-64)	2821/2851 (3 PVDM2-64)	3825, 3845 (4 PVDM2-64)
Transcoding						
G.711 a-law/u-law <-> G.729a/G.729ab/ GSM FR	8 sessions	24 sessions	128 sessions	64 sessions	96 sessions	128 sessions
G.711 a-law/u-law <-> G.729/G.729b/ GSM EFR	6 sessions	18 sessions	96 sessions	48 sessions	72 sessions	96 sessions
G.722-64 <-> G.711	8 sessions	24 sessions	128 sessions	64 sessions	96 sessions	128 sessions
G.722-64<-> any	4 sessions	12 sessions	64 sessions	32 sessions	48 sessopms	64 sessions
iLBC <-> G.711	6 sessions	18 sessions	96 sessions	48 sessions	72 sessions	96 sessions
iLBC <-> any	3 sessions	9 sessions	48 sessions	24 sessions	36 sessions	48 sessions
Voice Termination						
G.711 a-law/u-law	16 sessions	48 sessions	256 sessions	128 sessions	192 sessions	256 sessions
G.722-64, G.726, G.729a, G.729ab, iLBC	8 sessions	24 sessions	128 sessions	64 sessions	96 sessions	128 sessions
G.729, G.729b, G.723.1, G.728	6 sessions	18 sessions	96 sessions	48 sessions	72 sessions	96 sessions

Table 2 Theoretical System Capacities for One DSP

Application	G.711 a-law/u-law	G.722-64	G729 a/ab	G.729, G.729b	iLBC
Conferencing	8 sessions (8 x 8 = 64 conferees)	2 sessions (8 x 2 = 16 conferees)	2 sessions (8 x 2 = 16 conferees)	2 sessions (8 x 2 = 16 conferees)	1 session (1 x 8 = 8 conferees)
Conferencing on PVDM2-8	4 sessions (4 x 8 = 32 conferees)	1 session (1 x 8 = 8 conferees)	1 session (1 x 8 = 8 conferees)	1 session (1 x 8 = 8 conferees)	1 session (1 x 8 = 8 conferees)
Hardware MTP	16 sessions	—	—	—	—
Transcoding	8 sessions	8 sessions	8 sessions	6 sessions	8 sessions

How to Configure G.722-64 and iLBC Codecs for Cisco Unified Border Elements

The G.722-64 and iLBC codecs can be used to set up transcoding on Cisco Unified Border Elements (Cisco UBEs). To configure these codecs on a Cisco UBE, see the "Fundamental Cisco Multiservice IP-to-IP Gateway Configuration" chapter of the *Cisco Multiservice IP-to-IP Gateway* document.

Additional References

The following sections provide references related to G.722-64 and iLBC Codec Support on Cisco UBEs, DSP farms, and voice gateways.

Related Documents

Related Topic	Document Title
Conferencing and transcoding for voice gateways	*Cisco Communications Manager and Cisco IOS Interoperability Guide*
Transcoding for Cisco Unified Communications Manager Express	*Cisco Unified Communications Manager Express System Administrator Guide*
Transcoding for Cisco Unified Border Elements	*Cisco Multiservice IP-to-IP Gateway*
Dial-peer configuration	*Dial Peer Configuration on Voice Gateway Routers*

Standards

Standard	Title
H245, Annex S	*Control protocol for multimedia communication*

MIBs

MIB	MIBs Link
• CISCO-VOICE-COMMON-DIAL-CONTROL-MIB.my • CISCO-VOICE-DIAL-CONTROL-MIB.my	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC3951	Internet Low Bit Rate Codec (iLBC)
RFC3952	Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech

Technical Assistance

Description	Link
The Cisco Technical Support & Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Command Reference

This section documents new and modified commands:

- codec (dial-peer)
- codec (DSP Farm profile)
- codec preference

codec (dial-peer)

To specify the voice coder rate of speech for a dial peer, use the **codec** command in dial-peer configuration mode. To reset the default value, use the **no** form of this command.

Cisco 1750 and Cisco 1751 Modular Access Routers, Cisco AS5300 and AS5800 Universal Access Servers, and Cisco MC3810 Multiservice Concentrators

codec *codec* [**bytes** *payload_size*]

no codec *codec* [**bytes** *payload_size*]

Cisco 2600 and 3600 Series Routers and Cisco 7200 and 7500 Series Routers

codec {*codec* [**bytes** *payload_size*] | **transparent**}

no codec {*codec* [**bytes** *payload_size*] | **transparent**}

Syntax Description

codec	Codec options available for the various platforms are described in Table 3, below.
bytes	(Optional) Specifies the number of bytes in the voice payload of each frame.
payload-size	(Optional) Number of bytes in the voice payload of each frame. See Table 4 for valid entries and default values.
transparent	Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element. **Note** The **transparent** keyword is only available on the Cisco 2600 and 3600 Series Router and Cisco 7200 and 7500 Series Router platforms.

Table 3 Codec support by platform

Codec	Cisco 1750 and Cisco 1751 Modular Access Routers	Cisco 2600 and 3600 Series Routers and Cisco 7200 and 7500 Series Routers	Cisco AS5300 and AS5800 Universal Access Servers	Cisco MC3810 Multiservice Concentrators
clear-channel—Clear channel at 64,000 bits per second (bps)	Yes	Yes	—	Yes
g711alaw—G.711 A-Law at 64,000 bps	Yes	Yes	Yes	Yes
g711ulaw—G.711 u-Law at 64,000 bps	Yes	Yes	Yes	Yes
g722-64—G.722-64 at 64,000 bps	Yes	Yes	Yes	—
g723ar53—G.723.1 Annex A at 5300 bps	—	Yes	Yes	Yes
g723ar63—G.723.1 Annex A at 6300 bps	—	Yes	Yes	Yes
g723r53—G.723.1 at 5300 bps	—	Yes	Yes	Yes
g723r63—G.723.1 at 6300 bps	—	Yes	Yes	Yes
g726r16—G.726 at 16,000 bps	Yes	Yes	Yes	Yes

Table 3 Codec support by platform (continued)

Codec	Cisco 1750 and Cisco 1751 Modular Access Routers	Cisco 2600 and 3600 Series Routers and Cisco 7200 and 7500 Series Routers	Cisco AS5300 and AS5800 Universal Access Servers	Cisco MC3810 Multiservice Concentrators
g726r24—G.726 at 24,000 bps	Yes	Yes	Yes	Yes
g726r32—G.726 at 32,000 bps	Yes	Yes	Yes	Yes
g726r53—G.726 at 53,000 bps	Yes	Yes	Yes	—
g726r63—G.726 at 63,000 bps	Yes	Yes	Yes	—
g728—G.728 at 16,000 bps	—	Yes	Yes	Yes
g729abr8—G.729 Annex A and B at 8000 bps	Yes	Yes	Yes	Yes
g729ar8—G729 Annex A at 8000 bps	Yes	Yes	Yes	Yes
g729br8—G.729 Annex B at 8000 bps	Yes	Yes	Yes	Yes
g729r8—G.729 at 8000 bps. This is the default codec	Yes	Yes	Yes	Yes

Defaults

g729r8, 30-byte payload for VoFR and VoATM
g729r8, 20-byte payload for VoIP
See Table 4 for valid entries and default values.

Command Modes

dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
11.3(3)T	This command was implemented on the Cisco 2600 series.
12.0(3)T	This command was implemented on the Cisco AS5300. This release does not support the **clear-channel** keyword.
12.0(4)T	This command was implemented on the Cisco 3600 series, Cisco 7200 series and the Cisco MC3810. This release modified the command for VoFR dial peers.
12.0(5)XE	Additional *codec* choices and other options were implemented.
12.0(5)XK	The **g729br8** and **pre-ietf** *codec* choices were added for the Cisco 2600 and Cisco 3600 series.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0.(7)T and implemented on the Cisco AS5800. Additional voice coder rates of speech were added. This release does not support the **clear-channel** keyword on this platform.
12.0(7)XK	The **g729abr8** and **g729ar8** *codec* choices were for the Cisco MC3810, and the keyword **pre-ietf** was deleted.
12.1(1)T	This command was integrated in Cisco IOS Release 12.1(1)T.
12.1(5)T	The **gsmefr** and **gsmfr** codec keywords were added.

Release	Modification
12.2(8)T	The command was implemented on Cisco 1750 and Cisco 1751.
12.2(13)T3	The **transparent** keyword was added. This keyword is available only in js2 images.
12.4(4)T	The **gsmefr** and **gsmfr** codec keywords were removed.
12.4(15)XY	The **g722-64** keyword was added.

Usage Guidelines

Use this command to define a specific voice coder rate of speech and payload size for a VoIP or VoFR dial peer. This command is also used for VoATM.

A specific codec type can be configured on the dial peer as long as it is supported by the setting used with the **codec complexity** voice-card configuration command. The **codec complexity** command is voice-card specific and platform specific. The **codec complexity** voice-card configuration command is set to either high or medium.

If the **codec complexity** command is set to high, the following keywords are available: **g711alaw, g711ulaw**, **g722-64**, **g723ar53**, **g723ar63**, **g723r53**, **g723r63**, **g726r16**, **g726r24**, **g726r32**, **g728**, **g729r8**, and **g729br8**.

If the **codec complexity** command is set to medium, the following keywords are available: **g711alaw, g711ulaw, g726r16, g726r24, g726r32, g729r8,** and **g729br8**.

The **codec** dial-peer configuration command is particularly useful when you must change to a small-bandwidth codec. Large-bandwidth codecs, such as G.711, do not fit in a small-bandwidth link. However, the g711alaw and g711ulaw codecs provide higher quality voice transmission than other codecs. The g729r8 codec provides near-toll quality with considerable bandwidth savings.

If codec values for the dial peers of a connection do not match, the call fails.

You can change the payload of each VoIP frame by using the **bytes** keyword; you can change the payload of each VoFR frame by using the **bytes** keyword with the *payload-size* argument. However, increasing the payload size can add processing delay for each voice packet.

Table 4 describes the voice payload options and default values for the codecs and packet voice protocols.

Table 4 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g711alaw **g711ulaw**	VoIP VoFR VoATM	80, 160 40 to 240 in multiples of 40 40 to 240 in multiples of 40	160 240 240
g722-64	VoIP	80, 160. 240	160
g723ar53 **g723r53**	VoIP VoFR VoATM	20 to 220 in multiples of 20 20 to 240 in multiples of 20 20 to 240 in multiples of 20	20 20 20
g723ar63 **g723r63**	VoIP VoFR VoATM	24 to 216 in multiples of 24 24 to 240 in multiples of 24 24 to 240 in multiples of 24	24 24 24

Table 4 Voice Payload-per-Frame Options and Defaults (continued)

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g726r16	VoIP VoFR VoATM	20 to 220 in multiples of 20 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60
g726r24	VoIP VoFR VoATM	30 to 210 in multiples of 30 15 to 240 in multiples of 15 30 to 240 in multiples of 15	60 90 90
g726r32	VoIP VoFR VoATM	40 to 200 in multiples of 40 20 to 240 in multiples of 20 40 to 240 in multiples of 20	80 120 120
g728	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60
g729abr8 **g729ar8** **g729br8** **g729r8**	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	20 30 30

For toll quality, use the **g711alaw** or **g711ulaw** keyword. These values provide high-quality voice transmission but use a significant amount of bandwidth. For nearly toll quality (and a significant savings in bandwidth), use the **g729r8** keyword.

Note The **clear-channel** keyword is not supported on Cisco AS5300.

Note The G.723 and G.728 codecs are not supported on the 1700 platform for Cisco Hoot and Holler applications.

Note The **transparent** keyword affects only H.323 to H.323 connections.

Note The G.722-64 codec is only supported for H.323 and SIP.

Examples

The following example shows how to configure a voice coder rate that provides toll quality voice with a payload of 120 bytes per voice frame on a router that is acting as a terminating node. The sample configuration begins in global configuration mode and is for VoFR dial peer 200.

```
dial-peer voice 200 vofr
 codec g711ulaw bytes 240
```

The following example configures a voice coder rate for VoIP dial peer 10 that provides toll quality but uses a relatively high amount of bandwidth:

```
dial-peer voice 10 voip
 codec g711alaw
```

The following example configures the transparent codec used by the Cisco Unified Border Element:

```
dial-peer voice 1 voip
 incoming called-number .T
 destination-pattern .T
 session target ras
 codec transparent
```

Related Commands

Command	Description
codec (DSP interface dsp farm)	Specifies call density and codec complexity.
codec (voice port)	Specifies voice compression.
codec complexity	Specifies call density and codec complexity based on the codec used.
show dial peer voice	Displays the codec setting for dial peers.

codec (DSP Farm profile)

To specify the codecs supported by a digital signal processor (DSP) farm profile, use the **codec** command in DSP farm profile configuration mode. To remove the codec, use the **no** form of this command.

codec {*codec-type* | **pass-through**}

no codec {*codec-type* | **pass-through**}

Syntax Description

codec-type	Specifies the codec preferred. • **g711alaw**—G.711 a–law 64,000 bps. • **g711ulaw**—G.711 u–law 64,000 bps. • **g722r-64**—G.722-64 at 64,000 bps • **g729abr8**—G.729 ANNEX A and B 8000 bps. • **g729br8**—G.729 ANNEX B 8,000 bps. • **g729ar8—**G.729 ANNEX A and R 8000 bps. • **g729r8**—G.729 8000 bps.
pass-through	Enables codec pass-through. Supported for transcoding and MTP profiles.

Command Default

Transcoding

- **g711alaw**
- **g711ulaw**
- **g729abr8**
- **g729ar8**

Conferencing

- **g711alaw**
- **g711ulaw**
- **g729abr8**
- **g729ar8**
- **g729br8**
- **g729r8**

MTP

- **g711ulaw**

Command Modes

DSP farm profile configuration

Command History

Release	Modification
12.3(8)T	This command was introduced.
12.4(4)T	The **pass-through** keyword was added.
12.4(15)XY	The **g722-64** keyword was added.

Usage Guidelines

Only one codec is supported for each Media Termination Point (MTP) profile. To support multiple codecs, you must define a separate MTP profile for each codec.

Hardware MTPs support only G.711 a–law and G.711 u–law. If you configure a profile as a hardware MTP, and you want to change the codec to other than G.711, you must first remove the hardware MTP by using the **no maximum sessions hardware** command.

The **pass-through** keyword is supported for transcoding and MTP profiles only; it is not supported for conferencing profiles. To support the RSVP agent on a SCCP device, you must use the **codec pass-through** command. In pass-through mode, the SCCP device processes the media stream using a pure software MTP regardless of the nature of the stream. This enables video and data streams to be processed in addition to audio. When pass-through mode is set in a transcoding profile, no transcoding is done for the session; the transcoding device performs a pure software MTP function. Pass-through mode can be used for Secure RTP sessions.

Examples

The following example shows the call density and codec complexity being set to GSMEFR:

```
Router(config)# dspfarm profile 123 transcode
Router(config-dspfarm-profile)# codec gsmefr
```

Related Commands

Command	Description
associate application	Associates the SCCP protocol to the DSP farm profile.
dspfarm profile	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
maximum sessions (DSP Farm profile)	Specifies the maximum number of sessions that are supported by the profile.
rsvp	Enables RSVP support on a transcoding or MTP device.
shutdown (DSP Farm profile)	Disables a DSP farm profile.

codec preference

To specify a list of preferred codecs to use on a dial peer, use the **codec preference** command in voice-class configuration mode. To disable this functionality, use the **no** form of this command.

codec preference *value codec-type* [**mode** *frame_size*][**bytes** *payload-size*] [**packetization-period 20**] [**encap rfc3267**] [**frame-format** {**bandwidth-efficient** | **octet-aligned** [**crc** | **no-crc**]}] [**modes** *modes-value*]

no codec preference *value codec-type*

Syntax Description

value	The order of preference, with 1 being the most preferred and 14 being the least preferred.
codec-type	The codec preferred. Values are as follows: • **clear-channel**—Clear Channel 64,000 bps • **g711alaw**—G.711 a law 64,000 bps • **g711ulaw**—G.711 mu-law 64,000 bps • **g722r-64**—G.722-64 at 64,000 bps • **g723ar53**—G.723.1 ANNEX-A 5300 bps • **g723ar63**—G.723.1 ANNEX-A 6300 bps • **g723r53**—G.723.1 5300 bps • **g723r63**—G.723.1 6300 bps • **g726r16**—G.726 16,000 bps • **g726r24**—G.726 24,000 bps • **g726r32**—G.726 32,000 bps • **g728**—G.728 16,000 bps • **g729abr8**—G.729 ANNEX-A and B 8000 bps • **g729br8**—G.729 ANNEX-B 8000 bps • **g729r8**—G.729 8000 bps • **gsmamr-nb**—Enables GSMAMR codec capability • **ilbc**—internet Low Bitrate Codec (iLBC) at 13,330 bps or 15,200 bps. • **transparent**—Enables codec capabilities to be passed transparently between endpoints **Note** The **transparent** keyword not supported when the **call-start** command is configured.
mode	(Optional) For iLBC codecs only. Specifies the iLBC operating frame mode that is encapsulated in each packet.

frame_size	(Optional) For iLBC codecs only. iLBC operating frame in milliseconds (ms). Valid entries are: • 20—20ms frames for 15.2kbps bit rate • 30—30ms frames for 13.33 kbps bit rate Default is 20.
bytes	(Optional) Specifies that the size of the voice frame is in bytes.
payload-size	(Optional) Number of bytes you specify as the voice payload of each frame. Values depend on the codec type and the packet voice protocol.
packetization-period 20	(Optional) Sets the packetization period at 20 ms. Applicable only to GSMAMR-NB codec support.
encap rfc3267	(Optional) Sets the encapsulation value to comply with RFC 3267. Applicable only to GSMAMR-NB codec support.
frame-format	(Optional) Specifies a frame format. Supported values are **octet-aligned** and **bandwidth-efficient**. The default is **octet-aligned**. Applicable only to GSMAMR-NB codec support.
crc | **no-crc**	(Optional) CRC is applicable only for **octet-aligned** frame format. If you enter **bandwidth-efficient** frame format, the **crc** | **no-crc** options will not be available because they are inapplicable. Applicable only to GSMAMR-NB codec support.
modes *modes-values*	(Optional) Valid values are from 0 to 7. You can specify modes as a range (for example, 0-2), or individual modes separated by commas (for example, 2,4,6), or a combination of the two (for example, 0-2,4,6-7). Applicable only to GSMAMR-NB codec support.

Command Default

If the **gsmamr-nb** keyword is entered, the default values are as follows:

Packetization period is **20** ms.
Encap is **rfc3267**.
Frame format is **octet-aligned**.
CRC is **no-crc**.
Modes value is **0-7**.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced on the Cisco AS5300.
12.0(7)T	This command was implemented on the Cisco 2600 series and Cisco 3600 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco Release IOS Release 12.1(2)T.
12.1(5)T	The codecs **gsmefr** and **gsmfr** were added.
12.2(13)T3	The **transparent** keyword was added.
12.4(4)T	The codecs **gsmefr** and **gsmfr** were removed.

Release	Modification
12.4(4)XC	This command was extended to include GSMAMR-NB codec parameters on the Cisco AS5350XM and Cisco AS5400XM platforms.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)T	The **ilbc** codec and **mode** keyword were added.
12.4(15)XY	The **g722r-64** keyword was added.

Usage Guidelines

The routers at opposite ends of the WAN may have to negotiate the codec selection for the network dial peers. The **codec preference** command specifies the order of preference for selecting a negotiated codec for the connection. Table 5 describes the voice payload options and default values for the codecs and packet voice protocols.

Note The **transparent** keyword not supported when the **call start** command is configured.

Table 5 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g711alaw **g711ulaw**	VoIP VoFR VoATM	80, 160 40 to 240 in multiples of 40 40 to 240 in multiples of 40	160 240 240
g722r-64	VoIP	80, 160, 240	160
g723ar53 **g723r53**	VoIP VoFR VoATM	20 to 220 in multiples of 20 20 to 240 in multiples of 20 20 to 240 in multiples of 20	20 20 20
g723ar63 **g723r63**	VoIP VoFR VoATM	24 to 216 in multiples of 24 24 to 240 in multiples of 24 24 to 240 in multiples of 24	24 24 24
g726r16	VoIP VoFR VoATM	20 to 220 in multiples of 20 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60
g726r24	VoIP VoFR VoATM	30 to 210 in multiples of 30 15 to 240 in multiples of 15 30 to 240 in multiples of 15	60 90 90
g726r32	VoIP VoFR VoATM	40 to 200 in multiples of 40 20 to 240 in multiples of 20 40 to 240 in multiples of 20	80 120 120
g728	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	40 60 60

Table 5 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (in Bytes)	Default Voice Payload (in Bytes)
g729abr8 **g729ar8** **g729br8** **g729r8**	VoIP VoFR VoATM	10 to 230 in multiples of 10 10 to 240 in multiples of 10 10 to 240 in multiples of 10	20 30 30
ilbc	VoIP	For **mode 20**, 38, 76, 114, 152, 190, 228. For **mode 30**, 50, 100, 150, 200	38 50

Examples

The following example sets the codec preference to the GSMAMR-NB codec and specifies parameters:

```
Router(config-voice-class)# codec preference 1 gsmamr-nb packetization-period 20 encap
rfc3267 frame-format octet-aligned crc
```

The following example creates codec preference list 99 and applies it to dial peer 1919:

```
voice class codec 99
codec preference 1 g711alaw
codec preference 2 g711ulaw bytes 80
codec preference 3 g723ar53
codec preference 4 g723ar63 bytes 144
codec preference 5 g723r53
codec preference 6 g723r63 bytes 120
codec preference 7 g726r16
codec preference 8 g726r24
codec preference 9 g726r32 bytes 80
codec preference 10 g729br8
codec preference 11 g729r8 bytes 50
codec preference 12 gsmefr
end
dial-peer voice 1919 voip
 voice-class codec 99
```

The following example configures the transparent codec used by the Cisco Unified Border Element:

```
voice class codec 99
codec preference 1 transparent

codec preference 1 transparent
```

Note You can only assign a preference value of 1 to the transparent codec. Additional codecs assigned to other preference values are ignored if the transparent codec is used.

The following example shows how to configure the iLBC codec used by the Cisco Unified Border Element:

```
voice class codec 99
codec preference 1 ilbc 30 200
```

Related Commands

Command	Description
call-start	Forces an H.323 Version 2 gateway to use fast connect or slow connect procedures for a dial peer.
voice class codec	Enters voice-class configuration mode and assigns an identification tag number to a codec voice class.
voice-class codec (dial peer)	Assigns a previously configured codec selection preference list to a dial peer.

Feature Information for G.722-64 and iLBC Codec Support on Cisco UBEs, DSP Farms, and Voice Gateways

Table 6 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Cisco IOS software images are specific to a Cisco IOS software release, a feature set, and a platform. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Note Table 6 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 6 Feature Information for G.722-64 and iLBC Codec Support on Cisco UBEs, DSP Farms, and Voice Gateways

Feature Name	Releases	Feature Information
iLBC Codec Support	12.4(11)T	The internet Low Bitrate Codec (iLBC) is a standard, high-complexity speech codec that is suitable for robust voice communication over IP. iLBC has built-in error correction functionality that helps the codec perform in networks with a high-packet loss.
G.722-64 and iLBC Codec Support on Cisco UBEs, DSP Farms, and Voice Gateways	12.4(15)XY	The G.722-64 and iLBC codecs are supported for Cisco UBEs, DSP farms, and voice gateways. Conferencing and universal transcoding are supported on both codecs.