



Global System for Mobile Communications Full Rate and Enhanced Full Rate Codecs

This feature module describes the Global System for Mobile Communications (GSM) Full Rate (FR) and Enhanced Full Rate (EFR) Codecs feature. It includes information on the following topics:

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Feature Overview

This feature supports Cisco Mobile Office Network (MNET) GSM mobile telephony products and solutions that leverage the IP functionality of the Cisco network and its voice gateways, thereby enhancing the effectiveness of individuals in an enterprise environment. The feature includes GSM Full Rate (FR) and Enhanced Full Rate (EFR) Codecs in the Digital Signal Processor (DSP) firmware of the voice gateway, and supplementary services such as Blind Call Transfer.

Call Transfer allows an H.323 endpoint to redirect an answered call to another H.323 endpoint. Cisco gateways support H.450.2 Call Transfer as the transferred and transferred-to party. The transferring endpoint must be an H.450-capable terminal; the Cisco gateway cannot act as the transferring endpoint. Gatekeeper-controlled or Gatekeeper-initiated Call Transfer is not supported.

Benefits

Wireless Mobile Office System

The Cisco voice gateway supports the Cisco Mobile Office Network (MNET) solution.

Restrictions

- Call Manager and IP phones are not integrated into the MNET solution. The endpoints that can interwork with the user are internal and external interfaces connected through an H.323 gateway, such as PBX users, FXS and FXO analog interfaces, and T1 CAS and T1-PRI digital interfaces.
- For Call Transfer, only Blind Transfer is supported.
- Call Diversion according to H.450.3 is not supported.
- GSM codec is converted to Pulse Code Modulation (PCM) via the voice gateway. Transcoding of GSM to another code type is not supported.
- The Cisco GSM Mobility Controller provides centralized dialing plan management and routing, but does not provide RAS (registration, admission, and status), according to H.323.
- The GSMEFR and GSMFR codecs are not supported for VoFR and VoATM.

Related Documents

- *Cisco AS5300 Voice-over-IP Feature Card Installation and Configuration*

Supported Platforms

- Cisco VG200
- Cisco 2600 series
- Cisco 3600 series
- Cisco AS5300
- Cisco 7200 series
- Cisco 7500 series

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

No new or modified MIBs are supported by this feature.

To obtain lists of MIBs supported by platform and Cisco IOS release and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (CCO) at <http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>.

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

Please refer to *Cisco AS5300 Voice-over-IP Feature Card Installation and Configuration* before you configure your Cisco AS5300 to use Voice over IP.

Configuration Tasks

See the following sections for configuration tasks for the Cisco GSM FR and EFR Codecs feature. Each task in the list indicates if the task is optional or required.

- Configuring Dial Peers (Required)

Configuring Dial Peers

The H.323 gateway must be configured to interwork with the Cisco GSM Mobility Controller as a peer-to-peer H.323 entity and must also be configured to be H.450-capable.

	Command	Purpose
Step 1	Router(config)# dial-peer voice 5000 voip	Enters dial-peer configuration mode and define a local dial peer that will connect to the Voice over IP (VoIP) network. The number (in this case, 5000) is one or more digits identifying the dial peer. Valid entries are from 1 to 2,147,483,647.
Step 2	Router(config-dial-peer)# application session	Enables H.450 features.
Step 3	Router(config-dial-peer)# destination-pattern 5...	Routes all 5... numbers to the session target in Step 4.
Step 4	Router(config-dial-peer)# session target ipv4:172.20.71.26	Identifies the session target (the IP address of the Cisco GSM Mobility Controller (GMC)).
Step 5	Router(config-dial-peer)# codec gsmfr	Specifies the voice coder rate of speech for the dial peer to be 13,200 bps.
Step 6	Router(config-dial-peer)# exit	Exits from interface configuration mode.
Step 7	Router(config)# dial-peer voice 408 voip	Defines a local dial peer that will connect to the VoIP network.
Step 8	Router(config-dial-peer)# application session	Enables H.450 features.
Step 9	Router(config-dial-peer)# incoming called-number 408666	Associates incoming calls to 408666 numbers.
Step 10	Router(config-dial-peer)# codec gsmfr	Specifies the voice coder rate of speech for the dial peer to be 13,200 bps.
Step 11	Router(config-dial-peer)# exit	Exits from interface configuration mode.
Step 12	Router(config)# dial-peer voice 408666 pots	Defines a local dial peer that will connect to the VoIP network.
Step 13	Router(config-dial-peer)# application session	Enables H.450 features.

	Command	Purpose
Step 14	Router(config-dial-peer)# destination-pattern 4086668888	Identifies the telephone number associated with this dial peer.
Step 15	Router(config-dial-peer)# port 1/0/0	Associates this dial peer with a specific logical dial interface.

Verifying Gateway Configuration

Step 1 Enter the **show dial-peer voice** command to display codec information:

```

Router# show dial-peer voice 555
VoiceOverIpPeer555
  information type = voice,
  tag = 555, destination-pattern = `',
  answer-address = `', preference=0,
  numbering Type = `unknown'
  group = 555, Admin state is up, Operation state is up,
  incoming called-number = `4085264320', connections/maximum =
0/unlimited,
  DTMF Relay = disabled,
  modem passthrough = system,
  huntstop = disabled,
  in bound application associated:DEFAULT
  out bound application associated:
  permission :both
  incoming COR list:maximum capability
  outgoing COR list:minimum requirement
  type = voip, session-target = `',
  technology prefix:
  settle-call = disabled
  ip precedence = 0, UDP checksum = disabled,
  session-protocol = cisco, session-transport = udp, req-qos =
best-effort,
  acc-qos = best-effort,
  fax rate = voice, payload size = 20 bytes
  fax protocol = system
  fax NSF = 0xAD0051 (default)
→  codec = gsmefr, payload size = 32 bytes,
  codec display
  Expect factor = 0, Icpif = 20,
  Playout:Mode adaptive,
  Expect factor = 0,
  Max Redirects = 1, Icpif = 20,signaling-type = cas,
  CLID Restrict = disabled
  VAD = enabled, Poor QOV Trap = disabled,
  voice class perm tag = ` '
  Connect Time = 0, Charged Units = 0,
  Successful Calls = 0, Failed Calls = 0,
  Accepted Calls = 0, Refused Calls = 0,
  Last Disconnect Cause is "",
  Last Disconnect Text is "",
  Last Setup Time = 0.
Router#

```

Step 2 Enter the **show running-configuration** command to view **voice class codec** information.

```
Router# show running-config
Building configuration...

Current configuration:
!
version 12.1
. . .
!
!
voice class codec 99
  codec preference 1 g711alaw
  codec preference 2 g723ar53
  codec preference 3 g723r53
  codec preference 4 g726r16
  codec preference 5 g726r24
  codec preference 6 g728
  codec preference 7 g729br8
  codec preference 8 gsmefr
  codec preference 9 gsmfr
!
!
....
```

Configuration Examples

This section provides the following configuration examples:

- Frame Relay for Voice over IP

Frame Relay for Voice over IP

For Frame Relay, it is customary to configure a main interface and several subinterfaces, one subinterface per permanent virtual connection (PVC). The following example configures a Frame Relay main interface and a subinterface so that voice and data traffic can be successfully transported:

```
interface Serial0/0
ip mtu 300
no ip address
encapsulation frame-relay
no ip route-cache
no ip mroute-cache
fair-queue 64 256 1000
frame-relay ip rtp header-compression
interface Serial0/0.1 point-to-point
ip mtu 300
ip address 40.0.0.7 255.0.0.0
ip rsvp bandwidth 48 48
no ip route-cache
no ip mroute-cache
bandwidth 64
traffic-shape rate 32000 4000 4000
frame-relay interface-dlci 16
frame-relay ip rtp header-compression
```

In this configuration example, the main interface has been configured as follows:

- MTU size of IP packets is 300 bytes.
- No IP address is associated with this serial interface. The IP address must be assigned for the subinterface.
- Encapsulation method is Frame Relay.
- Fair queueing is enabled.
- IP Real-Time Transport protocol (RTP) header compression is enabled.

The subinterface has been configured as follows:

- MTU size is inherited from the main interface.
- IP address for the subinterface is specified.
- Bandwidth is set to 64 kbps.
- Generic traffic shaping is enabled with 32-kbps CIR, where Bc = 4000 bits and Be = 4000 bits.
- Frame Relay data-link connection identifier (DLCI) number is specified.
- IP RTP header compression is enabled.

**Note**

When traffic bursts over the committed information rate (CIR), output rate is held at the speed configured for the CIR (for example, traffic will not go beyond 32 kbps if CIR is set to 32 kbps).

For more information about Frame Relay, refer to the Cisco IOS Release 12.1 Wide-Area Networking Configuration Guide.

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications.

- **codec (dial-peer)**
- **codec preference**

codec (dial-peer)

To specify the voice coder rate of speech for a Voice over IP (VoIP), Voice over ATM (VoATM), or Voice over Frame Relay (VoFR) dial peer, use the **codec** dial-peer configuration command. To reset to the default value, use the **no** form of this command.

```
codec { g711alaw | g711ulaw | g723ar53 | g723ar63 | g723r53 | g723r63 | g726r16 | g726r24 |
g726r32 | g728 | g729abr8 | g729ar8 | g729br8 | g729r8 | gsmefr | gsmfr }
[ bytes payload_size ]
```

```
no codec { g711alaw | g711ulaw | g723ar53 | g723ar63 | g723r53 | g723r63 | g726r16 |
g726r24 | g726r32 | g728 | g729abr8 | g729ar8 | g729br8 | g729r8 | gsmefr | gsmfr }
```

Syntax Description		
clear-channel		Clear Channel at 64,000 bits per second (bps).
g711alaw		G.711 A-Law at 64,000 bps.
g711ulaw		G.711 u-Law at 64,000 bps.
g723ar53		G.723.1 ANNEX A at 5,300 bps.
g723ar63		G.723.1 at 6,300 bps.
g726r53		G.723.1 at 5,300 bps.
g726r63		G.723.1 at 6,300 bps.
g726r16		G.726 at 16,000 bps.
g726r24		G.726 at 24,000 bps.
g726r32		G.726 at 32,000 bps.
g728		G.728 at 16,000 bps.
g729abr8		G.729 ANNEX A & B at 8,000 bps.
g729ar8		G.729 ANNEX A at 8,000 bps.
g729br8		G.729 ANNEX B at 8,000 bps.
g729r8		G.729 at 8,000 bps. This is the default codec.
gsmefr		GSMEFR at 12,200 bps.
gsmfr		GSMFR at 13,200 bps.
bytes		(Optional) Used to specify the number of bytes in the voice payload of each frame.
<i>payload_size</i>		The number of bytes in the voice payload of each frame. Enter a ? character after the keyword bytes to get a list of valid payload values for your specific dial peer.

Defaults

g729r8, 30-byte payload for VoFR and VoATM
g729r8, 20-byte payload for VoIP

Command Modes Dial-peer configuration

Command History

Cisco IOS Release	Modification
11.3(1)T	This command was introduced.
11.3(3)T	Support for Cisco 2600 series routers was added.
12.0(3)T	Support for the Cisco AS5300 universal access server was added.
12.0(4)T	Support for the Cisco 2600, Cisco 3600, and Cisco 7200 series routers and the Cisco MC3810 were added.
12.0(7)T	Support for the Cisco AS5800 universal access server was added. Additional voice coder rates of speech were added.
12.1(5)T	Additional voice coder rates of speech were added.

Usage Guidelines

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

VoFR and VoATM do not support the **gsmefr** and **gsmfr** codecs.

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 Cisco 2600 series acting as a terminating node. The example configuration, starting from global configuration mode, is for VoIP dial peer 200:

```
Router(config)# dial-peer voice 200 voip
Router(config-dial-peer)# codec gsmfr
Router(config-dial-peer)#
```

codec preference

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

codec preference *value codec_type* [**bytes** *size*]

no codec preference *value codec_type*

Syntax Description	
<i>value</i>	Specifies the order of preference with 1 being the most preferred and 12 being the least preferred.
<i>codec_type</i>	Specifies the type of codec preferred. <ul style="list-style-type: none"> • clear-channel — Clear Channel 64,000 bps • g711alaw — G.711 A Law 64,000 bps • g711ulaw — G.711 u Law 64,000 bps • g723ar53 — G.723.1 ANNEX-A 5,300 bps • g723ar63 — G.723.1 ANNEX-A 6,300 bps • g723r53 — G.723.1 5,300 bps • g723r63 — G.723.1 6,300 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 & B 8,000 bps • g729br8 — G.729 ANNEX-B 8,000 bps • g729r8 — G.729 8000 bps • gsmefr — Global System for Mobile Communications Enhanced Full Rate (GSMEFR) 12,200 bps • gsmfr — Global System for Mobile Communications (GSM) Full Rate (GSMFR) 13,200 bps
bytes	(Optional) Specifies that the size of the voice frame is in bytes.
<i>size</i>	Number of voice data bytes per frame. Valid sizes vary by codec.

Defaults No default behavior or values.

Usage Guidelines Cisco gateways do not support a codec preference order when using H.323 signaling. All codecs listed are given equal preference.

Command Modes Class configuration

Command History

Release	Modification
12.0(2)XH	This command was introduced.
12.1(5)T	The codecs gsmeifr and gsmfr were added.

Examples

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

```

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

```

Glossary

CAS—channel associated signaling.

CIR—committed information rate. Rate at which a Frame Relay network agrees to transfer information under normal conditions, averaged over a minimum increment of time. CIR, measured in bits per second, is one of the key negotiated tariff metrics.

DLCI—data-link connection identifier. Value that specifies a PVC or SVC in a Frame Relay network. In the basic Frame Relay specification, DLCIs are locally significant (connected devices might use different values to specify the same connection). In the LMI extended specification, DLCIs are globally significant (DLCIs specify individual end devices).

DSP—Digital Signal Processor.

EFR—Enhanced Full Rate. GSM codec.

FR—Full Rate. GSM codec.

GMC—Cisco GSM Mobility Controller

GSM—Global System for Mobile Communication.

GW—gateway.

ITU—International Telecommunication Union.

RTP—Real-Time Transport protocol. An IETF standard protocol. The H.225.0 standard describes how to use RTP to handle the packetization of video and audio in H.323.

VIC—Voice Interface Card.

