Configuring Cisco MC3810 Series Concentrators to Use High-Performance Compression Modules

This document describes feature changes introduced in Cisco IOS Release 12.0(7)XK for Cisco MC3810 series concentrators that contain high-performance voice compression modules (HCMs).

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Feature Overview

High-performance voice compression modules (HCMs) provide voice compression according to the voice compression coding algorithm (codec) specified when the Cisco MC3810 is configured. Table 1 shows the number of voice channels each type of compression module can support.

Table 1 Voice Compression Module Application Information

Type	Codec Packaging Complexity (see Table 2)	Voice Channels per HCM
HCM2	High complexity (**codec complexity** set to **high**)	4
	Medium complexity (**codec complexity** set to **medium**)	8
HCM6	High complexity (**codec complexity** set to **high**)	12
	Medium complexity (**codec complexity**) set to **medium**:	24

Table 2 Codec Packaging Information

Codec Packaging	Codecs
High complexity	G.711ulaw, G.711alaw G.723.1(r5.3), G.723.1 Annex A(r5.3), G.723.1(r6.3), G.723.1 Annex A(r6.3) G.726(r16), G.726(r24), G.726(r32) G.728 G.729, G.729 Annex B fax relay
Medium complexity	G.711ulaw, G.711alaw G.726(r16), G.726(r24), G.726(r32) G.729 Annex A, G.729 Annex B with Annex A fax relay

Benefits

HCMs support more voice channels than the earlier voice compression modules (VCMs) used in the Cisco MC3810.

Restrictions

The following restrictions apply to HCMs:

- HCMs should not be combined with VCMs in a Cisco MC3810 chassis.
- HCMs require Cisco IOS Release 12.0(7)XK or later.

Related Features and Technologies

Voice over Internet Protocol (VoIP)

Voice over Frame Relay (VoFR)

Voice over Asynchronous Transfer Mode (VoATM)

Supported Platforms

This feature is supported on the following platforms:

- Cisco MC3810 series

Supported Standards, MIBs, and RFCs

RFCs

- RFC 1890— RTP: A Transport Protocol for Real-Time Applications
- RFC 1889—RTP Profile for Audio and Video Conferences with Minimal Control

MIBs

- CISCO-ENTITY-VENDORTYPE-OID-MIB
- OLD-CISCO-CHASSIS-MIB

- CAS_INTF_MIB

International Telecommunication Union (ITU-T) G-Series Codec Compression Specifications

— G.711 A Law at 64,000 bps

— G.711 u Law at 64,000 bps

— G.723.1 Annex A at 5300 bps

— G.723.1 Annex A at 6300 bps

— G.723.1 at 5300 bps

— G.723.1 at 6300 bps

— G.726 at 16,000 bps

— G.726 at 24,000 bps

— G.726 at 32,000 bps

— G.728 at 16,000 bps

— G.729 at 8000 bps

— G.729 Annex A and B at 8000 bps

— G.729 Annex A at 8000 bps

— G.729 Annex B at 8000 bps

Prerequisites

One or two HCM modules must be installed in your Cisco MC3810.

Note An HCM may not be combined with a VCM in one chassis.

Configuration Tasks

Complete the following tasks to configure the Cisco MC3810 voice ports for operation with one or two HCMs installed:

- Configuring Codec Complexity
- Specifying Codecs for Network Dial Peers

Configuring Codec Complexity

To configure codec complexity for voice ports, enter the following commands, beginning in privileged EXEC mode. Commands apply to both analog and digital voice ports unless otherwise indicated. You enter the codec complexity command in voice-card configuration mode. On the Cisco MC3810, voice-card 0 is used as a virtual voice-card, and the setting applies to all voice ports on this Cisco MC3810.

This procedure does not cover other voice-port configuration commands that may be required. To learn more, see *Voice, Video, and Home Applications Configuration Guide* and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0, and the *Cisco MC3810 Multiservice Concentrator Software Configuration Guide*.

Step	Command	Purpose
1	`router#` **`show voice dsp`**	Check the DSP voice channel activity. If any DSP voice channels are in the busy state, you cannot change the codec complexity. When all of the DSP channels are in the idle state, continue to step 2.
2	`router#` **`configure terminal`**	Enter global configuration mode.
3	`router(config)#` **`voice-card 0`**	Enter voice-card configuration mode and specify voice card 0. Voice card 0 provides the configuration mode for setting the codec complexity on a Cisco MC3810.
4	`router(config-voicecard)#` **`codec complexity {high | medium}`**	Specify the codec complexity for this Cisco MC3810 according to the bandwidth requirements and the number of voice channels to be supported per DSP. The default is medium complexity, which provides four voice channels per DSP. See the "codec complexity" section on page 11 in the Command Reference for information about the **codec complexity** command.
5	`router(config-voicecard)#` **`exit`**	Exit from voice-card configuration mode.

Verifying Codec Complexity Settings

To verify the codec complexity configuration, enter the **show running-config** command to display the current voice-card setting. If medium complexity is specified, the codec complexity setting is not displayed. If high complexity is specified, the setting `codec complexity high` is displayed. The following example shows an excerpt from the command output if high complexity has been specified:

```
Router# show running-config
.
.
.
hostname router-alpha

voice-card 0
 codec complexity high
.
.
```

Specifying Codecs for Network Dial Peers

Follow these steps to specify a codec for each network dial peer according to the codec complexity setting selected for this Cisco MC3810. If you do not set codec complexity and specify a codec, the defaults remain in effect: medium complexity and G.729, 8000 bps.

This does not cover the complete dial-peer configuration procedure. To learn more, see *Voice, Video, and Home Applications Configuration Guide* and *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0

Step	Command	Purpose
1	`Router#` **`configure terminal`**	Enter global configuration mode.
2	`Router(config)#` **`dial-peer voice`** *`tag`* {**`voip`** | **`vofr`** | **`voatm`**}	Enter dial-peer configuration mode and specify a network dial peer for which to specify a codec. *tag* is one or more digits to identify the dial peer. Valid entries are 1 to 2147483647.
3	`Router(config-dial-peer)#` **`codec`** *`codec`* [**`bytes`** *`payload-size`*]	Specify a codec for the dial peer. Codec types are as follows: **g711alaw**, **g711ulaw**, **g723ar53**, **g723ar63**, **g723r53**, **g723r63**, **g726r16**, **g726r24**, **g726r32**, **g728**, **g729abr8**, **g729ar8**, **g729br8**, **g729r8** The default is **g729r8**. Optionally specify the voice payload (in bytes) of each frame. For information about the options available for the **codec** command, see the command reference section or enter **?**.
4	`Router(config-dial-peer)#` **`exit`**	Exit from dial-peer configuration mode.

Verifying Network Dial Peers

Follow the procedure below to verify dial-peer configuration. To learn more about these commands, see *Voice, Video, and Home Applications Command Reference* for Cisco IOS Release 12.0.

Enter the privileged EXEC **show dial-peer voice** command. The following text is sample output from the **show dial-peer voice** command for a VoIP dial peer:

```
Router# show dial-peer voice 10
VoiceOverIpPeer10
        information type = voice,
        tag = 10, destination-pattern = `555....',
        answer-address = `', preference=0,
        group = 10, Admin state is up, Operation state is up,
        incoming called-number = `', connections/maximum = 0/unlimited,
        DTMF Relay = disabled,
        application associated:
        type = voip, session-target = `ipv4:10.1.1.1',
        technology prefix:
        ip precedence = 0, UDP checksum = disabled,
        session-protocol = cisco, req-qos = best-effort,
        acc-qos = best-effort,
        fax-rate = voice,   payload size =  20 bytes
        codec = g711alaw,   payload size =  160 bytes,
        Expect factor = 10, Icpif = 30,signaling-type = cas,
        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.
```

The following text is sample output from the **show dial-peer voice** command for a VoFR dial peer:

```
Router# show dial-peer voice 20
VoiceOverFRPeer20
        information type = voice,
        tag = 20, destination-pattern = `555....',
        answer-address = `', preference=0,
        group = 20, Admin state is up, Operation state is up,
        incoming called-number = `', connections/maximum = 0/unlimited,
        DTMF Relay = disabled,
        application associated:
        type = vofr, session-target = `Serial0 120',
        called number = `',
        session-protocol = cisco-switched,
        fax-rate = voice,   payload size =  30 bytes
        codec = g729r8,   payload size =  30 bytes,
        signaling-type = cas,
        VAD = enabled,
        Voice Sequence Numbers = 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.
```

The following text is sample output from the **show dial-peer voice** command for a VoATM dial peer:

```
Router# show dial-peer voice 1
VoiceOverATMPeer1
        information type = voice,
        tag = 1, destination-pattern = `555....',
        answer-address = `', preference=5,
        group = 1, Admin state is up, Operation state is up,
        incoming called-number = `', connections/maximum = 0/unlimited,
        DTMF Relay = disabled,
        application associated:
        type = voatm, session-target = `ATM0 pvc 101/1001',
        session-protocol = cisco-switched,
        fax-rate = voice,   payload size =  30 bytes
        codec = g729r8,   payload size =  30 bytes,
        signaling-type = cas,
        VAD = enabled,
        Voice Sequence Numbers = 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.
```

Command Reference

This section documents new or modified commands. Modified commands are indicated by an asterisk (*). All other commands used with this feature are documented in the Cisco IOS Release 12.0 command references.

- **codec (dial-peer)***
- **codec complexity***
- **ds0-group***
- **voice-card***

codec (dial-peer)

To specify the voice codec for a network dial peer, enter the **codec** dial-peer configuration command. Use the **no** form of this command to restore the default value.

codec *codec* [**bytes** *payload-size*]
no codec

Syntax Description

codec	Codec options on Cisco MC3810 series equipped with HCM, and with codec complexity set to **high** or **medium**: • **g711alaw**—G.711 A Law, 64000 bps • **g711ulaw**—G.711 u Law, 64000 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, 16000 bps • **g726r24**—G.726, 24000 bps • **g726r32**—G.726, 32000 bps • **g728**—G.728, 16000 bps • **g729abr8**—G.729 Annex A and Annex B, 8000 bps • **g729ar8**—G.729 Annex A, 8000 bps • **g729br8**—G.729 Annex B, 8000 bps • **g729r8**—G.729, 8000 bps
bytes	(Optional) The voice payload for each frame.
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. See Table 3 for valid entries and default values.

Default

If no codec is specified, dial peers are configured for **g729r8** and the voice payload is as shown in Table 3 for G.729r8.

If a codec is specified without the **bytes** keyword, the voice payload is as shown in Table 3.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced as a Cisco 3600 VoIP dial-peer configuration command.
12.0(4)T	This command was modified for VoFR dial peers. On the Cisco MC3810, this command was first supported as a dial-peer command.
12.0(5)XK and 12.0(7)T	The **g729br8** codec and **pre-ietf** keyword were added for the Cisco 2600 and 3600 platforms.
12.0(7)XK	The **g729abr8** and **g729ar8** codecs were added for the Cisco MC3810 and the keyword **pre-ietf** was deleted.

Usage Guidelines

A codec type can be configured on the dial-peer if it is supported under the **codec complexity** setting you have specified.

The 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, **g711alaw** and **g711ulaw** provide higher-quality voice transmission than other codecs. For almost toll quality (and a significant savings in bandwidth), **g729r8** provides near-toll quality with considerable bandwidth savings.

If the destination router does not support a codec required by the originating router, the call setup fails.

You can change the payload of each voice packet frame by using the **bytes** *payload-size* setting. However, increasing the payload size can add processing delay for each voice packet. Table 3 describes the voice payload options and default values for the codecs and packet voice protocols.

Table 3 ***Voice Payload-per-Frame Options and Defaults***

Codec	Protocol	Voice Payload Options (bytes)	Default Voice Payload (bytes)
g711alaw	VoIP	80, 160	160
g711ulaw	VoFR	40 to 240 in multiples of 40	240
	VoATM	40 to 240 in multiples of 40	240
g723ar53	VoIP	20 to 220 in multiples of 20	20
g723r53	VoFR	20 to 240 in multiples of 20	20
	VoATM	20 to 240 in multiples of 20	20
g723ar63	VoIP	24 to 216 in multiples of 24	24
g723r63	VoFR	24 to 240 in multiples of 24	24
	VoATM	24 to 240 in multiples of 24	24
g726r16	VoIP	20 to 220 in multiples of 20	40
	VoFR	10 to 240 in multiples of 10	60
	VoATM	10 to 240 in multiples of 10	60
g726r24	VoIP	30 to 210 in multiples of 30	60
	VoFR	15 to 240 in multiples of 15	90
	VoATM	30 to 240 in multiples of 15	90

Table 3 ***Voice Payload-per-Frame Options and Defaults***

Codec	Protocol	Voice Payload Options (bytes)	Default Voice Payload (bytes)
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

Example

The following example configures VoIP dial peer number 10 to use codec type **g723r53** (G.723.1 at 5300 bps), and specifies a non-default voice payload size of 40 bytes:

```
router(config)# dial-peer voice 10 voip
router(config-dialpeer)# codec g723r53 bytes 40
```

Related Commands

Command	Description
codec complexity	This voice-card configuration command sets codec complexity and call density.
show dial-peer voice	Displays the codec setting for dial peers.

codec complexity

To match the DSP complexity packaging to the codec(s) to be supported, enter the **codec complexity** voice-card configuration command. The **no** form of the command restores the default value.

codec complexity {**high** | **medium**}
no codec complexity

Syntax Description

high	With high complexity packaging, each DSP supports two voice channels encoded in any of the following formats: G.711ulaw, G.711alaw, G.723.1(r5.3), G.723.1 Annex A(r5.3), G.723.1(r6.3), G.723.1 Annex A(r6.3), G.726(r16), G.726(r24), G.726(r32), G.729, G.729 Annex B, G.728, and fax relay.
medium	With medium complexity packaging, each DSP supports four voice channels encoded in any of the following formats: G.711ulaw, G.711alaw, G.726(r16), G.726(r24), G.726(r32), G.729 Annex A, G.729 Annex B with Annex A, and fax relay. This is the default.

Defaults

The DSP supports medium complexity codecs.

Command Mode

Voice-card configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	The command was introduced for the Cisco 2600 and 3600 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform for use with the high performance compression module (HCM).

Usage Guidelines

Select a higher codec complexity if that is required in order to support a particular codec or combination of codecs.

Select a lower codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use.

To change codec complexity, all of the DSP voice channels must be in the idle state.

Codec complexity refers to the amount of processing required to perform voice compression. Codec complexity affects the call density—the number of calls that can take place on the digital signal processors (DSPs). With higher codec complexity, fewer calls can be handled.

Note On the Cisco MC3810, this command is valid only with HCM(s) installed, and you must specify voice card 0 in the command mode. If two HCMs are installed, the **codec complexity** command configures both HCMs at once.

Examples

The following example sets the codec complexity to high on a Cisco MC3810 containing one or two HCMs:

```
router(config)# voice-card 0
router(config-voicecard)# codec complexity high
```

The following example sets the codec complexity to high on voice card 1 in a Cisco 2600 or 3600 series router:

```
router(config)# voice-card 1
router(config-voicecard)# codec complexity high
```

Related Command

Command	Description
show voice dsp	Shows the current status of all DSP voice channels.

ds0-group

To specify the DS0 timeslots that make up a logical voice port on a T1 or E1 controller, and to specify the signaling type, enter the **ds0-group** controller configuration command. Use the **no** form of the command to remove the DS0 group and signaling setting.

ds0-group *ds0-group-no* **timeslots** *timeslot-list* **type** *signal-type*

no ds0-group *ds0-group-no*

Syntax Description

ds0-group-no	A number from 0 to 23 (for T1) or from 0 to 30 (for E1) that identifies the DS0 group.
timeslot-list	*timeslot-list* can be a single timeslot number, a single range of numbers, or multiple ranges of numbers separated by commas. Allowable values are 1 to 24 for T1 and 1 to 32 for E1. Examples are: • 2 • 1-15, 17-23 • 1-23 • 2, 4, 6-12, 27-32
type	The signaling method selection for **type** depends on the connection that you are making. The E&M interface allows connection for PBX trunk lines (tie lines) and telephone equipment. The FXS interface allows connection of basic telephone equipment and PBXs. The FXO interface is for connecting the central office (CO) to a standard PBX interface where permitted by local regulations; it is often used for off-premises extensions. The following options are available on T1 and E1 controllers: • **e&m-immediate-start—**no specific off-hook and on-hook signaling • **e&m-delay-dial—**the originating endpoint sends an off-hook signal and then waits for an off-hook signal followed by an on-hook signal from the destination • **e&m-wink-start—**the originating endpoint sends an off-hook signal and waits for a wink signal from the destination • **fxs-ground-start—**Foreign Exchange Station ground-start signaling support • **fxs-loop-start —**Foreign Exchange Station loop-start signaling support • **fxo-ground-start—**Foreign Exchange Office ground-start signaling support • **fxo-loop-start—**Foreign Exchange Office loop-start signaling support

The following options are available only on E1 controllers on the Cisco MC3810:

- **e&m-melcas-immed**—E&M Mercury Exchange Limited Channel Associated Signaling (MELCAS) immediate start signaling support
- **e&m-melcas-wink**—E&M MELCAS wink start signaling support
- **e&m-melcas-delay**—E&M MELCAS delay start signaling support
- **fxo-melcas**—MELCAS Foreign Exchange Office signaling support
- **fxs-melcas**—MELCAS Foreign Exchange Station signaling support

The following options are available only when the **mode ccs** command is enabled on the Cisco MC3810 for transparent CCS support:

- **ext-sig-master—**For the specified channel(s), automatically generates the off-hook signal and stays in the off-hook state
- **ext-sig-slave—**For the specified channel(s), automatically generates the answer signal when a call is terminated to that channel

Default

No DS0 group is defined.

Command Mode

Controller configuration

Command History

Release	Modification
11.2	This command was introduced for the Cisco AS5300 as **cas-group**.
12.0(1)T	The **cas-group** command was first supported on the Cisco 3600 series.
12.0(5)T	This command was renamed **ds0-group** on the Cisco AS5300 and on the Cisco 2600 and 3600 series (requires Digital T1 Packet Voice Trunk Network Modules).
12.0(7)XK	Support for this command was extended to the Cisco MC3810. When the **ds0-group** command became available on the Cisco MC3810, the **voice-group** command was removed and no longer supported.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows on the Cisco MC3810: *slot:ds0-group-no*.

On the Cisco MC3810, the *slot* number is the controller number. Although only one voice port is created for each DS0 group, applicable calls are routed to any channel in the group.

On the Cisco MC3810 when configured for transparent CCS, the channel type configured as the **ext-sig-master** is considered the master side of the permanent virtual circuit (PVC) connection which is responsible for establishing the PVC connection. After the master channel is configured, a fixed timer of 30 seconds starts. The voice-signaling driver then generates an off-hook signal on the master voice channel after the timer expires. The call is treated as a regular call, and the master channel does not hang up after the connection is made. If the call does not go through, or if the T1/E1 trunk is down, the 30-second timer on the master channel side restarts. A new off-hook signal is then generated at the master channel side after the timer expires.

Example

The following example configures DS0 groups 0 and 1 with different CAS signaling on controller T1 0:

```
router(config)# controller T1 0
router(config-controller)# mode cas
router(config-controller)# framing esf
router(config-controller)# linecode b8zs
router(config-controller)# ds0-group 0 timeslot 1-10 type fxs-ground-start
router(config-controller)# ds0-group 1 timeslot 11-24 type fxo-loop-start
```

The following example configures DS0 groups 1 and 2 on controller T1 0 to support transparent CCS:

```
router(config)# controller T1 0
router(config-controller)# mode ccs cross-connect
router(config-controller)# ds0-group 1 timeslot 1-10 type ext-sig-master
router(config-controller)# ds0-group 2 timeslot 11-24 type ext-sig-slave
```

Related Command

Command	Description
codec complexity	Matches the DSP complexity packaging to the codec(s) to be supported. Voice channels in DS0 groups must be in the idle state before codec complexity can be changed.
mode ccs	Configures the T1/E1 controller to support CCS cross-connect or CCS frame-forwarding.

voice-card

To configure a voice card and enter voice-card configuration mode, enter the **voice-card** command.

voice-card *slot*

Syntax Description

slot	On the Cisco 2600 and 3600 series: • A number from 0 to 3 that identifies the physical slot in the chassis where the voice card is located. On Cisco MC3810 series concentrators with one or two HCMs installed: • Enter 0 only; this applies to the entire chassis.

Command Mode

Global configuration

Command History

Release	Modification
12.0(5)XK and 12.0(7)T	The command was introduced for the Cisco 2600 and 3600 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

You can configure codec complexity only in voice-card configuration mode. On the Cisco 2600 and 3600 series routers, the slot corresponds to the physical slot in the chassis. On the Cisco MC3810 series, the slot is always 0, and all changes made in voice-card mode apply to the entire Cisco MC3810. On Cisco MC3810 series concentrators, this command is available only if the chassis is equipped with one or two HCMs.

Example

The following example enters voice-card configuration mode for the voice card in slot 1 on a Cisco 2600 or 3600 router:

```
router(config)# voice-card 1
   router(config-voicecard)#
```

The following example enters voice-card configuration mode on a Cisco MC3810 concentrator:

```
router(config)# voice-card 0
   router(config-voicecard)#
```

Related Commands

Command	Description
codec complexity	Matches the DSP complexity packaging to the codec(s) to be supported. Codec complexity changes are made in the voice-card configuration mode.