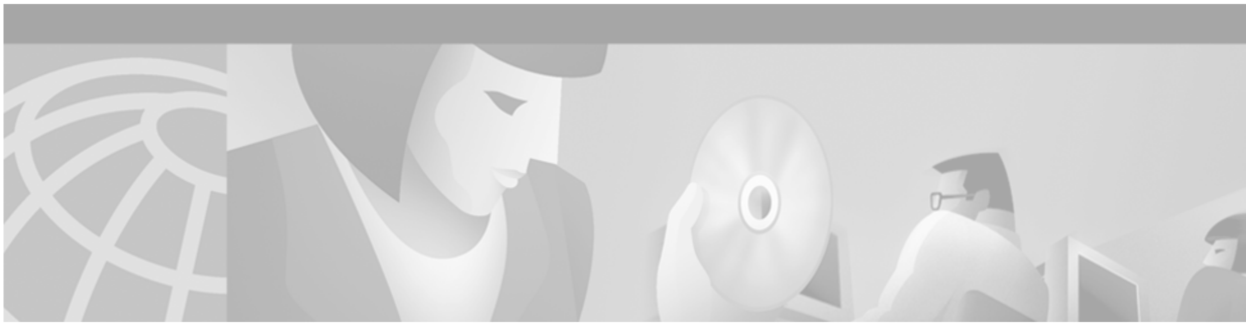

Global System for Mobile Communications Full Rate and Enhanced Full Rate Codecs

This appendix describes the global system for mobile communications (GSM) full rate (FR) and enhanced full rate (EFR) codecs feature. The appendix includes the following sections:

For a complete description of the commands used to configure VoIP for modem support, refer to the *Cisco IOS Voice, Video, and Fax Command Reference*. To locate documentation of other commands that appear in this chapter, use the command reference master index or search online.

To identify the hardware platform or software image information associated with a feature in this chapter, use the Feature Navigator on Cisco.com to search for information about the feature or refer to the software release notes for a specific release. For more information, see the "Identifying Supported Platforms" section in the "Using Cisco IOS Software" chapter.

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

The global system for mobile communications full rate and enhanced full rate codecs supports Cisco mobile office network (MNET) GSM mobile telephony products and solutions. By leveraging the IP functionality of the Cisco network and its voice gateways, these products and solutions enhance the effectiveness of individuals in an enterprise environment. The feature includes GSM full rate and enhanced full rate 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.

The global system for mobile communications full rate and enhanced full rate codecs is supported on the following platforms:

- Cisco VG200
- Cisco 2600, 3600, 7200, and 7500 series routers
- Cisco AS5300 universal access server

The Cisco voice gateway supports the Cisco MNET solution.

Prerequisite Tasks and Restrictions

Before configuring your Cisco AS5300 to use Voice over IP (VoIP), refer to *Cisco AS5300 Voice-over-IP Feature Card Installation and Configuration*.

The following restrictions apply to the global system for mobile communications full rate and enhanced full rate codecs:

- 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, Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) analog interfaces, and T1 channel-associated signaling (CAS) and T1 primary rate interface (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 standards.

GSM Configuration Tasks

See the following section to configure the Cisco global system for mobile communications full rate and enhanced full rate codecs feature. The "Configuring Dial Peers" configuration task is 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. To configure dial peers, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`tag`* **`voip`**	Enters dial-peer configuration mode and defines a local dial peer that will connect to the Voice over IP (VoIP) network. The *tag* argument 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.

<table>
<tr><th></th><th>Command</th><th>Purpose</th></tr>
<tr><td>Step 3</td><td><code>Router(config-dial-peer)# destination-pattern [+] string [T]</code></td><td>
Specifies the E.164 address associated with this dial peer.
The keywords and arguments are as follows:

+—(Optional) Specifies a character indicating an E.164 standard number. The plus sign (+) is not supported on the Cisco MC3810.
string—Indicates a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters:

The asterisk (*) and pound sign (#)—Indicate the keys that appear on standard touch-tone dial pads.
Comma (,)—Inserts a pause between digits.
Period (.)—Matches any entered digit (this character is used as a wildcard).
Percent sign (%)—Indicates that the previous digit/pattern occurred zero or multiple times, similar to the wild card usage in the regular expression.
Plus sign (+)—Matches a sequence of one or more matches of the character/pattern.

Note The plus sign used as part of the digit string is different from the plus sign that can be used in front of the digit string to indicate that the string is an E.164 standard number.

Circumflex (^)—Indicates a match to the beginning of the string.
Dollar sign ($)—Matches the null string at the end of the input string.
Backslash symbol (\)—Is followed by a single character matching that character or used with a single character having no other significance (matching that character).
Question mark (?)— Indicates that the previous digit occurred zero or one time.
Brackets ([])—Indicates a range of digits. A range is a sequence of characters enclosed in the brackets, and only numeric characters from “0” to “9” are allowed in the range. This is similar to a regular expression rule.

</td></tr>
</table>

	Command	Purpose
		– Parentheses ( )—Indicates a pattern and is the same as the regular expression rule—for example, 408(555). Parentheses are used in conjunction with symbols ?, %, or +. For more information on applying wildcard symbols to destination patterns and the dial strings that result, see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide. • **T**—(Optional) Control character indicating that the **destination-pattern** value is a variable length dial string.
Step 4	`Router(config-dial-peer)#` **`session target`**	Identifies the session target (the IP address of the Cisco GSM mobility controller (GMC).
Step 5	`Router(config-dial-peer)#` **`codec`**	Specifies the voice coder rate of speech for the dial peer. (In this case, the **gsmr** keyword will be used to indicate that it is 13,200 bps.)
Step 6	`Router(config-dial-peer)#` **`exit`**	Exits dial-peer configuration mode.
Step 7	`Router(config)#` **`dial-peer voice`** *`tag`* **`voip`**	Enters dial-peer configuration mode and defines a local dial peer that will connect to the VoIP network. (This dial peer is different from the one in Step 1.)
Step 8	`Router(config-dial-peer)#` **`application session`**	Enables H.450 features.
Step 9	`Router(config-dial-peer)#` **`incoming called-number`** *`string`*	Specifies the incoming called number of a Multimedia Mail over Internet Protocol (MMoIP) or Plain Old Telephone Service (POTS) dial peer. The *string* argument specifies the incoming called telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
Step 10	`Router(config-dial-peer)#` **`codec`**	Specifies the voice coder rate of speech for the dial peer. (In this case, the **gsmr** keyword will be used to indicate that it is 13,200 bps.)
Step 11	`Router(config-dial-peer)#` **`exit`**	Exits dial-peer configuration mode.
Step 12	`Router(config)#` **`dial-peer voice`** *`tag`* **`pots`**	Enters dial-peer configuration mode and configures a POTS dial peer. The *tag* value of the dial-peer voice POTS command uniquely identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 13	`Router(config-dial-peer)#` **`application session`**	Enables H.450 features.
Step 14	`Router(config-dial-peer)#` **`destination-pattern`** `[+]` *`string`* `[`**`T`**`]`	Identifies the telephone number associated with this dial peer. For an explanation of the keywords and arguments, see Step 3 in this configuration task table.
Step 15	`Router(config-dial-peer)#` **`port`** *`slot-number/subunit-number/port`*	Associates this dial peer with a specific logical dial interface.

Verifying Gateway Configuration

To confirm the gateway configuration, perform the following steps:

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.
```

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

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

Current configuration:
!
version 12.2
.
.
.
!
```

```
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
!
.
.
.
```

GSM Configuration Example

This section provides a Frame Relay for voice over IP configuration example.

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:

- Maximum transmission unit (MTU) size of IP packets is 300 bytes.
- An IP address is not 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 committed information rate (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 CIR, the output rate is held at the speed configured for the CIR (for example, traffic will not go beyond 32 kbps if the CIR is set to 32 kbps).

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