New and Changed Commands in Cisco IOS Release 12.0(7)XK

This document provides a consolidated command reference of all the new, changed, and removed commands in Cisco IOS Release 12.0(7)XK.

This document includes the following sections:

Related Documents

The command reference entries in this document are also included in one or more of the following 12.0(7)XK online documents:

- *Configuring Voice over IP on Cisco MC3810 Concentrators*
- *Voice over ATM on the Cisco 3600 Series Routers*
- *Voice over Frame Relay Using FRF.11 and FRF.12 Configuration Updates*
- *Voice Port Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
- *Voice Port Testing Enhancements on Cisco 2600, 3600 and MC3810*
- *Configuring Cisco MC3810 Series Concentrators to Use High-Performance Compression Modules*
- *QSIG Protocol Support on Cisco MC3810, 7200, 2600, and 3600 Series Routers*
- *Transparent CCS and Frame Forwarding Enhancements on the Cisco MC3810*
- *Trunk Conditioning Enhancements in MC3810 Series Concentrators*
- *Voice Busyout Enhancements on the Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
- *Configuring Digital E1 Packet Voice Trunk Network Module Interfaces*

Command Reference

This section documents new, modified and removed commands. Modified commands are indicated by an asterisk (*). All other commands used on these platforms are documented in the Cisco IOS Release 12.0 command reference publications.

- **auto-cut-through***
- **battery-reversal**
- **ccs connect**
- **ccs encap frf11**
- **codec (dial-peer)***
- **codec complexity***
- **condition***
- **connection***
- **define***
- **dial-peer hunt**
- **dial-peer terminator***
- **dial-peer voice***
- **disconnect-ack***
- **ds0-group***
- **encapsulation***
- **encapsulation ftc-trunk***
- **forward-digits***
- **frame-relay interface-dlci***
- **huntstop***
- **icpif***
- **ignore***
- **incoming called-number***
- **isdn contiguous-bchan**
- **isdn incoming-voice***
- **isdn protocol-emulate***
- **isdn switch type***
- **loss-plan***
- **num-exp***
- **playout delay***
- **pri-group***
- **ring cadence***
- **session target***

- **show call active voice***
- **show call history voice***
- **show num-exp***
- **show voice call***
- **show voice dsp***
- **show voice port***
- **show voice trunk-conditioning signaling**
- **show voice trunk-conditioning supervisory**
- **signal pattern***
- **signal sequence oos**
- **signal timing idle suppress-voice**
- **signal-type***
- **test voice port detector**
- **test voice port inject-tone**
- **test voice port loopback**
- **test voice port relay**
- **test voice port switch**
- **timeouts ringing**
- **timeouts wait-release***
- **timing guard-out***
- **timing percentbreak***
- **vbr-rt***
- **vofr***
- **voice-card***
- **voice local-bypass**
- **voice vad-time**

The following commands have been removed in Cisco IOS Release 12.0(7)XK:

- **codec (voice-port)***
- **connect**
- **connect voice**
- **frag-pre-queuing**
- **ftc-trunk frame-relay-dlci**
- **ftc-trunk management-dlci**
- **ftc-trunk management-protocol**
- **ftc-trunk voice-dlci**
- **voice-encap**

- **voice-group**

auto-cut-through

To enable call completion when a PBX does not provide an M-lead response, use the **auto-cut-through** voice-port configuration command. Use the **no** form of this command to disable the auto-cut-through operation.

auto-cut-through
no auto-cut-through

Syntax Description

This command has no arguments or keywords.

Defaults

Auto-cut-through is enabled.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

The **auto-cut-through** command applies to E&M voice ports only.

Examples

The following example enables call completion on a Cisco MC3810 when a PBX does not provide an M-lead response:

```
router(config)# voice-port 1/1
router(config-voiceport)# auto-cut-through
```

The following example enables call completion on a Cisco 2600 or 3600 when a PBX does not provide an M-lead response:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# auto-cut-through
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

battery-reversal

To specify battery polarity reversal on an FXO or FXS port, use the **battery-reversal** voice-port configuration command. Use the **no** form of this command to disable battery reversal.

battery-reversal
no battery-reversal

Syntax Description

This command has no arguments or keywords.

Defaults

Battery reversal is enabled.

Command Mode

Voice-port configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

The **battery-reversal** command applies to FXO and FXS voice ports. On Cisco 2600 and 3600 series routers, only analog voice ports in VIC-2FXO-M1 and VIC-2FXO-M2 voice interface cards are able to detect battery reversal; analog voice ports in VIC-2FXO and VIC-2FXO-EU voice interface cards do not detect battery reversal. On digital voice ports, battery reversal is only supported on E1 MELCAS; it is not supported in T1 channel associated signaling (CAS) or E1 CAS.

FXS ports normally reverse battery upon call connection. If an FXS port is connected to an FXO port that does not support battery reversal detection, you can use the **no battery-reversal** command on the FXS port to prevent unexpected behavior.

FXO ports in loopstart mode normally disconnect calls when they detect a second battery reversal (back to normal). You can use the **no battery-reversal** command on FXO ports to disable this action.

The **battery-reversal** command restores voice ports to their default battery-reversal operation.

Examples

The following example disables battery reversal on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
   router(config-voiceport)# no battery-reversal
```

The following example disables battery reversal on voice port 1/0/0 on a Cisco 2600 or 3600 series router:

```
router(config)# voice-port 1/0/0
   router(config-voiceport)# no battery-reversal
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

ccs connect

To configure a CCS connection on an interface configured to support CCS frame forwarding, use the **ccs connect** interface configuration command. To disable the CCS connection on the interface, use the **no** form of this command.

ccs connect {serial | atm} *number* [*dlci* | **pvc** *vpi/vci* | **pvc** *name*] [*cidnumber*]
no ccs connect {serial | atm} *number* [*dlci* | **pvc** *vpi/vci* | **pvc** *name*] [*cidnumber*]

Syntax Description

number	Specify the connection number.

The following parameters are used for Frame Relay configuration:

serial	Make a serial CCS connection.
dlci	Specify the DLCI number.
cidnumber	(Optional) If you have executed the **ccs encap frf11** command, the **cid** option allows you to specify any CID number from 5 to 255.

The following parameters are used for ATM configuration:

atm	Make an ATM CCS connection.
pvc *vpi/vci*	Specify the PVC virtual path identifier/virtual channel identifier. Acceptable values are from 0 to 255; the slash is required.
pvc *name*	Specify the PVC string that names the PVC for recognition.

Defaults

No CCS connection is made.

Command Mode

Serial interface configuration mode

Command History

Release	Modification
12.0(2)T	This command was introduced for the Cisco MC3810.
12.0(7)XK	Added CID syntax, removed **dlci** keyword and **vcd** options.

Usage Guidelines

Use this command to configure a CCS connection. If the CCS connection is over Frame Relay, specify a serial interface and the DLCI. If the CCS connection is over ATM, specify **atm**, the interface number (0 only on the Cisco MC3810), and the PVC.

If you have executed the **ccs encap frf11** command, the *cidnumber* option allows you to specify any CID from 5 to 255. If you do not issue the **ccs encap frf11** command, Cisco encapsulation is used, and any CID value other than 254 is ignored.

> **Note** CDP and keepalives are disabled by default on a D channel interface.

Examples

To configure a frame relay CCS frame-forwarding connection on DLCI 100 by using the default CID of 254, enter the following command:

```
ccs connect serial 1 100
```

or:

```
ccs connect serial 1 100 10
```

To configure a CCS frame-forwarding connection over an ATM PVC, enter the following command:

```
ccs connect atm0 pvc 100/10
```

or:

```
ccs connect atm0 pvc 10/100 21
```

or:

```
ccs connect atm0 pvc mypvc_10 21
```

To configure a Frame Relay CCS frame-forwarding connection on DLCI 100 using a CID of 110, enter the following command:

```
ccs connect serial 1 100 110
```

Related Commands

Command	Description
ccs encap frf11	Allows the specification of the standard Annex-C FRF.11 format.

ccs encap frf11

To configure the common channel signaling (CCS) packet encapsulation format for FRF.11, use the **ccs encap frf11** command. Use the **no** form of this command to disable ccs encapsulation for FRF11.

ccs encap frf11

no ccs encap frf11

Syntax Description

There are no keywords or arguments.

Defaults

By default, the format is a Cisco packet format, using a channel ID (CID) of 254.

Command Mode

Serial configuration mode

Command History

Release	Modification
12.0(7)XK	This command was introduced for the Cisco MC3810.

Usage Guidelines

This command allows the specification of the standard Annex-C format. Use this command to define the packet format for the CCS packet; it places the FRF.11 Annex-C (Data Transfer Syntax) standard header on the CCS packets only.

Once the **ccs encap frf11** command is executed, you can use the **ccs connect** command to specify a CID other than 254.

Examples

The following example shows how to configure a serial interface for Frame Relay:

```
router(config)# interface Serial1:15
router(config-if)# ccs encap frf11
router(config-if)# ccs connect Serial0 990 100
```

Related Commands

Command	Description
mode ccs frame-forwarding	Set to forward frames on the controller.

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 {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8**}[**bytes** *payload-size*]
no codec

Syntax Description

codec	Codec options on the Cisco MC3810 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. Acceptable values are from 10 to 240 in increments of 10 (10, 20, 30 ... 220, 230, 240). Any other value is rounded down.

Defaults

Dial peers are configured for **g729r8**.

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

Table 1 Voice Payload-per-Frame Options and Defaults

Codec	Protocol	Voice Payload Options (bytes)	Default Voice Payload (bytes)
g711alaw **g711ulaw**	VoIP	80, 160	160
	VoFR	40 to 240 in multiples of 40	240
	VoATM	40 to 240 in multiples of 40	240
g723ar53 **g723r53**	VoIP	20 to 220 in multiples of 20	20
	VoFR	20 to 240 in multiples of 20	20
	VoATM	20 to 240 in multiples of 20	20
g723ar63 **g723r63**	VoIP	24 to 216 in multiples of 24	24
	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 1 Voice Payload-per-Frame Options and Defaults (Continued)

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):

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

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 (voice-port)

The **codec** voice-port configuration command on the Cisco MC3810 is no longer supported beginning in this release. This command was first supported in Cisco IOS Release 11.3(1)MA. Configure the codec value using the **codec** dial-peer configuration command.

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 default is medium complexity.

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 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.

condition

To manipulate the signaling format bit-pattern for all voice signaling types, use the **condition** command. Use the **no** form of this command to turn off conditioning on the voice port.

condition {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}
no condition {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}

Syntax Description

tx-a-bit	Transmit A bit.
tx-b-bit	Transmit B bit.
tx-c-bit	Transmit C bit.
tx-d-bit	Transmit D bit.
rx-a-bit	Receive A bit.
rx-b-bit	Receive B bit.
rx-c-bit	Receive C bit.
rx-d-bit	Receive D bit.
on	Forces the bit state to be 1.
off	Forces the bit state to be 0.
invert	Inverts the bit state.

Defaults

The signaling format is not manipulated (for all transmit or receive A, B, C, and D bits).

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Use the **condition** command to manipulate the sent or received bit patterns to match expected patterns on a connected device. Be careful not to destroy the information content of the bit pattern. For example, forcing the A-bit on or off will prevent FXO interfaces from being able to generate both an on-hook and off-hook state.

Examples

The following example manipulates the signaling format bit-pattern on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# condition tx-a-bit invert
router(config-voiceport)# condition rx-a-bit invert
```

The following example manipulates the signaling format bit-pattern on voice port 1/1/2 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# condition tx-a-bit invert
router(config-voiceport)# condition rx-a-bit invert
```

Related Commands

Command	Description
define	Defines the transmit and receive bits for E&M and E&M MELCAS voice signaling.
ignore	Configures the E&M or E&M MELCAS voice port to ignore specific receive bits.

connect

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

connect voice

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

connection

To specify a connection mode for a voice port, use the **connection** voice-port configuration command. Use the **no** form of this command to disable the selected connection mode.

connection {**plar** | **tie-line** | **plar-opx**} *digits* | {**trunk** *digits* [**answer-mode**]}
no connection {**plar** | **tie-line** | **plar-opx**} *digits* | {**trunk** *digits* [**answer-mode**]}

Syntax Description

plar	Specifies a private line automatic ring down (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off hook a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX.
tie-line	Specifies a connection that emulates a temporary tie-line trunk to a private branch exchange (PBX). A tie-line connection is automatically set up for each call and torn down when the call ends.
plar-opx	Specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice port provides a local response before the remote voice port receives an answer. On FXO interfaces, the voice port will not answer until the remote side answers.
trunk	Specifies a connection that emulates a permanent trunk connection to a private branch exchange (PBX). A trunk connection remains "nailed up" in the absence of any active calls.
digits	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
answer-mode	(Optional; used only with the **trunk** keyword.) Specifies that the router should not attempt to initiate a trunk connection, but should wait for an incoming call before establishing the trunk.

Defaults

No connection mode is specified.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA1	This command was first supported on the Cisco MC3810, and the **tie-line** keyword was first made available on the Cisco MC3810.
11.3(1)MA5 and 12.0(2)T	The **plar-opx** keyword was first made available on the Cisco MC3810 as the **plar-opx-ringrelay** keyword. The keyword was shortened in a subsequent release.
12.0(3)XG	The **trunk** keyword was made available on the Cisco MC3810. The **trunk answer-mode** option was added.
12.0(7)XK	This command was unified across the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

Use this command to specify a connection mode for a specific interface. For example, use the **connection plar** command to specify a PLAR interface. The string you configure for this command is used as the called number for all incoming calls over this connection. The destination peer is determined by the called number.

Use the **connection trunk** command to specify a permanent, “nailed up” tie-line connection to a PBX. You can use the **connection trunk** command for E&M-to-E&M trunks, FXO-to-FXS trunks, and FXS-to-FXS trunks. Signaling will be transported for E&M-to-E&M trunks and FXO-to-FXS trunks; signaling will not be transported for FXS-to-FXS trunks.

To configure one of the devices in the trunk connection to act as slave and only receive calls, use the **answer-mode** option with the **connection trunk** command when configuring that device.

Note When using the **connection trunk** command, you must perform a **shutdown/no shutdown** command sequence on the voice port.

Use the **connection tie-line** command when the dial plan requires that additional digits be added in front of any digits dialed by the PBX, and that the combined set of digits be used to route the call onto the network. The operation is similar to the **connection plar** command operation, but in this case the tie-line port waits to collect digits from the PBX. The tie-line digits are automatically stripped by a terminating port.

If the **connection** command is not configured, the standard session application outputs a dial tone when the interface goes off-hook until enough digits are collected to match a dial-peer and complete the call.

Examples

The following example selects PLAR as the connection mode on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection trunk 5559262
```

The following example selects tie-line as the connection mode on a Cisco MC3810, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/1
router(config-voiceport)# connection tie-line 5559262
```

The following example specifies a PLAR off-premises extension connection on a Cisco 3600, with a destination telephone number of 555-9262:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection plar-opx 5559262
```

The following example configures a Cisco 3600 series router for a trunk connection and specifies that it will establish the trunk only when it receives an incoming call:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# connection trunk 5559262 answer-mode
```

Related Commands

Command	Description
session-protocol	Establishes a session protocol for calls between the local and remote routers via the packet network.
session-target	Configures a network-specific address for a dial peer.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

define

To define the transmit and receive bits for E&M and E&M Mercury Exchange Limited (MELCAS) voice signaling, use the **define** voice-port configuration command. Use the **no** form of this command to restore the default value.

define {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}
no define {**Tx-bits** | **Rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}

Syntax Description

Tx-bits	Transmit signaling bits.
Rx-bits	Receive signaling bits.
seize	The bit pattern defines the seized state.
idle	The bit pattern defines the idle state.
0000 through **1111**	Specifies the bit pattern.

Defaults

The default is to use the preset signaling patterns as defined in ANSI and CEPT standards, as follows:

For E&M:

Tx-bits idle 0000 (0001 if on E1 trunk)
Tx-bits seize 1111
Rx-bits idle 0000
Rx-bits seize 1111

For E&M MELCAS:

Tx-bits idle 1101
Tx-bits seize 0101
Rx-bits idle 1101
Rx-bits seize 0101

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1) MA3	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to E&M digital voice ports associated with T1/E1 controllers.

Use the **define** command to match the E&M bit patterns with the attached telephony device. Be careful not to define invalid configurations, such as all 0000 on E1, or identical seized and idle states. Use this command with the **ignore** command.

Examples

To configure a voice port on a Cisco 2600 or 3600 router sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# define rx-bits idle 1101
router(config-voiceport)# define rx-bits idle 0101
router(config-voiceport)# define tx-bits seize 1101
router(config-voiceport)# define tx-bits seize 0101
```

To configure a voice port on a Cisco MC3810 sending traffic in North American E&M signaling format to convert the signaling to MELCAS format, enter the following commands:

```
router(config)# voice-port 0/8
router(config-voiceport)# define rx-bits idle 1101
router(config-voiceport)# define rx-bits idle 0101
router(config-voiceport)# define tx-bits seize 1101
router(config-voiceport)# define tx-bits seize 0101
```

Related Commands

Command	Description
condition	Manipulate the signaling bit-pattern for all voice signaling types.
ignore	Configures an E&M or E&M MELCAS voice port to ignore specific receive bits.

dial-peer hunt

To specify a hunt selection order for dial-peers, use the **dial-peer hunt** dial-peer configuration command. Use the **no** form of this command to restore the default selection order.

dial-peer hunt *hunt-order-number*
no dial-peer hunt

Syntax Description

hunt-order-number	A number from 0 to 7 that selects a predefined hunting selection order: 0—Longest match in phone number, explicit preference, random selection. This is the default hunt order number. 1—Longest match in phone number, explicit preference, least recent use. 2—Explicit preference, longest match in phone number, random selection. 3—Explicit preference, longest match in phone number, least recent use. 4—Least recent use, longest match in phone number, explicit preference. 5—Least recent use, explicit preference, longest match in phone number. 6—Random selection. 7—Least recent use.

Defaults

The default is the longest match in phone number, explicit preference, and random selection (hunt order number 0).

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced, and was first supported on the Cisco 2600 and 3600 series routers and on the Cisco MC3810 multiservice access concentrator.

Usage Guidelines

Use the **dial-peer hunt** dial-peer configuration command if you have configured hunt groups. “Longest match in phone number” refers to the destination pattern that matches the greatest number of the dialed digits. “Explicit preference” refers to the **preference** setting in the dial-peer configuration. “Least recent use” refers to the destination pattern that has waited the longest since being selected. “Random selection” weighs all of the destination patterns equally in a random selection mode.

Example

The following example configures the dial peers to hunt in the following order: (1) longest match in phone number, (2) explicit preference, (3) random selection.

```
router# configure terminal
router(config)# dial-peer hunt 0
```

Related Commands

Command	Description
preference	Specifies the preferred selection order of a dial peer within a hunt group.
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer terminator

To change the character used as a terminator for variable length dialed numbers, use the **dial-peer terminator** global configuration command. Use the **no** form of this command to restore the default terminating character.

dial-peer terminator *character*
no dial-peer terminator

Syntax Description

character	Designates the terminating character for a variable-length dialed number. Valid numbers and characters are #, *, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, a, b, c, and d. The default is #.

Defaults

The default terminating character is #.

Command Mode

Global configuration

Command History

Release	Modification
12.0	This command was introduced.
12.0(7)XK	Usage was restricted to variable-length dialed numbers.

Usage Guidelines

There are certain areas in the world, 0for example, in certain European countries, where telephone numbers can vary in length. When a dialed-number string is identified as a variable length dialed-number, the system does not place a call until the configured value for the **timeouts interdigits** command has expired, or until the caller dials the terminating character. Use the **dial-peer terminator** global configuration command to change the terminating character.

Example

The following example specifies “9” as the terminating character for variable-length dialed numbers:

```
router# configure terminal
router(config)# dial-peer terminator 9#
```

Related Commands

Command	Description
answer-address	Specifies the preferred selection order of a dial peer within a hunt group.
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
timeouts interdigit	Specifies the interdigit timeout value for a voice port in seconds.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer voice

To enter dial-peer configuration mode and specify the method of voice encapsulation, use the **dial-peer voice** global configuration command.

For the Cisco 2600 series:

dial-peer voice *tag* {**pots** | **voip** | **vofr**}
no dial-peer voice *tag*

For the Cisco 3600 series and the Cisco MC3810:

dial-peer voice *tag* {**pots** | **voip** | **vofr** | **voatm**}
no dial-peer voice *tag*

Syntax Description

tag	A number identifying a particular dial peer. Valid entries are 1 to 2147483647.
pots	POTS dial peer using basic telephone service.
voip	VoIP dial peer using voice encapsulation on the POTS network.
vofr	Voice over Frame Relay dial peer using encapsulation on the Frame Relay backbone network.
voatm	(Cisco 3600 and MC3810 only) Voice over ATM dial peer using real-time AAL5 voice encapsulation on the ATM backbone network.

Defaults

No default behavior or values.

Command Mode

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	This command was first supported on the Cisco MC3810 with support for POTS, VoFR and VoATM.
12.0(3)XG	This command added VoFR to the Cisco 2600 and 3600 series routers.
12.0(4)T	This command added VoFR to the Cisco 7200 series platform.
12.0(7)XK	This command added VoIP to the Cisco MC3810 and VoATM to the Cisco 3600 series routers. Support for VoHDLC on the Cisco MC3810 was removed in this release.

Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global configuration mode.

Examples

The following example accesses dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
router# configure terminal
router(config)# dial-peer voice 10 pots
```

The following example accesses dial-peer configuration mode and configures a VoATM peer identified as dial peer 20:

```
router# configure terminal
router(config)# dial-peer voice 20 voatm
```

Related Commands

Command	Description
voice-port	Enters voice-port configuration mode.

disconnect-ack

To configure an FXS voice port to return an acknowledgment upon receipt of a disconnect signal, use the **disconnect-ack** voice-port configuration command. To disable the acknowledgment, use the **no** form of this command.

disconnect-ack
no disconnect-ack

Syntax Description

This command has no arguments or keywords.

Defaults

FXS voice ports return an acknowledgment upon receipt of a disconnect signal.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command configures an FXS voice port to remove line power if the equipment on an FXS loop-start trunk disconnects first.

Examples

The following example turns off the disconnect acknowledgment signal on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# no disconnect-ack
```

The following example turns off the disconnect acknowledgment signal on voice port 1/1/0 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# no disconnect-ack
```

Related Commands

Command	Description
show voice port	Displays voice port configuration information.

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, use 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 value from 0 to 23 that identifies the DS0 group.
timeslot-list	*timeslot-list* is a single timeslot number, a single range of numbers, or multiple ranges of numbers separated by commas. For T1, allowable values are from 1 to 24. Examples are: • 2 • 1-15, 17-24 • 1-23 • 2, 4, 6-12
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. The FXO interface is often used for off-premises extensions. The options are as follows: • **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 **ext-sig** option is available only when the **mode ccs** command is enabled on the Cisco MC3810 for FRF.11 transparent CCS support.

Defaults

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. **ext-sig** replaces the **ext-sig-master** and **ext-sig-master** options that were available with the **voice-group** command.

Usage Guidelines

The **ds0-group** command automatically creates a logical voice port that is numbered as follows:

Cisco 2600 and 3600 series:

slot/*port*:*ds0-group-no*.

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 group, applicable calls are routed to any channel in the group.

Examples

The following example configures ranges of T1 controller timeslots for FXS ground-start and FXO loop-start signaling on a Cisco 2600 or 3600 Series router:

```
router(config)# controller T1 1/0
router(config-controller)# framing esf
router(config-controller)# linecode b8zs
router(config-controller)# ds0-group 1 timeslot 1-10 type fxs-ground-start
router(config-controller)# ds0-group 2 timeslot 11-24 type fxo-loop-start
```

The following example configures DS0 groups 1 and 2 on controller T1 1 on the Cisco MC3810 to support Transparent CCS:

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

Related Commands

Command	Description
codec complexity	Matches the DSP complexity packaging to the codec(s) to be supported
mode ccs	Configures the T1/E1 controller to support CCS cross-connect or CCS frame-forwarding.

encapsulation

To configure the ATM adaptation layer (AAL) and encapsulation type for an ATM PVC class, use the **encapsulation** command in the appropriate command mode. Use the **no** form of this command to remove an encapsulation from a PVC class.

encapsulation *aal-encap*
no encapsulation *aal-encap*

Note This document only describes encapsulation settings for Voice over ATM. For the full syntax of the encapsulation command, refer to Cisco IOS 12.0 *Wide Area Networking Command Reference*.

Syntax Description

aal-encap	ATM adaptation layer (AAL) and encapsulation type. Possible values for *aal-encap* are as follows: • **aal5mux voice**—For a MUX-type virtual circuit for Voice over ATM. • **aal5snap**—The only encapsulation supported for Inverse ARP. Logical Link Control/Subnetwork Access Protocol (LLC/SNAP) precedes the protocol datagram.

Defaults

The global default encapsulation is aal5snap. See the "Usage Guidelines" section for other default characteristics.

Command Mode

Interface-ATM-VC configuration (for an ATM PVC or SVC)

Command History

Release	Modification
11.3 T	This command was introduced.
12.0	This command superseded the **encapsulation atm** command for the Cisco MC3810, and the **aal5mux frame** and **aal5mux voice** suboptions appeared.
12.0(7)XK	Support for the **aal5mux voice** option was added to the Cisco 3600 series routers.

Usage Guidelines

Use one of the **aal5mux** encapsulation options to dedicate the specified PVC to a single protocol; use the **aal5snap** encapsulation option to multiplex two or more protocols over the same PVC. Whether you select **aal5mux** or **aal5snap** encapsulation depends on practical considerations, such as the type of network and the pricing offered by the network. If the network's pricing depends on the number of PVCs set up, **aal5snap** may be the appropriate choice. If pricing depends on the number of bytes transmitted, **aal5mux** may be the appropriate choice because it has slightly less overhead.

If you specify virtual template parameters after the ATM PVC is configured, issue a **shutdown** command followed by a **no shutdown** command on the ATM subinterface to restart the interface, causing the newly configured parameters (such as an IP address) to take effect.

Example

The following example configures a PVC to support encapsulation for Voice over ATM:

```
router(config-if)# pvc 20
router(config-if-atm-pvc)# encapsulation aal5mux voice
```

encapsulation ftc-trunk

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

forward-digits

To specify which digits to forward for voice calls, use the **forward-digits** dial-peer configuration command. If the **no** form of this command is entered, any digits not matching the destination-pattern are not forwarded. Use the **default** form of this command to restore the default state.

forward-digits {*num-digit* | **all** | **extra**}
no forward-digits
default forward-digits

Syntax Description

num-digit	The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. The valid range is 0 to 32. Setting the value to 0 is equivalent to entering **no forward-digits**.
all	Forward all digits. If **all** is entered, the full length of the destination pattern is used.
extra	If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length (ending with character "T", for example: T, 123T, 123...T), extra digits are not forwarded.

Defaults

Dialed digits not matching the destination-pattern are forwarded.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1) MA	This command was introduced on the Cisco MC3810.
12.0(2) T	The **implicit** option was added.
12.0(4) T	This command was modified to support ISDN PRI QSIG signaling calls.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 platforms, the **implicit** keyword was removed, and the **extra** keyword was added.

Usage Guidelines

This command applies only to POTS dial peers.

Forwarded digits are always right-justified, so that extra leading digits are stripped.

The destination pattern includes both explicit digits and wildcards if present.

Use the **default** form of this command if a non-default digit-forwarding scheme was entered previously and you wish to restore the default.

For QSIG ISDN connections, entering **forward-digits all** implies that all the digits of the called party number are sent to the ISDN connection. When you enter **forward-digits** *num-digit* and enter a number from 1 to 32, the number of digits specified (right justified) of the called part number are sent to the ISDN connection.

Examples

The following example forwards all of the digits in the destination pattern of a POTS dial peer:

```
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# destination-pattern 8...
router(config-dial-peer)# forward-digits all
```

The following example forwards 4 of the digits in the destination pattern of a POTS dial peer:

```
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# destination-pattern 555....
router(config-dial-peer)# forward-digits 4
```

The following example forwards the extra right-justified digits that exceed the length of the destination pattern of a POTS dial peer:

```
router(config)# dial-peer voice 1 pots
router(config-dial-peer)# destination-pattern 555....
router(config-dial-peer)# forward-digits extra
```

Related Commands

Command	Description
destination-pattern	Defines the prefix or the full E.164 telephone number to be used for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

frag-pre-queuing

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

frame-relay interface-dlci

To assign a data link connection identifier (DLCI) to a specified Frame Relay subinterface on the router or access server, use the **frame-relay interface-dlci** interface configuration command. Use the **no** form of this command to remove this assignment.

frame-relay interface-dlci *dlci* [**ietf** | **cisco**] [**voice-cir** *cir*]
no frame-relay interface-dlci *dlci* [**ietf** | **cisco**] [**voice-cir** *cir*]

Syntax Description

dlci	DLCI number to be used on the specified subinterface.
ietf | **cisco**	(Optional) Encapsulation type: Internet Engineering Task Force (IETF) Frame Relay encapsulation or Cisco Frame Relay encapsulation.
voice-cir *cir*	(Optional; supported on the Cisco MC3810 only.) Specifies the upper limit on the voice bandwidth that may be reserved for this DLCI. The default is the CIR configured for the Frame Relay map class. For more information, see the "Usage Guidelines" section.

Defaults

No DLCI is assigned.

Command Mode

Interface configuration

Command History

Release	Modification
10.0	This command was introduced.
11.3(1) MA	The **voice-encap** option was added for the Cisco MC3810.
12.0(2) T	The **voice-cir** option was added for the Cisco MC3810.
12.0(3)XG and 12.0(4)T	Additional usage guidelines added.
12.0(7)XK	The **voice-encap** option on the Cisco MC3810 was removed. This option is no longer supported.

ftc-trunk frame-relay-dlci

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

ftc-trunk management-dlci

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

ftc-trunk management-protocol

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

ftc-trunk voice-dlci

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

huntstop

To disable all further dial-peer hunting if a call fails when using hunt groups, enter the **huntstop** dial-peer configuration command. To reenable dial-peer call hunting, enter the **no** form of this command.

huntstop
no huntstop

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Mode

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	Support for this command was extended to the Cisco 2600 and 3600 series routers.

Usage Guidelines

Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

This command can be used with all types of dial peers.

Examples

The following example shows how to disable dial-peer hunting on a specific dial peer:

```
router(config)# dial peer voice 100 vofr
router(config-dial-peer)# huntstop
```

The following example shows how to reenable dial-peer hunting on a specific dial peer:

```
router(config)# dial peer voice 100 vofr
router(config-dial-peer)# no huntstop
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

icpif

To specify the Impairment/Calculated Planning Impairment Factor (ICPIF) for calls sent by a dial peer, use the **icpif** dial peer configuration command. Use the **no** form of this command to restore the default value for this command.

icpif *number*
no icpif *number*

Syntax Description

number	Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are from 0 to 55.

Defaults

The default value for this command is 30.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

Use the **icpif** command to specify the maximum acceptable impairment factor for the voice calls sent by the selected dial peer.

This command is applicable only to VoIP peers.

Example

The following example disables the **icpif** command:

```
dial-peer voice 10 voip
 icpif 0
```

ignore

To configure the E&M or E&M MELCAS voice port to ignore specific receive bits, use the **ignore** voice-port configuration command. Use the **no** form of this command to restore the default value.

ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}
no ignore {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}

Syntax Description

rx-a-bit	Ignores the receive A bit.
rx-b-bit	Ignores the receive B bit.
rx-c-bit	Ignores the receive C bit.
rx-d-bit	Ignores the receive D bit.

Defaults

The default is mode-dependent:

E&M:

no ignore rx-a-bit
ignore rx-b-bit, rx-c-bit, rx-d-bit

E&M MELCAS:

no ignore rx-b-bit, rx-c-bit, rx-d-bit

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to E&M digital voice ports associated with T1/E1 controllers. Repeat the command for each receive bit to be configured. Use this command with the **define** command.

Examples

To configure voice-port 1/1 on a Cisco MC3810 to ignore receive bits a, b, and c and to monitor receive bit d, enter the following commands:

```
router(config)# voice-port 1/1
router(config-voiceport)# ignore rx-a-bit
router(config-voiceport)# ignore rx-b-bit
router(config-voiceport)# ignore rx-c-bit
router(config-voiceport)# no ignore rx-d-bit
```

To configure voice-port 1/0/0 on a Cisco 3600 to ignore receive bits a, c, and d and to monitor receive bit b, enter the following commands:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# ignore rx-a-bit
router(config-voiceport)# ignore rx-c-bit
router(config-voiceport)# ignore rx-d-bit
router(config-voiceport)# no ignore rx-b-bit
```

Related Commands

Command	Description
condition	Manipulates the signaling bit-pattern for all voice signaling types.
define	Defines the transmit and receive bits for E&M and E&M MELCAS voice signaling.
show voice port	Displays configuration information for voice ports.

incoming called-number

To identify the service type for a call on a router handling both voice and modem calls, use the **incoming called-number** dial peer configuration command. To return to the default value, use the **no** form of this command.

incoming called-number *string*
no incoming called-number *string*

Syntax Description

string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

Defaults

The default value for this command is no associated called number.

Command Mode

Dial peer configuration

Command History

Release	Modification
11.3NA	This command was introduced on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

When the Cisco MC3810 is handling both modem and voice calls, it needs to be able to identify the service type of the call—meaning whether the incoming call to the server is a modem or a voice call. When the access server handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the called number (DNIS). In a mixed environment, where the server receives both modem and voice calls, you need to identify the service type of a call by using the **incoming called-number** command.

If you do not use the **incoming called-number** command, the server attempts to resolve whether an incoming call is a modem or voice call based on the interface over which the call comes. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a dial peer, the call is assumed to be a voice call.

By default, there is no called number associated with the dial peer, which means that incoming calls will be associated with dial peers based on matching calling number with answer address, call number with destination pattern, or calling interface with configured interface.

This command applies to both VoIP and POTS dial peers.

Example

The following example configures calls coming in to the server with a called number of “3799262” as voice calls:

```
dial peer voice 10 pots
 incoming called-number 3799262
```

isdn contiguous-bchan

To configure contiguous bearer channel handling on an E1 Primary Rate Interface (PRI) interface, use the **isdn contiguous-bchan** interface configuration command. To disable the contiguous B channel handling, use the **no** form of this command.

isdn contiguous-bchan

no isdn contiguous-bchan

Syntax Description

This command has no arguments or keywords.

Defaults

By default, contiguous B channel handling is off.

Command Mode

Interface configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **isdn contiguous-bchan** command to specify contiguous bearer channel handling so that B channels 1 through 30, skipping 16, map to timeslots 1 through 31). This is available for E1 PRI interfaces only, when the **primary-qsig** switch type option is configured by using the **isdn switch-type** command.

Example

The following example shows the command configuration on a Cisco 3660 series router E1 interface:

```
interface Serial5/0:15
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
 isdn continguous-bchan
```

Related Commands

Command	Description
isdn switch-type primary-qsig	In global or interface configuration mode, configures the **primary-qsig** switch type for PRI support.

isdn incoming-voice

To route all incoming voice calls as voice calls, to route them the modem and treat them as analog data, or to ensure that calls bypass the modems and are treated as digital data, use the **isdn incoming-voice** interface configuration command. Use the **no** form of this command to disable the setting.

isdn incoming-voice {**data** [**56** | **64**] | **modem** [**56** | **64**] | **voice**}

no isdn incoming-voice {**data** [**56** | **64**] | **modem** [**56** | **64**] | **voice**}

Syntax Description

data	Specifies that incoming voice calls bypass the modems and are handled as digital data.
modem	Specifies that incoming voice calls are passed over to the digital modems, where they negotiate the appropriate modem connection with the far-end modem.
voice	Specifies that incoming voice calls are treated as voice calls rather than being routed to the modem or handled as digital data.
56	Specifies that the bandwidth for this connection is 56 kbps.
64	Specifies that the bandwidth for this connection is 64 kbps. If no argument is entered for either the **data** or **modem** keywords, the default value is 64.

Defaults

When a PRI or BRI interface is created, **isdn incoming-voice voice** is the default, except on a Cisco 2600 or 3600 BRI S/T TE interface. In this case, if the command is not specified, the default **isdn incoming-voice modem** configuration setting is converted to **isdn incoming-voice voice** when the interface receives an incoming call.

Command Mode

Interface configuration

Command History

Release	Modification
11.1	This command was introduced.
12.0(2)XC and 12.0(3)T	This command was made available for BRI interfaces.
12.0(7)XK	This command was modified to include the **voice** keyword.

Usage Guidelines

Unless you specify otherwise, all calls received by the router and characterized as voice calls are treated as such and not handled as digital data or not passed over to the modem.

On a Cisco 2600 or 3600 series router BRI S/T TE interface where the **isdn incoming-voice** command is not specified, the default **isdn incoming-voice modem** configuration setting is converted to **isdn incoming-voice voice** when the interface receives an incoming call.

To establish speedier connections for analog calls to the router, use the **isdn incoming-voice** command with the **modem** keyword to have voice calls routed through digital modems (as pulse-code modulated analog data) instead of being treated as digital data.

Example

The following example shows the command configuration on a Cisco 3660 series router T1 PRI interface:

```
interface Serial5/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn incoming-voice voice
```

isdn protocol-emulate

To configure a Primary Rate Interface (PRI) interface to serve as either the primary QSIG slave or the primary QSIG master, use the **isdn protocol-emulate** interface command. To disable QSIG signaling, use the **no** form of this command.

isdn protocol-emulate { user | network }

no isdn protocol-emulate { user | network }

Syntax Description

user	Enter **user** (equivalent to the QSIG term *slave*) to configure the port as the terminating end. This is the default.
network	Enter **network** (equivalent to the QSIG term *master*) to configure the port as NT; the PINX is the slave.

Defaults

User

Command Mode

Interface configuration mode.

Command History

Release	Modification
12.0(7)T	This command was introduced for the Cisco AS5300.
12.0(7)XK	This command was introduced for the Cisco MC3810, and for the Cisco 7200 VXR, Cisco 2600, and Cisco 3600 series routers.

Usage Guidelines

On the Cisco MC3810, this command replaces the command **isdn switch-type** [**primary-qsig-slave** | **primary-qsig-master**] command.

Examples

The following example shows the command configuration on a Cisco 3660 series router T1 PRI interface:

```
interface Serial5/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn protocol-emulate user
```

isdn switch type

To specify a central office switch type or configure a Primary Rate Interface (PRI) interface to support Q.SIG signaling, use the **isdn switch-type** global or interface command. To disable the central office switch type or QSIG signaling, use the **no** form of this command.

isdn switch-type {*switch-type* | **primary-qsig** | **basic-qsig**}

no isdn switch-type {*switch-type* | **primary-qsig** | **basic-qsig**}

Syntax Description

switch-type	Service provider switch type. See Table 2 for a list.
primary-qsig	PRI
basic-qsig	BRI

Defaults

The switch type defaults to none, which disables the switch type.

Command Modes

Global configuration mode or interface configuration mode.

Command History

Release	Modification
9.21	Introduced as a global command.
11.3 T	Introduced as an interface command.
12.0(2)T	**primary-qsig-slave** and **primary-qsig-master** keywords introduced for the Cisco MC3810.
12.0(7)K	**primary-qsig-slave** and **primary-qsig-master** keywords for the Cisco MC3810 are no longer supported. **primary-qsig** and **basic-qsig** keywords supported on the Cisco MC3810, Cisco 7200 VXR, 2600 and 3600 series routers.

Usage Guidelines

You can enter the **isdn switch-type** command to support QSIG at either the global configuration level or at the interface configuration level. For example, if you have a QSIG connection on one line as well as on the BRI or PRI port, you can configure the ISDN switch type in one of the following combinations:

- Set the global **isdn switch-type** command to support QSIG by entering either the **isdn-switch-type basic-qsig** command (BRI) or **isdn-switch-type primary-qsig** command (BRI); and set the interface **isdn-switch-type** command for the interfaces to a regular central office switch type, such as those shown in Table 2.
- Set the global **isdn switch-type** command to support the CO switch type (see Table 2), and set the interface **isdn switch-type** command for the interface to support QSIG.

- Configure the global **isdn switch-type** command to another setting (see Table 2); then, set the interface **isdn switch-type** command for **interface bri** to a BRI setting; set the interface **isdn switch-type** command for the serial interface to support QSIG.

Table 2 ISDN CO Switch Types

Country	ISDN Switch Type	Description
Australia	basic-ts013	Australian TS013 switches
Europe	basic-1tr6	German 1TR6 ISDN switches
	basic-nwnet3	Norwegian NET3 ISDN switches (phase 1)
	basic-net3	NET3 ISDN switches (UK and others)
	vn2	French VN2 ISDN switches
	vn3	French VN3 ISDN switches
Japan	ntt	Japanese NTT ISDN switches
New Zealand	basic-nznet3	New Zealand NET3 switches
North America	basic-5ess	Lucent Technologies basic rate switches
	basic-dms100	NT DMS-100 basic rate switches
	basic-ni1	National ISDN-1 switches

Examples

The following example shows the command configuration on a Cisco 3660 series router T1 PRI interface:

```
interface Serial5/0:23
 no ip address
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn protocol-emulate user
```

Related Commands

Command	Description
isdn protocol-emulate	Configures the interface to serve as either the QSIG slave or the QSIG master.

loss-plan

To specify the analog-to-digital gain offset for an analog FXO or FXS voice port, enter the **codec** dial-peer configuration command. Use the **no** form of this command to restore the default value.

loss-plan {**plan1** | **plan2** | **plan3** | **plan4** | **plan5** | **plan6** | **plan7** | **plan8** | **plan9**}
no loss-plan

Syntax Description

plan1	FXO: A-D gain = 0 dB, D-A gain = 0 dB FXS: A-D gain = –3 dB, D-A gain = –3 dB
plan2	FXO: A-D gain = 3 dB, D-A gain = 0 dB FXS: A-D gain = 0 dB, D-A gain = –3 dB
plan3	FXO: A-D gain = –3 dB, D-A gain = 0 dB FXS: Not applicable
plan4	FXO: A-D gain = –3 dB, D-A gain = –3 dB FXS: Not applicable
plan5	FXO: Not applicable FXS: A-D gain = –3 dB, D-A gain = –10 dB
plan6	FXO: Not applicable FXS: A-D gain = 0 dB, D-A gain = –7 dB
plan7	FXO: A-D gain = 7 dB, D-A gain = 0 dB FXS: A-D gain = 0 dB, D-A gain = –6 dB
plan8	FXO: A-D gain = 5 dB, D-A gain = –2 dB FXS: Not applicable
plan9	FXO: A-D gain = 6 dB, D-A gain = 0 dB FXS: Not applicable

Defaults

FXO: A-D gain = 0 dB, D-A gain = 0 dB (loss plan 1)

FXS: A-D gain = –3 dB, D-A gain = –3 dB (loss plan 1)

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	The following additional signal level choices were added: plan 3, plan 4, plan 8, and plan 9.

Usage Guidelines

This command sets the analog signal level difference (offset) between the analog voice port and the digital signal processor (DSP). Each loss plan specifies a level offset in both directions—from the analog voice port to the DSP (A-D) and from the DSP to the analog voice port (D-A).

Use this command to obtain the required levels of analog voice signals to and from the DSP.

This command is supported only on Cisco MC3810 series concentrators, on FXO and FXS analog voice ports.

Example

The following example configures FXO voice port 1/6 for a –3 dB offset from the voice port to the DSP and a 0 dB offset from the DSP to the voice port:

```
router(config)# voice-port 1/6
router(config-voiceport)# loss-plan plan3
```

The following example configures FXS voice port 1/1 for a 0 dB offset from the voice port to the DSP and a –7 dB offset from the DSP to the voice port:

```
router(config)# voice-port 1/1
router(config-voiceport)# loss-plan plan6
```

Related Commands

Command	Description
impedance	Specifies the terminating impedance of the voice port interface. Used on FXO voice ports in correcting input levels.
input gain	Specifies the gain applied by a voice port to the input signal from the PBX or other customer premises equipment.
output attenuation	Specifies the attenuation applied by a voice port to the output signal toward the PBX or other customer premises equipment.

num-exp

To define a complete telephone number for an extension, use the **num-exp** global configuration command. Use the **no** form of this command to cancel a configured number expansion.

num-exp *extension-number expanded-number*
no num-exp *extension-number*

Syntax Description

extension-number	Digit(s) defining an extension number to be expanded.
expanded-number	Digit(s) defining the expanded telephone number or destination pattern.

Defaults

No number expansion is configured.

Command Mode

Global configuration

Command History

Release	Modification
11.3(1)T	This command was first introduced on the Cisco 3600 platform.
12.0(3)T	This command was first supported on the Cisco AS5300 platform.
12.0(4)XL	This command was first supported on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

Use the **num-exp** global configuration command to expand a set of numbers (for example, an extension number) into a destination pattern. With this command, you can map specific extensions and expanded numbers together by explicitly defining each number, or you can define extensions and expanded numbers using variables. You can also use this command to convert seven-digit numbers to numbers containing less than seven digits.

Use a period (.) as a variable or wild card, representing a single number. Use a separate period for each number you want to represent with a wildcard; if you want to replace four numbers in an extension with wildcards, type in four periods.

Example

The following example specifies that extension number 55541 be expanded to 14085555541:

```
num-exp 55541 14085555541
```

The following example specifies that all five-digit extensions beginning with 5 be expanded to 1408555 . . . .

```
num-exp 5.... 1408555....
```

Related Commands

Command	Description
forward-digits	Specifies which digits to forward for voice calls.
prefix	Specifies a prefix for a dial peer.
dial-peer terminator	Change the character used as a terminator for variable length dialed numbers.

playout delay

To tune the playout buffer to accommodate packet jitter caused by switches in the WAN, use the **playout-delay** voice-port configuration command. Use the **no** form of this command to restore the default value.

playout-delay {**maximum** | **nominal**} *milliseconds*

no playout-delay {**maximum** | **nominal**}

Syntax Description

maximum	The delay time the DSP allows before starting to discard voice packets. The default is 160 milliseconds.
nominal	The initial (and minimum allowed) delay time the DSP inserts before playing out voice packets. The default is 80 milliseconds
milliseconds	Playout-delay value in milliseconds. The range for maximum playout delay is 40 to 320, and the range for nominal playout delay is 40 to 240.

Defaults

The default maximum delay is 160 milliseconds.
The default nominal delay is 80 milliseconds.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

If there is excessive break-up of voice due to jitter with the default playout delay settings, increase the delay times. If your network is small and jitter is minimal, decrease the delay times to reduce delay.

Examples

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice-port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# playout-delay nominal 80
router(config-voiceport)# playout-delay maximum 160
```

The following example configures a nominal playout delay of 80 milliseconds and a maximum playout delay of 160 milliseconds on voice-port 1/0/0 on the Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# playout-delay nominal 80
router(config-voiceport)# playout-delay maximum 160
```

Related Commands

Command	Description
vad	Enables voice activity detection.

pri-group

To specify a ISDN Primary Rate interface (PRI) on a channelized T1 or E1 controller, enter the **pri-group** controller configuration command. Enter the **no** form of this command removes the remove the ISDN-PRI configuration.

pri-group timeslots *timeslot-range*

no pri-group

Syntax Description

timeslot-range	*timeslot-list* is a single timeslot number, a single range of values. For T1, the allowable range is from 1 to 23. For E1, the allowable values are from 1 to 15.

Default

There is no ISDN-PRI group configured.

Command Mode

Controller configuration

Command History

Release	Modification
12.0(2)T	The command was introduced for the Cisco MC3810 multiservice access concentrator.
12.0(7)XK	The command was introduced for the Cisco 2600 and 3600 series with a different name and some keyword modifications.

Usage Guidelines

The **pri-group** command applies to the configuration of Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810 multiservice concentrator and the Cisco 2600 and 3600 series routers.

Before you enter the **pri-group** command, you must specify an ISDN-PRI switch type and an E1 or T1 controller. Only one pri group can be configured on a controller.

Example

The following example configures configures ISDN-PRI on all timeslots of controller E1 1 on a Cisco 2600 series router::

```
isdn switch-type primary-qsig-master
controller T1 1
pri-group timeslots 1-23
```

Related Command

Command	Description
isdn switch-type	To configure the Cisco 2600 series router PRI interface to support QSIG signalling, enter this command.

ring cadence

To specify the ring cadence for an FXS voice port, use the **ring cadence** voice-port configuration command. Use the **no** form of this command to restore the default value.

ring cadence {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse interval*]}
no ring cadence

Syntax Description

pattern01	2 seconds on, 4 seconds off
pattern02	1 second on, 4 seconds off
pattern03	1.5 seconds on, 3.5 seconds off
pattern04	1 second on, 2 seconds off
pattern05	1 second on, 5 seconds off
pattern06	1 second on, 3 seconds off
pattern07	0.8 second on, 3.2 seconds off
pattern08	1.5 seconds on, 3 seconds off
pattern09	1.2 seconds on, 3.7 seconds off
pattern09	1.2 seconds on, 4.7 seconds off
pattern11	0.4 second on, 0.2 second off, 0.4 second on, 2 seconds off
pattern12	0.4 second on, 0.2 second off, 0.4 second on, 2.6 seconds off
define	User-definable ring cadence pattern. Each number pair specifies one ring-pulse time and one ring-interval time. You must enter numbers in pairs, and you can enter 1 to 6 pairs. The second number in the last pair that you enter specifies the interval between rings.
pulse	A number (1 or 2 digits) specifying ring pulse (on) time in hundreds of milliseconds. The range is 1 to 50, for pulses of 100 ms to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.
interval	A number (1 or 2 digits) specifying ring interval (off) time in hundreds of milliseconds. The range is 1 to 50, for pulses of 100 to 5000 ms. For example: 1 = 100 ms; 10 = 1 s, 40 = 4 s.

Defaults

Ring cadence defaults to the pattern you specify with the **cptone** command.

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers, and the **patternXX** syntax was introduced.

Usage Guidelines

The **patternXX** keyword provides preset ring cadence patterns for use on any platform. The **define** keyword allows you to create a custom ring cadence. On the Cisco 2600 and 3600 series routers, only one or two pairs of digits can be entered under the **define** keyword.

Examples

The following example configures the ring cadence for 1 second on and 4 seconds off on voice port 1/1 on a Cisco MC3810:

```
router(config)# voice-port 1/1
router(config-voiceport)# ring cadence pattern02
```

The following example configures the ring cadence for 1 second on, 1 second off, 1 second on, and 5 seconds off on voice port 1/2 on a Cisco MC3810:

```
voice-port 1/2
router(config-voiceport)# ring cadence define 10 10 10 50
```

The following example configures the ring cadence for 1 second on and 2 seconds off on voice port 1/0/0 on a Cisco 2600 or 3600:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# ring cadence pattern04
```

Related Commands

Command	Description
ring frequency	Specifies the ring frequency for an FXS voice port.
cptone	Specifies the default tone, ring, and cadence settings according to country.

session target

To configure a network-specific address for a dial peer, use the **session target** dial-peer configuration command. Use the **no** form of this command to disable this feature.

Cisco MC3810 Voice over IP:

session target {ipv4:*destination-address* | dns:[s. | d. | e. | u.] *host-name* | loopback:rtp | loopback:compressed | loopback:uncompressed}
no session target

Cisco 3600 Voice over ATM:

session target *interface* **pvc** {*name* | *vpi*/*vci* | *vci*}
no session target

Syntax Description

For the Cisco MC3810 Voice over IP:

ipv4:*destination-address*	IP address of the dial peer.
dns:*host-name*	Indicates that the domain name server will be used to resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. (Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers: • **s.**—Indicates that the source destination pattern will be used as part of the domain name. • **d.**—Indicates that the destination number will be used as part of the domain name. • **e.**—Indicates that the digits in the called number will be reversed, periods will be added in-between each digit of the called number, and that this string will be used as part of the domain name. • **u.**—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.
loopback:rtp	Indicates that all voice data will be looped back to the originating source. This is applicable for VoIP peers.
loopback:compressed	Indicates that all voice data will be looped back in compressed mode to the originating source. This is applicable for POTS peers.
loopback:uncompressed	Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers.

For Cisco 3600 series Voice over ATM dial peers:

interface	Interface type and interface number on the router.
pvc	The specific ATM permanent virtual circuit (PVC) for this dial peer.
name	The PVC name.

vpi/vci	ATM network virtual path identifier (VPI) and virtual channel identifier (VCI) of this PVC. On the Cisco 3600, if you have the Multiport T1/E1 ATM network module with IMA installed, the valid range for *vpi* is 0-15, and the valid range for *vci* is 1-255. If you have the OC3 ATM Network Module installed, the valid range for *vpi* is 0-15, and the valid range for *vci* is 1-1023.
vci	ATM network virtual channel identifier (VCI) of this PVC.

Defaults

Enabled with no IP address or domain name defined.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1) T	This command was first introduced.
11.3(1) MA	Support was added for VoFR,VoATM and VoHDLC dial peers on the Cisco MC38110.
12.0(3) XG and 12.0(4)T	The *cid* option was added. Support was added for VoFR dial peers on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	Support was added for VoATM dial peers on the Cisco 3600 series routers. Support was added for VoIP dial peers on the Cisco MC3810. Support for VoHDLC on the Cisco MC3810 was removed in this release.

Usage Guidelines

This command applies to both the Cisco 3600 series and the Cisco MC3810.

Use the **session target** command to specify a network-specific address or domain name for a dial peer. Whether you select a network-specific address or a domain name depends on the session protocol you select.

The **session target loopback** command is used for testing the voice transmission path of a call. The loopback point will depend on the call origination and the loopback type selected.

The **session target dns** command can be used with or without the specified wildcards. Using the optional wildcards can reduce the number of VoIP dial peer session targets you need to configure if you have groups of numbers associated with a particular router.

Examples

The following example configures a session target using DNS for a host, “voice_router,” in the domain “cisco.com”:

```
dial-peer voice 10 voip
 session target dns:voice_router.cisco.com
```

The following example configures a session target using DNS, with the optional **u**. wildcard. In this example, the destination pattern has been configured to allow for any four-digit extension, beginning with the numbers 1310222. The optional wildcard **u.** indicates that the router will use the unmatched portion of the dialed number—in this case, the four-digit extension, to identify the dial peer. As in the previous example, the domain is “cisco.com.”

```
dial-peer voice 10 voip
 destination-pattern 1310222....
 session target dns:$u$.cisco.com
```

The following example configures a session target using dns, with the optional **d.** wildcard. In this example, the destination pattern has been configured for 13102221111. The optional wildcard **d**. indicates that the router will use the destination pattern to identify the dial peer in the “cisco.com” domain.

```
dial-peer voice 10 voip
 destination-pattern 13102221111
 session target dns:$d$.cisco.com
```

The following example configures a session target using DNS, with the optional **e.** wildcard. In this example, the destination pattern has been configured for 12345. The optional wildcard **e.** indicates that the router will reverse the digits in the destination pattern, add periods between the digits, and then use this reverse-exploded destination pattern to identify the dial peer in the “cisco.com” domain.

```
dial-peer voice 10 voip
 destination-pattern 12345
 session target dns:$e$.cisco.com
```

The following example configures a session target for Voice over ATM on the Cisco 3600 series. The session target is sent to ATM interface 0, and is for a PVC with a VPI/VCI of 1/100.

```
router(config)# dial-peer voice 12 voatm
router(config-dial-peer)# destination-pattern 13102221111
router(config-dial-peer)# session target atm1/0 pvc 1/100
```

Related Commands

Command	Description
called-number	Enables an incoming VoFR call leg to be bridged to the correct POTS call leg.
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
cptone	Specifies a regional tone, ring, and cadence setting for an analog voice port.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
dtmf-relay	Enables the DSP to generate FRF.11 Annex A frames for a dial peer.
preference	Indicates the preferred selection order of a dial peer within a hunt group.
session protocol	Establishes a VoFR protocol for calls between the local and the remote routers via the packet network.

show call active voice

To show the active call table, use the **show call active voice** EXEC command.

show call active voice

Syntax Description

This command has no arguments or keywords.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 and 3600.
12.0(3)XG	Support for VoFR was added.
12.0(4)T	This command was first supported on the Cisco 7200 series.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

This command applies to Voice over IP, Voice over Frame Relay, and Voice over ATM on the Cisco 2600, 3600, and MC3810 series.

Use this command to display the contents of the active call table, which shows all of the calls currently connected through the router. This command displays information about call times, dial peers, connections, Quality of Service, and other status and statistical information.

See Table 3 for a listing of the information types associated with this command.

Example

The following is sample output from the **show call active voice** command:

```
router# show call active voice
GENERIC: SetupTime=21072 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 ConnectTime=0 CallState=3 CallOrigin=2 ChargedUnits=0
InfoType=0 TransmitPackets=375413 TransmitBytes=7508260 ReceivePackets=377734
ReceiveBytes=7554680

VOIP: ConnectionId[0x19BDF910 0xAF500007 0x0 0x58ED0] RemoteIPAddress=17635075
RemoteUDPPort=16394 RoundTripDelay=0 SelectedQoS=0 SessionProtocol=1
SessionTarget= OnTimeRvPlayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
```

```
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPlayoutDelay=110
LoWaterPlayoutDelay=64 ReceiveDelay=94 VADEnable=0 CoderTypeRate=0

GENERIC: SetupTime=21072 Index=1 PeerAddress=+14085271001 PeerSubAddress=
PeerId=0 PeerIfIndex=0 LogicalIfIndex=5 ConnectTime=21115 CallState=4 CallOrigin=1
ChargedUnits=0 InfoType=1 TransmitPackets=377915 TransmitBytes=7558300
ReceivePackets=375594 ReceiveBytes=7511880

TELE: ConnectionId=[0x19BDF910 0xAF500007 0x0 0x58ED0] TxDuration=16640
VoiceTxDuration=16640 FaxTxDuration=0 CoderTypeRate=0 NoiseLevel=0 ACOMLevel=4
OutSignalLevel=-440 InSignalLevel=-440 InfoActivity=2 ERLLevel=227
SessionTarget=
```

Table 3 provides an alphabetical listing of the fields in this output and a description of each field.

Table 3 Show Call Active Voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CallState	Current state of the call.
CoderTypeRate	Negotiated coder transmit rate of voice/fax compression during the call.
ConnectionId	Global call identifier of a gateway call.
ConnectTime	Time at which the call was connected.
Dial-Peer	Tag of the dial peer transmitting this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. An example of such pullout is frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received on time from voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because voice data was lost or not received on time from voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during this call.
Index	Dial peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call.
InSignalLevel	Active input signal level from the telephony interface used by this call.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the call.

Table 3 Show Call Active Voice Field Descriptions (Continued)

Field	Description
NoiseLevel	Active noise level for the call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer.
PeerSubaddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone during the call.
SelectedQoS	Selected RSVP quality of service (QoS) for the call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetupTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted from this peer during the call.
TransmitPackets	Number of packets transmitted from this peer during the call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show call history voice

To display the call history table, use the **show call history voice** EXEC command.

show call history voice [**last** *number* | **brief**]

Syntax Description

last *number*	(Optional) Displays the last calls connected, where the number of calls displayed is defined by the argument *number.* Valid entries for the argument *number* are numbers from 1 to 2147483647.
brief	(Optional) Displays abbreviated call history information for each leg of a call.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600.
12.0(3)XG	Support for VoFR was added.
12.0(4)T	The **brief** keyword was added and the command was first supported on the Cisco 7200 series.
12.0(7)XK	Support for **brief** the keyword was added on the Cisco MC3810 platform.

Usage Guidelines

This command applies to all voice applications on the Cisco 2600, 3600, MC3810, and 7200 platforms.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all voice calls connected through this router in descending time order. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the keyword **last**, and define the number of calls to be displayed with the argument *number*. To display a shortened version of the call history table, use the keyword **brief**.

Example

The following is sample output from the **show call history voice** command for a VoFR call using the frf11-trunk session protocol:

```
router# show call history voice last 1
   GENERIC:
   SetupTime=8283963 ms
   Index=3149
   PeerAddress=3623110
   PeerSubAddress=
   PeerId=3400
   PeerIfIndex=18
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283963
   DisconectTime=8285463
   CallOrigin=1
   ChargedUnits=0
   InfoType=2
   TransmitPackets=94
   TransmitBytes=2751
   ReceivePackets=0
   ReceiveBytes=0

   VOFR:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   Subchannel=[Interface Serial0/0, DLCI 160, CID 10]
   SessionProtocol=frf11-trunk
   SessionTarget=Serial0/0 160 10
   CalledNumber=2603100
   VADEnable=ENABLED
   CoderTypeRate=g729r8
   CodecBytes=30
   SignalingType=cas
   DTMFRelay=DISABLED
   UseVoiceSequenceNumbers=DISABLED

   GENERIC:
   SetupTime=8283963 ms
   Index=3150
   PeerAddress=2601100
   PeerSubAddress=
   PeerId=1100
   PeerIfIndex=7
   LogicalIfIndex=0
   DisconnectCause=3F
   DisconnectText=service or option not available, unspecified
   ConnectTime=8283964
   DisconectTime=8285464
   CallOrigin=2
   ChargedUnits=0
   InfoType=2
   TransmitPackets=0
   TransmitBytes=-121
   ReceivePackets=94
   ReceiveBytes=2563
   TELE:
   ConnectionId=[0x3D4B232D 0x6A900627 0x0 0x4F00852]
   TxDuration=15000 ms
   VoiceTxDuration=2010 ms
   FaxTxDuration=0 ms
   CoderTypeRate=g729r8
   NoiseLevel=-68
```

```
ACOMLevel=20
SessionTarget=
```

The following is sample output from the **show call history voice** command for a VoIP call:

```
router# show call history voice
   GENERIC:
   SetupTime=20405
   Index=0
   PeerAddress=
   PeerSubAddress=
   PeerId=0
   PeerIfIndex=0
   LogicalIfIndex=0
   DisconnectCause=NORMAL
   DisconnectText=
   ConnectTime=0
   DisconectTime=20595
   CallOrigin=2
   ChargedUnits=0
   InfoType=0
   TransmitPackets=0
   TransmitBytes=0
   ReceivePackets=0
   ReceiveBytes=0

   VOIP:
   ConnectionId[0x19BDF910 0xAF500006 0x0 0x56590]
   RemoteIPAddress=17635075
   RemoteUDPPort=16392
   RoundTripDelay=0
   SelectedQoS=0
   SessionProtocol=1
   SessionTarget=
   OnTimeRvPlayout=0
   GapFillWithSilence=0
   GapFillWithPrediction=0
   GapFillWithInterpolation=0
   GapFillWithRedundancy=0
   HiWaterPlayoutDelay=0
   LoWaterPlayoutDelay=0
   ReceiveDelay=0
   VADEnable=0
   CoderTypeRate=0

   TELE: ConnectionId=[0x19BDF910 0xAF500006 0x0 0x56590]
   TxDuration=3030
   VoiceTxDuration=2700
   FaxTxDuration=0
   CoderTypeRate=0
   NoiseLevel=0
   ACOMLevel=0
   SessionTarget=
```

Table 4 provides an alphabetical listing of the fields in this output and a description of each field.

Table 4 Show Call History Voice Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
CallOrigin	Call origin; answer versus originate.
CoderTypeRate	Negotiated coder rate. This value specifies the transmit rate of voice/fax compression to its associated call leg for the call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time the call was connected.
DisconnectCause	Description explaining why the call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time the call was disconnected.
FaxDuration	Duration of fax transmitted from this peer to the voice gateway for this call. You can derive the Fax Utilization Rate by dividing this value by the TxDuration value.
GapFillWithSilence	Duration of voice signal replaced with silence because the voice data was lost or not received on time for this call.
GapFillWithPrediction	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithInterpolation	Duration of voice signal played out with signal synthesized from parameters or samples of data preceding and following in time because the voice data was lost or not received on time from the voice gateway for this call.
GapFillWithRedundancy	Duration of voice signal played out with signal synthesized from redundancy parameters available because the voice data was lost or not received on time from the voice gateway for this call.
HiWaterPlayoutDelay	High water mark Voice Playout FIFO Delay during the voice call.
Index	Index number identifying the voice-peer for this call.
InfoType	Information type for this call.
LogicalIfIndex	Index of the logical voice port for this call.
LoWaterPlayoutDelay	Low water mark Voice Playout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPlayout	Duration of voice playout from data received on time for this call. You can derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPlayout value to the GapFill values.
PeerAddress	Destination pattern or number to which this call is connected.
PeerId	ID value of the peer entry table to which this call was made.
PeerIfIndex	Index number of the logical interface through which this call was made. For ISDN media, this would be the index number of the B channel used for the call.
PeerSubAddress	Subaddress to which this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the decoder delay during the voice call.
ReceivePackets	Number of packets received by this peer during the call.

Table 4 Show Call History Voice Field Descriptions (Continued)

Field	Description
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP quality of service for the call.
SessionProtocol	Session protocol to be used for an Internet call between the local and remote router via the IP backbone.
SessionTarget	Session target of the peer used for the call.
SetUpTime	Value of the System UpTime when the call associated with this entry was started.
TransmitBytes	Number of bytes transmitted by this peer during the call.
TransmitPackets	Number of packets transmitted by this peer during the call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for the call.
VADEnable	Whether or not voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmitted from this peer to voice gateway for this call. You can derive the Voice Utilization Rate by dividing the VoiceTxDuration by the TxDuration value.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays the number expansions configured.
show voice port	Displays configuration information about a specific voice port.

show num-exp

To show the number expansions configured, use the **show num-exp** privileged EXEC command.

show num-exp [*dialed-number*]

Syntax Description

dialed-number	(Optional) Dialed number.

Command Mode

User EXEC and Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was first introduced on the Cisco 3600 platform.
12.0(3)T	This command was first supported on the Cisco AS5300 platform.
12.0(4)XL	This command was first supported on the Cisco AS5800 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 platform.

Usage Guidelines

This command applies to VoFR, VoATM, and Voice over IP on the Cisco 2600 series, 3600 series, and MC3810 platforms.

Use the **show num-exp** privileged EXEC command to display all of the number expansions configured for this router. To display number expansion for only one number, specify that number by using the *dialed-number* argument.

Example

The following is sample output from the **show num-exp** command:

```
router# show num-exp
Dest Digit Pattern = '0...'     Translation = '+14085270...'
Dest Digit Pattern = '1...'     Translation = '+14085271...'
Dest Digit Pattern = '3..'      Translation = '+140852703..'
Dest Digit Pattern = '4..'      Translation = '+140852804..'
Dest Digit Pattern = '5..'      Translation = '+140852805..'
Dest Digit Pattern = '6....'    Translation = '+1408526....'
Dest Digit Pattern = '7....'    Translation = '+1408527....'
Dest Digit Pattern = '8...'     Translation = '+14085288...'
```

Table 5 explains the fields in the sample output.

Table 5 Show Num-Exp Voice Field Descriptions

Field	Description
Dest Digit Pattern	Index number identifying the destination telephone number digit pattern.
Translation	Expanded destination telephone number digit pattern.

Related Commands

Command	Description
show call active voice	Displays the contents of the active call table.
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information for dial peers.
show voice port	Displays configuration information about a specific voice port.

show voice call

To show the call status for voice ports on the Cisco router or concentrator, use the **show voice call** EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

show voice call [*slot/subunit/port* | **summary**]

For the Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

show voice call [*slot/port:ds0-group* | **summary**]

For the Cisco MC3810 series with analog voice ports:

show voice call [*slot/port* | **summary**]

For the Cisco MC3810 series with digital voice ports:

show voice call [*slot:ds0-group* | **summary**]

Syntax Description

summary	(Optional) Show a summary of the call status, not the detailed report.
voice-port	(Optional) Displays the call status for a specified voice port.

Command Mode

User EXEC

Command History

Release	Modification
11.3 MA	This command was introduced for the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over IP.

This command shows call-processing and protocol state-machine information for a voice port, if it is available. It also shows information on the DSP channel associated with the voice port, if it is available. All real-time information in the DSP channel, such as jitter and buffer overrun for example, is queried to the DSP channel, and asynchronous responses are returned to the host side.

If no call is active on a voice port, the **show voice call summary** command displays only the VPM (shutdown) state. If a call is active on a voice port, the VTSPS state is shown. For an on-net call or a local call without local-bypass (not cross-connected), the CODEC and VAD fields are displayed. For an off-net call or a local call with local-bypass, the CODEC and VAD fields are not displayed.

CODEC and VAD are not displayed in the **show voice call** *port* command, because this information is in the summary display.

This command provides the status at these levels of the call handling module:

- Call processing state machine
- Protocol state machine

Sample Display

The following is a sample display from the **show voice call summary** command for voice ports on a Cisco MC3810, showing two local calls connected without local bypass:

```
router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ==================== ========================
0:17.18                                    *shutdown*
0:18.19 g729ar8   n  S_CONNECT            FXOLS_OFFHOOK
0:19.20                                   FXOLS_ONHOOK
0:20.21                                   FXOLS_ONHOOK
0:21.22                                   FXOLS_ONHOOK
0:22.23                                   FXOLS_ONHOOK
0:23.24                                   EM_ONHOOK
1/1                                       FXSLS_ONHOOK
1/2                                       FXSLS_ONHOOK
1/3                                       EM_ONHOOK
1/4                                       EM_ONHOOK
1/5                                       FXOLS_ONHOOK
1/6     g729ar8   n  S_CONNECT            FXOLS_CONNECT
```

The following is a sample display from the **show voice call summary** command for voice ports on a Cisco MC3810, showing two local calls connected with local bypass:

```
router# show voice call summary

PORT    CODEC    VAD VTSP STATE            VPM STATE
======= ======== === ==================== ========================
0:17.18                                    *shutdown*
0:18.19              S_CONNECT            FXOLS_OFFHOOK
0:19.20                                   FXOLS_ONHOOK
0:20.21                                   FXOLS_ONHOOK
0:21.22                                   FXOLS_ONHOOK
0:22.23                                   FXOLS_ONHOOK
0:23.24                                   EM_ONHOOK
1/1                                       FXSLS_ONHOOK
1/2                                       FXSLS_ONHOOK
1/3                                       EM_ONHOOK
1/4                                       EM_ONHOOK
1/5                                       FXOLS_ONHOOK
1/6                  S_CONNECT            FXOLS_CONNECT
```

The following is a sample display from the **show voice call** command for analog voice ports on a Cisco MC3810:

```
router# show voice call

1/1 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/2 vpm level 1 state = FXSLS_ONHOOK
vpm level 0 state = S_UP
1/3 is shutdown
1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
1/5 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP
1/6 vpm level 1 state = EM_ONHOOK
vpm level 0 state = S_UP
sys252#show voice call 1/4
1/4 vtsp level 0 state = S_CONNECT
vpm level 1 state = S_TRUNKED
vpm level 0 state = S_UP
router#***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms): 1445779863, Rx Delay Est(ms): 95
Rx Delay Lo Water Mark(ms): 95, Rx Delay Hi Water Mark(ms): 125
***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 10, Interpolate Conceal(ms): 0
Silence Conceal(ms): 0, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 20, Talkspurt Endpoint Detect Err: 0
***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 537, Rx Signal Pkts: 0, Rx Comfort Pkts: 0
Rx Dur(ms): 50304730, Rx Vox Dur(ms): 16090, Rx Fax Dur(ms): 0
Rx Non-seq Pkts: 0, Rx Bad Hdr Pkts: 0
Rx Early Pkts: 0, Rx Late Pkts: 0
***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 567, Tx Sig Pkts: 0, Tx Comfort Pkts: 0
Tx Dur(ms): 50304730, Tx Vox Dur(ms): 17010, Tx Fax Dur(ms): 0
***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -70.3 from PBX/Phone, Tx -68.0 to PBX/Phone
TDM ACOM Levels(dBm0): +2.0, TDM ERL Level(dBm0): +5.6
TDM Bgd Levels(dBm0): -71.4, with activity being voice
```

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsp	Shows the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.

show voice dsp

To show the configuration status for all configured DSP voice channels on the Cisco router or concentrator, use the **show voice dsp** EXEC command.

show voice dsp

Syntax Description

This command has no arguments or keywords.

Command Mode

User EXEC

Command History

Release	Modification
11.3 MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600, and the display format was modified.

Usage Guidelines

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over IP.

Use this command when abnormal behavior in the DSP voice channels occurs.

Sample Display

The following is a sample display from the **show voice dsp** command on a Cisco MC3810:

```
Router#show voice dsp
                                BOOT                          PAK
TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 001 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 002 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 003 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 004 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 005 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
C549 006 01 {high}    3.3 idle  idle      6  0                0       1365/1364
         02 {high}        idle                                0             0/0
```

Table 6 provides an alphabetical listing of the fields in this output and a description of each field.

Table 6 Show Voice DSP Field Descriptions

Field	Description
AI	Number of alarm indications received from the DSP, which may point to abnormality of DSP firmware.
BOOT STATE	Applicable to Cisco MC3810 only of dynamic reload of DSP is permitted.
CH	Voice channel number in DSP.
CODEC	Cisco MC3810 with HCM and Cisco 2600 and 3600 digital: If the DSP channel is in use, this indicates what codec it is using. If a DSP channel is not in use, this indicates the complexity level configured. Cisco MC3810 with VCM and Cisco 2600 and 3600 analog: Indicates what codec is loaded.
DSP	DSP number.
PAK ABORT	The number of DSP packets dropped due to DSP failure in picking up packets from the host.
PORT	The port number associated with the DSP channel. This is a fixed port number on the Cisco 2600 and 3600; this number may change with each new call on the Cisco MC3810.
RST	The number of DSP resets since the most recent **clear counters** entry.
STATE	The busy/idle state of the DSP channel.
TS	The backplane timeslot associated with this DSP channel. This is a fixed timeslot on the Cisco 2600 and 3600; this number may change with each new call on the Cisco MC3810.
TX/RX-PAK-CNT	An ordered pair of transmit and receive packet counts processed by the DSP since the previous **clear counters** command was entered.
TYPE	DSP hardware type.
VERS	Version and revision of DSP hardware, in X,Y format.

Related Commands

Command	Description
clear counters	Clears all the current interface counters from the interface.
show voice port	Displays configuration information about a specific voice port.

show voice port

To display configuration information about a specific voice port, use the **show voice port** EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

show voice port [*slot/subunit/port* | **summary**]

For the Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

show voice port [*slot/port:ds0-group* | **summary**]

For the Cisco MC3810 series with analog voice ports:

show voice port [*slot/port* | **summary**]

For the Cisco MC3810 series with digital voice ports:

show voice port [*slot:ds0-group* | **summary**]

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	(Optional) Displays information for the analog voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) *port* specifies an analog voice port number. Valid entries are 0 and 1.
summary	(Optional) Displays a summary of all voice ports.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	(Optional) Displays information for the digital voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.

For the Cisco MC3810 series with analog voice ports:

slot/port	(Optional) Displays information for the analog voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.
summary	(Optional) Displays a summary of all voice ports.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	(Optional) Displays information for the digital voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.
summary	(Optional) Displays a summary of all voice ports.

Command Mode

User EXEC

Command History

Release	Modification
11.3(1) T	This command was introduced.
12.0(5)XK and 12.0(7)T	The *ds0-group* argument was added for the Cisco 2600 and 3600 series routers.
12.0(7)XK	The **summary** keyword was added for the Cisco 2600 and 3600 series routers. The *ds0-group* argument was added for the Cisco MC3810.

Usage Guidelines

Use the **show voice port** privileged EXEC command to display configuration and voice-interface-card-specific information about a specific port.

Example

The following is sample output from the **show voice port summary** command for all voice ports on a Cisco MC3810 with an analog voice module (AVM):

```
router# show voice port summary
IN  OUT   ECHO
PORT SIG-TYPE     ADMIN OPER IN-STATUS OUT-STATUS GAIN ATTN CANCEL
1/1  fxs-ls       up    up   on-hook    idle        0    0    y
1/2  fxs-ls       up    up   on-hook    idle        0    0    y
1/3  e&m-wnk      up    up   idle       idle        0    0    y
1/4  e&m-wnk      up    up   idle       idle        0    0    y
1/5  fxo-ls       up    up   idle       on-hook     0    0    y
1/6  fxo-ls       up    up   idle       on-hook     0    0    y
```

The following is sample output from the **show voice port summary** command on a Cisco MC3810 with a digital voice module (DVM):

```
                                  IN      OUT
PORT   CH SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
0:17   18 fxo-ls     down  down idle     on-hook  y
0:18   19 fxo-ls     up    dorm idle     on-hook  y
0:19   20 fxo-ls     up    dorm idle     on-hook  y
0:20   21 fxo-ls     up    dorm idle     on-hook  y
0:21   22 fxo-ls     up    dorm idle     on-hook  y
0:22   23 fxo-ls     up    dorm idle     on-hook  y
0:23   24 e&m-imd    up    dorm idle     idle     y
1/1    -- fxs-ls     up    dorm on-hook  idle     y
1/2    -- fxs-ls     up    dorm on-hook  idle     y
1/3    -- e&m-imd    up    dorm idle     idle     y
1/4    -- e&m-imd    up    dorm idle     idle     y
1/5    -- fxo-ls     up    dorm idle     on-hook  y
1/6    -- fxo-ls     up    dorm idle     on-hook  y
Elements :
sys/voip/ccvpm               vpm_htsp.c (107)
sys/voip/ccvtsp              vtsp_core.c (167)
sys/voip/cli                 voiceport_action.c (58)
```

The following is sample output from the **show voice port** command for an E&M analog voice port on a Cisco 3600:

```
router# show voice port 1/0/0
E&M Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is E&M
 Operation State is unknown
 Administrative State is unknown
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is disabled
 Non Linear Processing is disabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is disabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 0 s
 Interdigit Time Out is set to 0 s
 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
```

```
 Voice card specific Info Follows:
 Signal Type is wink-start
 Operation Type is 2-wire
 Impedance is set to 600r Ohm
 E&M Type is unknown
 Dial Type is dtmf
 In Seizure is inactive
 Out Seizure is inactive
 Digit Duration Timing is set to 0 ms
 InterDigit Duration Timing is set to 0 ms
 Pulse Rate Timing is set to 0 pulses/second
 InterDigit Pulse Duration Timing is set to 0 ms
 Clear Wait Duration Timing is set to 0 ms
 Wink Wait Duration Timing is set to 0 ms
 Wink Duration Timing is set to 0 ms
 Delay Start Timing is set to 0 ms
 Delay Duration Timing is set to 0 ms
```

The following is sample output from the **show voice port** command for an FXS analog voice port on a Cisco 3600:

```
router# show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 The Interface Down Failure Cause is 0
 Alias is NULL
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to 0 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 16ms
 Connection Mode is Normal
 Connection Number is
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is inactive
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Hook Flash Duration Timing is set to 600 ms
```

The following is sample output from the **show voice port** command for an FXS analog voice port on a Cisco MC3810:

```
router# show voice port 1/2
Voice port 1/2 Slot is 1, Port is 2
 Type of VoicePort is FXS
 Operation State is UP
 Administrative State is UP
 No Interface Down Failure
```

```
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Coder Type is g729ar8
 Companding Type is u-law
 Voice Activity Detection is disabled
 Ringing Time Out is 180 s
 Wait Release Time Out is 30 s
 Nominal Playout Delay is 80 milliseconds
 Maximum Playout Delay is 160 milliseconds

 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing Voice
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Analog interface A-D gain offset = -3 dB
 Analog interface D-A gain offset = -3 dB
 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 20 Hz
 Hook Status is On Hook
 Ring Active Status is inactive
 Ring Ground Status is inactive
 Tip Ground Status is active
 Digit Duration Timing is set to 100 ms
 InterDigit Duration Timing is set to 100 ms
 Ring Cadence are [20 40] * 100 msec
 InterDigit Pulse Duration Timing is set to 500 ms
```

The following is sample output from the **show voice port** command for an E&M digital voice port on a Cisco 3600:

```
router# show voice port 1/0:1

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1
 Type of VoicePort is E&M
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Region Tone is set for US
```

Table 7 explains the fields in the sample output.

Table 7 Show Voice Port Field Descriptions

Field	Description
Administrative State	Administrative state of the voice port.
Alias	User-supplied alias for this voice port.
Analog interface A-D gain offset	Offset of the gain for analog-to-digital conversion.
Analog interface D-A gain offset	Offset of the gain for digital-to-analog conversion.
Clear Wait Duration Timing	Time of inactive seizure signal to declare call cleared.
Coder Type	Voice compression mode used.
Companding Type	Companding standard used to convert between analog and digital signals in PCM systems.
Connection Mode	Connection mode of the interface.
Connection Number	Full E.164 telephone number used to establish a connection with the trunk or PLAR mode.
Currently Processing	Type of call currently being processed: none, voice, or fax.
Delay Duration Timing	Maximum delay signal duration for delay dial signaling.
Delay Start Timing	Timing of generation of delayed start signal from detection of incoming seizure.
Description	Description of the voice port.
Dial Type	Out-dialing type of the voice port.
Digit Duration Timing	DTMF Digit duration in milliseconds.
E&M Type	Type of E&M interface.
Echo Cancel Coverage	Echo Cancel Coverage for this port.
Echo Cancellation	Whether or not echo cancellation is enabled for this port.
Hook Flash Duration Timing	Maximum length of hook flash signal.
Hook Status	Hook status of the FXO/FXS interface.
Impedance	Configured terminating impedance for the E&M interface.
In Gain	Amount of gain inserted at the receiver side of the interface.
In Seizure	Incoming seizure state of the E&M interface.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
InterDigit Duration Timing	DTMF interdigit duration in milliseconds.
InterDigit Pulse Duration Timing	Pulse dialing interdigit timing in milliseconds.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Maintenance Mode	Maintenance mode of the voice port.
Maximum Playout Delay	The amount of time before the Cisco MC3810 DSP starts to discard voice packets from the DSP buffer.
Music On Hold Threshold	Configured Music-On-Hold Threshold value for this interface.
Noise Regeneration	Whether or not background noise should be played to fill silent gaps if VAD is activated.
Nominal Playout Delay	The amount of time the Cisco MC3810 DSP waits before starting to play out the voice packets from the DSP buffer.

Table 7 Show Voice Port Field Descriptions (Continued)

Field	Description
Non-Linear Processing	Whether or not non-linear processing is enabled for this port.
Number of signaling protocol errors	Number of signaling protocol errors.
Operations State	Operation state of the port.
Operation Type	Operation of the E&M signal: two-wire or four-wire.
Out Attenuation	Amount of attenuation inserted at the transmit side of the interface.
Out Seizure	Outgoing seizure state of the E&M interface.
Port	Port number for this interface associated with the voice interface card.
Pulse Rate Timing	Pulse dialing rate in pulses per second (pps).
Region Tone	Configured regional tone for this interface.
Ring Active Status	Ring active indication.
Ring Cadence	Configured ring cadence for this interface.
Ring Frequency	Configured ring frequency for this interface.
Ring Ground Status	Ring ground indication.
Ringing Time Out	Ringing time out duration.
Signal Type	Type of signaling for a voice port: loop-start, ground-start, wink-start, immediate, and delay-dial.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Tip Ground Status	Tip ground indication.
Type of VoicePort	Type of voice port: FXO, FXS, and E&M.
The Interface Down Failure Cause	Text string describing why the interface is down,
Voice Activity Detection	Whether Voice Activity Detection is enabled or disabled.
Wait Release Time Out	The time a voice port stays in the call-failure state while the Cisco MC3810 sends a busy tone, reorder tone, or an out-of-service tone to the port.
Wink Duration Timing	Maximum wink duration for wink start signaling.
Wink Wait Duration Timing	Maximum wink wait duration for wink start signaling.

Related Commands

Command	Description
show voice call	Displays the call status for all voice ports on the Cisco router or concentrator.
show call history voice	Displays the call history table.
show dial-peer voice	Displays configuration information about dial peers.
show num-exp	Displays the number expansions that are configured.

show voice trunk-conditioning signaling

To display the status of trunk-conditioning signaling and timing parameters for a voice port, use the **show voice trunk-conditioning signaling** EXEC command.

show voice trunk-conditioning signaling [**summary** | *voice-port*]

Syntax Description

summary	(Optional) Show a summary of the status for all voice ports on the router or concentrator.
voice-port	(Optional) Show a detailed report for a specified voice port.

Command Mode

EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco MC3810 as **show voice permanent-call**.
12.0(7)XK	This command was renamed **show voice trunk-conditioning signaling**.

Usage Guidelines

This command displays the trunk signaling status for analog and digital voice ports on Cisco MC3810 concentrators.

Sample Display

The following is a sample display from the **show voice trunk-conditioning signaling summary** command for voice ports on a Cisco MC3810:

```
router# show voice trunk-conditioning signaling summary

1/1 is shutdown
1/4 is shutdown
1/5 :
TX INFO :slow-mode seq#= 25, sig pkt cnt= 40, last-ABCD=0000
hardware-state ACTIVE signal type is NorthamericanCAS signal path  is OPEN
RX INFO :slow-mode, sig pkt cnt= 36, prev-seq#= 25, last-ABCD=0000
```

The following is a sample display from the **show voice trunk-conditioning signaling** command for voice port 1/5 on a Cisco MC3810:

```
router# show voice trunk-conditioning signaling 1/5

1/5 :
TX INFO :slow-mode seq#= 25, sig pkt cnt= 42, last-ABCD=0000
hardware-state ACTIVE signal type is NorthamericanCAS
signal path  is OPEN
 0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
 0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
 0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
RX INFO :slow-mode, sig pkt cnt= 37
missing = 0, out of seq = 0, very late = 0
playout depth = 0 (ms), refill count = 1
prev-seq#= 25, last-ABCD=0000
trunk_down_timer = 4212 (ms), idle timer = 0 (sec),
tx_oos_timer = 0 (sec), rx_ais_duration = 0 (ms)
forced playout signal pattern = NONE
signaling playout history
0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
0000 0000 0000 0000 0000 0000 0000 0000 0000 0000
```

The following is a sample display from the **show voice trunk-conditioning signaling summary** command for voice ports on a Cisco 3600:

```
router# show voice trunk-conditioning signaling summary

2/0/0 is shutdown
2/0/1 is shutdown
3/0:0 8  is shutdown
3/0:1 1  is shutdown
3/0:2 2  is shutdown
3/0:3 3  is shutdown
3/0:5 5  is shutdown
3/0:6(6) :
 status :
3/0:7 7  is shutdown
3/1:0 8  is shutdown
3/1:1 1  is shutdown
3/1:3 3  is shutdown
3/1:5 5  is shutdown
3/1:7 7  is shutdown
```

The following is a sample display from the **show voice trunk-conditioning signaling** command for voice port 3/0:6 on a Cisco 3600:

```
router# show voice trunk-conditioning signaling 3/0:6

hardware-state ACTIVE signal type is NorthamericanCAS
status :
forced playout pattern = STOPPED
trunk_down_timer = 0, rx_ais_duration = 0, idle_timer = 0
```

Table 8 explains the fields in the sample output.

Table 8 Field Descriptions for show voice trunk-conditioning signaling Command

Field	Description
current timer	Time since last signaling packets were received.
forced playout pattern	Which forced playout pattern is sent to PBX: 0 = no forced playout pattern is sent 1 = receive IDLE playout pattern is sent 2 = receive OOS playout pattern is sent
hardware-state	Hardware state based on received IDLE pattern: IDLE = both sides are idle ACTIVE = at least one side is active
signal type	Signaling type used by lower level driver: Northamerica, MELCAS, transparent, or external.
idle timer	Time the hardware on both sides has been in idle state.
last-ABCD	Last received or transmitted signal bit pattern.
max inter-arrival time	Maximum interval between received signaling packets.
missing	Number of missed signal packets.
mode	Signaling packet generation frequency: fast mode = every 4 milliseconds slow mode = same frequency as keepalive timer
out of seq	Number of out-of-sequence signal packets.
playout depth	Number of packets in playout buffer.
prev-seq#	Sequence number of previous signaling packet.
refill count	Number of packets created to maintain nominal length of playout packet buffer.
rx_ais_duration	Time since receipt of AIS indicator.
seq#	Sequence number of signaling packet.
sig pkt cnt	Number of transmitted or received signaling packets.
signal path	Status of signaling path.
signaling playout history	Signaling bits received in last 60 milliseconds.
trunk_down_timer	Time since last signaling packets were received.
tx_oos_timer	Time since PBX started sending OOS signaling pattern defined by **signal pattern oos transmit**.
very late	Number of very late signaling packets.

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all VoIP and POTS dial peers configured on the router.
show voice dsp	Shows the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
show voice trunk-conditioning supervisory	Displays the status of trunk supervision and configuration parameters for voice ports.

show voice trunk-conditioning supervisory

To display the status of trunk supervision and configuration parameters for voice ports, use the **show voice trunk-conditioning supervisory** EXEC command.

show voice trunk-conditioning supervisory [**summary** | *voice-port*]

Syntax Description

summary	(Optional) Show a summary of the status for all voice ports on the router or concentrator.
voice-port	(Optional) Show a detailed report for a specified voice port.

Command Mode

EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco MC3810.

Usage Guidelines

This command displays the trunk supervision and configuration status for analog and digital voice ports.

Sample Display

The following is a sample display from the **show voice trunk-conditioning supervisory summary** command for voice ports on a Cisco MC3810:

```
router# show voice trunk-conditioning supervisory summary

1/1 is shutdown
1/4 is shutdown
1/5 : state : TRUNK_SC_CONNECT, voice : on , signal : on ,slave
```

The following is a sample display from the **show voice trunk-conditioning supervisory** command for voice port 1/5 on a Cisco MC3810:

```
router# show voice trunk-conditioning supervisory 1/5

1/5 : state : TRUNK_SC_CONNECT, voice : on, signal : on, slave
status: trunk connected
sequence oos : idle and oos
pattern :rx_idle = 0x0 rx_oos = 0xF tx_oos =  0xF
timing : idle = 0, restart = 0, standby = 0, timeout = 40
supp_all = 50, supp_voice = 0, keep_alive = 5
timer: oos_ais_timer = 0, timer = 0
```

The following is a sample display from the **show voice trunk-conditioning supervisory summary** command for voice ports on a Cisco 3600:

```
router# show voice trunk-conditioning supervisory summary

2/0/0 is shutdown
2/0/1 is shutdown
3/0:0 8  is shutdown
3/0:1 1  is shutdown
3/0:2 2  is shutdown
3/0:3 3  is shutdown
3/0:5 5  is shutdown
3/0:6(6) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/0:7(7) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:0(8) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:1(1) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:3(3) : state : TRUNK_SC_CONNECT, voice : on , signal : on ,master
3/1:5(5) is shutdown
3/1:7(7) is shutdown
```

The following is a sample display from the **show voice trunk-conditioning supervisory** command for voice port 3/0:6 on a Cisco 3600:

```
router# show voice trunk-conditioning supervisory 3/0:6

3/0:6(6) : state : TRUNK_SC_CONNECT, voice : on, signal : on, master
status: trunk connected
sequence oos : idle and oos
pattern :rx_idle = 0x0 rx_oos = 0xF
timing : idle = 0, restart = 0, standby = 0, timeout = 40
supp_all = 0, supp_voice = 0, keep_alive = 5
timer: oos_ais_timer = 0, timer = 0
```

Table 9 explains the fields in the sample output.

Table 9 Field Descriptions for show voice trunk-conditioning supervisory Command

Field	Description
keep_alive	Signaling packets periodically sent to the far end, even if there is no signal change. These signaling packets function as keepalive messages.
master	The voice port configured as **connection trunk xxxx**.
slave	The voice port configured as **connection trunk xxxx answer-mode**.
oos_ais_timer	Time since the signaling packet with AIS indicator was received.
pattern	4-bit signaling pattern.
restart	The restart timeout after far end is OOS.
rx-idle	The signaling bit pattern indicating that the far end is idle.
rx-oos	The signaling bit pattern sent to the PBX indicating that the network is OOS.
standby	The time before the slave side goes back to standby after far end goes OOS.
supp_all	The timeout before suppressing transmission of voice and signaling packets to the far end after detection of PBX OOS.
supp_voice	The timeout before suppressing transmission of voice packet to the far end after detection of PBX OOS.
timeout	The timeout for non-receipt of keepalive packets before the far end is considered to be OOS.
TRUNK_SC_CONNECT	Trunk conditioning supervisory component status.

Related Commands

Command	Description
show dial-peer voice	Displays the configuration for all dial peers configured on the router.
show voice dsp	Shows the current status of all DSP voice channels.
show voice port	Displays configuration information about a specific voice port.
show voice trunk-conditioning signaling	Displays the status of trunk-conditioning signaling and timing parameters for a voice port.

signal pattern

To configure the ABCD signaling bit pattern for Cisco trunks and FRF.11 trunks, use the **signal pattern** voice-class configuration command. Use the **no** form of this command to restore the default.

signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**} *bit-pattern*
no signal pattern {**idle receive** | **idle transmit** | **oos receive** | **oos transmit**}

Syntax Description

idle receive	Defines the signaling pattern for identifying an "idle" message from the network. and Defines the idle signaling pattern to be sent to the PBX if the network trunk is out of service and **signal sequence oos idle-only** or **signal sequence oos both** is configured.
idle transmit	Defines the signaling pattern for identifying an "idle" message from the PBX.
oos receive	Defines the OOS signaling pattern to be sent to the PBX if the network trunk is out of service and **signal sequence oos oos-only** or **signal sequence oos both** is configured.
oos transmit	Defines the signaling pattern for identifying an OOS message from the PBX.
bit-pattern	The ABCD signaling bit pattern. Values are 0000 to 1111.

Defaults

idle receive	For near-end E&M—0000 (for T1) or 0001 (for E1) For near-end FXO loop start—0101 For near-end FXO ground start—1111 For near-end FXS—0101 For near-end MELCAS—1101
idle transmit	For near-end E&M—0000 For near-end FXO—0101 For near-end FXS loop start—0101 For near-end FXS ground start—1111 For near-end MELCAS—1101

oos receive	For near-end E&M—1111 For near-end FXO loop start—1111 For near-end FXO ground start—0000 For near-end FXS loop start—1111 For near-end FXS ground start—0101 For near-end MELCAS—1111
oos transmit	No default signaling pattern is defined.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	Default signaling patterns were defined.

Usage Guidelines

This command defines the signaling patterns that are used to identify the idle and OOS states.

Before configuring the signaling pattern, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you will assign it to a dial peer.

To suppress voice packets whenever the transmit or receive trunk is in the idle state, use the **idle receive** and **idle transmit** commands in conjunction with the **signal timing idle suppress-voice** command.

To define the signaling bit patterns to be sent to the PBX when the trunk is out of service, use the **idle receive** and **oos receive** commands.

The **oos receive** pattern is the pattern sent to the PBX to indicate that the network trunk is out of service. The **oos receive** pattern is not used for pattern matching against the signaling packets received from the network.

To “busy out” a PBX if the network connection fails, set the **oos receive** pattern to match the seized state (busy); then set the **signal timing oos** timeout value. When the timeout expires and no signaling packets have been received, the router will send the **idle receive** and/or **oos receive** pattern to the PBX, depending on which pattern is specified by the **signal sequence oos** command.

Use the busy seized pattern only if the PBX does not have a pattern specifically intended to indicate an OOS state. If the PBX has a specific OOS pattern, use that pattern instead.

Examples

The following example restores default signaling bit patterns for the receive and transmit idle states:

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal timing idle suppress-voice
router(config-class)# no signal pattern idle receive
router(config-class)# no signal pattern idle transmit
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
```

The following example configures non-default signaling bit patterns for the receive and transmit idle states:

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal timing idle suppress-voice
router(config-class)# signal pattern idle receive 0101
router(config-class)# signal pattern idle transmit 0101
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
```

The following example restores default signaling bit patterns for the receive and transmit out-of-service states:

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal timing idle suppress-voice
router(config-class)# no signal pattern oos receive
router(config-class)# no signal pattern oos transmit
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
```

The following example configures non-default signaling bit patterns for the receive and transmit out-of-service states:

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal pattern oos receive 0001
router(config-class)# signal pattern oos transmit 0001
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial-peer type.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal sequence oos	Specifies which signaling pattern is sent to the PBX when the far-end keepalive message is lost or AIS is received from the far end.
signal timing idle suppress-voice	Specifies the length of time before the router stops sending voice packets after a trunk goes into the idle state.
signal timing oos restart	Specifies that a permanent voice connection be torn down and restarted after the trunk has been OOS for a specified time.

signal timing oos slave-standby	Specifies that a slave port return to its initial standby state after the trunk has been OOS for a specified time
signal timing oos suppress-all	Configures the router or concentrator to stop sending voice and signaling packets to the network if it detects a transmit OOS signaling pattern from the PBX for a specified time.
signal timing oos suppress-voice	Configures the router or concentrator to stop sending voice packets to the network if it detects a transmit OOS signaling pattern from the PBX for a specified time.
signal timing oos timeout	Changes the delay time between the loss of signaling packets from the network and the start time for the OOS state.
signal timing idle suppress-voice	Specifies the length of time before voice traffic is stopped after a trunk goes into the idle state.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal sequence oos

To specify which signaling pattern is sent to the PBX when the far-end keepalive message is lost or AIS is received from the far end, use the **signal sequence oos** voice-class configuration command. Use the no form of this command to restore the default value.

signal sequence oos {**no-action** | **idle-only** | **oos-only** | **both**}
no signal sequence oos

Syntax Description

no-action	No signaling pattern is sent.
idle-only	Only the idle signaling pattern is sent.
oos-only	Only the out-of-service (OOS) signaling pattern is sent.
both	Both idle and OOS signaling patterns are sent. This is the default value.

Defaults

Both idle and OOS signal patterns are sent.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco MC3810.

Usage Guidelines

Before configuring the idle or OOS signal patterns to be sent, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you will assign it to a dial peer.

Use this command to specify which signaling pattern(s) to send. Use the **signal pattern idle receive** or the **signal pattern oos receive** command to define the bit patterns of the signaling patterns if other than the defaults.

Examples

The following example defines voice class 10, sets the **signal sequence oos** to send only the idle signal pattern to the PBX, and applies the voice class configuration to VoFR dial peer 100.

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal sequence oos idle-only
router(config-class)# signal timing idle suppress-voice 5
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
router(config-dial-peer)# signal-type transparent
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial peer type.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal timing idle suppress-voice	Specifies the length of time before the router stops sending voice packets after a trunk goes into the idle state.
signal timing oos restart	Specifies that a permanent voice connection be torn down and restarted after the trunk has been OOS for a specified time.
signal timing oos slave-standby	Specifies that a slave port return to its initial standby state after the trunk has been OOS for a specified time
signal timing oos suppress-all	Configures the router or concentrator to stop sending voice and signaling packets to the network if it detects a OOS signaling pattern from the PBX for a specified time.
signal timing oos suppress-voice	Configures the router or concentrator to stop sending voice packets to the network if it detects a transmit OOS signaling pattern from the PBX for a specified time.
signal timing oos timeout	Changes the delay time between the loss of signaling packets from the network and the start time for the OOS state.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal timing idle suppress-voice

To specify the length of time before the router stops sending voice packets after a trunk goes into the idle state (no call in progress), use the **signal timing idle suppress-voice** voice-class configuration command. Use the **no** form of this command to restore the default value.

signal timing idle suppress-voice *seconds*
no signal timing idle suppress-voice

Syntax Description

seconds	Duration of the idle state in seconds before the transmission of voice packets is stopped. The range is 0 to 65535.

Defaults

The router or concentrator continues to send voice packets when the trunk is idle.

Command Modes

Voice-class configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was modified to simplify the configuration process.

Usage Guidelines

Before configuring the signal timing idle suppress-voice timer, you must use the **voice class permanent** command in global configuration mode to create a voice class for the Cisco trunk or FRF.11 trunk. After you finish defining the voice class, you will assign it to a dial peer.

This command is used when the **signal-type** command is set to **transparent** in the dial peer for the Cisco trunk or FRF.11 trunk connection. When the router or concentrator stops sending voice packets after the specified time, signaling packets continue to be sent.

To detect an idle trunk state, the router or concentrator monitors both transmit and receive signaling for the **idle transmit** and **idle receive** signaling patterns. These can be configured by the **signal pattern idle transmit** or **signal pattern idle receive** command, or they can be the defaults. The default **idle receive** pattern is the idle pattern of the local voice port. The default **idle transmit** pattern is the idle pattern of the far-end voice port.

Examples

The following example defines voice class 10, sets the idle detection time to 5 seconds, configures the trunk to use the default transmit and receive idle signal patterns, and applies the voice class configuration to VoFR dial peer 100.

```
router(config)# voice class permanent 10
router(config-class)# signal keepalive 3
router(config-class)# signal timing idle suppress-voice 5
router(config-class)# exit
router(config)# dial-peer voice 100 vofr
router(config-dial-peer)# voice-class permanent 10
router(config-dial-peer)# signal-type transparent
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies a dial peer type.
signal keepalive	Configures the keepalive signaling packet interval for Cisco trunks and FRF.11 trunks.
signal pattern	Configures the ABCD bit pattern for Cisco trunks and FRF.11 trunks.
signal sequence oos	Specifies which signaling pattern is sent to the PBX when the far-end keepalive message is lost or AIS is received from the far end.
signal timing idle suppress-voice	Specifies the length of time before the router stops sending voice packets after a trunk goes into the idle state.
signal timing oos restart	Specifies that a permanent voice connection be torn down and restarted after the trunk has been OOS for a specified time.
signal timing oos slave-standby	Specifies that a slave port return to its initial standby state after the trunk has been OOS for a specified time
signal timing oos suppress-all	Configures the router or concentrator to stop sending voice and signaling packets to the network if it detects a transmit OOS signaling pattern from the PBX for a specified time.
signal timing oos timeout	Changes the delay time between the loss of signaling packets from the network and the start time for the OOS state.
signal-type	Sets the signaling type to be used when connecting to a dial peer.
voice class permanent	Creates a voice class for a Cisco trunk or FRF.11 trunk.
voice-class permanent	Assigns a previously-configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.

signal-type

To set the signaling type to be used when connecting to a dial peer, use the **signal-type** command from dial-peer configuration mode. To return to the default signal-type, use the **no** form of this command.

signal-type {**cas** | **cept** | **ext-signal** | **transparent**}
no signal-type

Syntax Description

cas	North American EIA-464 Channel-Associated Signaling (robbed bit signaling). If the Digital T1 Packet Voice Trunk Network Module is installed, this option may not be available.
cept	Provides a basic E1 ABCD signaling protocol. Used primarily for E&M interfaces. When used with FXS/FXO interfaces, this protocol is equivalent to MELCAS.
ext-signal	External signaling. The DSP does not generate any signaling frames. Use this option when there is an external signaling channel, for example, CCS, or when you need to have a permanent “dumb” voice pipe.
transparent	On the Cisco MC3810, selecting this option produces different results depending on whether you are using a digital voice module (DVM) or an analog voice module (AVM). For a DVM: The ABCD signaling bits are copied from or transported through the T1/E1 interface “transparently” without modification or interpretation. This enables the MC3810 to handle arbitrary or unknown signaling protocols. For an AVM: It is not possible to provide “transparent” behavior because the Cisco MC3810 must interpret the signaling information in order to read and write the correct state to the analog hardware. This option is mapped to be equal to “cas.”

Defaults

cas

Command Mode

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced.
12.0(4)T	Support was added for the Cisco 7200 series routers.
12.0(7)XK	In previous releases, the **cept** and **transparent** options were only supported on the Cisco MC3810. Beginning in this release, these options are supported on the Cisco 2600, Cisco 3600 and Cisco 7200 routers.

Usage Guidelines

This command applies to VoFR and VoATM dial peers. It is used with permanent connections only (Cisco trunks and FRF.11 trunks), not with switched calls.

This command is used to inform the local telephony interface of the type of signaling it should expect to receive from the far-end dial peer. To turn signaling off at this dial peer, select the **ext-signal** option. If signaling is turned off and there are no external signaling channels, a “hot” line exists, enabling this dial peer to connect to anything at the far end.

When you connect an FXS to another FXS, or if you have anything other than an FXS/FXO or E&M/E&M pair, the appropriate signaling type on Cisco 2600 series and 3600 series routers is **ext-signal** (disabled).

If you have a digital E1 connection at the remote end that is running cept/MELCAS signaling and you then trunk that across to an analog port, you should make sure that you configure both ends for the **cept** signal-type.

If you have a T1 or E1 connection at both ends and the T1/E1 is running a signaling protocol that is neither EIA-464 or cept/MELCAS, you may want to configure the signal-type for the transparent option in order to pass through the signaling.

Examples

The following example shows how to disable signaling on a Cisco 2600 series or 3600 series router or on an MC3810 concentrator for VoFR dial peer 200, starting from global configuration mode:

```
router(config)# dial-peer voice 200 vofr
router(config-dial-peer)# signal-type ext-signal
router(config-dial-peer)#
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
connection	Specifies the connection mode for a voice port.
destination-pattern	Specifies the telephone number associated with a dial peer.
dtmf-relay	Enables the DSP to generate FRF.11 Annex A frames for a dial peer.
preference	Enables the preferred dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.
session protocol	Establishes the VoFR protocol for calls between local and remote routers.
session target	Specifies a network-specific address for a dial peer.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP.

test voice port detector

To test detector-related functions on a voice port, use the **test voice port detector** privileged EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

> **test voice port** *slot/subunit/port* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

For the Cisco 2600 and 3600 series with digital voice ports:

> **test voice port** *slot/port:ds0-group* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

For the Cisco MC3810 series with analog voice ports:

> **test voice port** *slot/port* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

For the Cisco MC3810 series with digital voice ports:

> **test voice port** *slot:ds0-group* **detector** {**m-lead** | **battery-reversal** | **ring** | **tip-ground** | **ring-ground** | **ring-trip**} {**on** | **off** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Tests the voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Tests the voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Tests the voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Tests the voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For all platforms:

m-lead	Forces the E&M m-lead detector to the specified state
loop	Forces the FXO loop detector to the specified state
battery-reversal	Forces the FXO battery-reversal detector to the specified state
ring	Forces the FXO ringing detector to the specified state
tip-ground	Forces the FXO tip-ground detector to the specified state
ring-ground	Forces the FXS ring-ground detector to the specified state
ring-trip	Forces the FXS ring-trip detector to the specified state
on	Forces the selected item to the **on** state
off	Forces the selected item to the **off** state
disable	Ends the forced state for the selected item

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **test voice port detector** privileged EXEC command to force a detector into specific states for testing. For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. When you are finished testing, be sure to enter the **disable** command to end the forced state. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces the tip-ground detector to the **off** state on an FXO voice port (1/3) on a Cisco MC3810, and ends any call in progress:

```
router# test voice port 1/3 detector tip-ground off
```

The following example ends the forced **off** state on an FXO voice port (1/3) on a Cisco MC3810:

```
router# test voice port 1/3 detector tip-ground disable
```

The following example forces the ring-trip detector to the **on** state on an FXS port (0/0/1) on a Cisco 3600 series router, and should start a call:

```
router# test voice port 0/0/1 detector ring-trip on
```

The following example ends the forced **on** state on an FXS port (0/0/1) on a Cisco 3600 series router:

```
router# test voice port 0/0/1 detector ring-trip disable
```

Related Commands

Command	Description
test voice port loopback	Performs loopback testing on a voice port.
test voice port inject-tone	Injects a test tone into a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port inject-tone

To inject a test tone into a voice port, use the **test voice port inject-tone** privileged EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

test voice port *slot/subunit/port* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

For the Cisco 2600 and 3600 series with digital voice ports:

test voice port *slot/port:ds0-group* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

For the Cisco MC3810 series with analog voice ports:

test voice port *slot/port* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

For the Cisco MC3810 series with digital voice ports:

test voice port *slot:ds0-group* **inject-tone** {**local** | **network**} {**1000hz** | **2000hz** | **200hz** | **3000hz** | **300hz** | **3200hz** | **3400hz** | **500hz** | **quiet** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Tests the voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Tests the voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Tests the voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Tests the voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For all platforms:

local	Directs the injected tone toward the local interface (near end)
network	Directs the injected tone toward the network (far end)
1000hz	Injects a 1-kilohertz test tone
2000hz	Injects a 2-kilohertz test tone
200hz	Injects a 200-hertz test tone
3000hz	Injects a 3-kilohertz test tone
300hz	Injects a 300-hertz test tone
3200hz	Injects a 3.2-kilohertz test tone
3400hz	Injects a 3.4-kilohertz test tone
500hz	Injects a 500-hertz test tone
quiet	Injects a quiet tone
disable	Ends test tone

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **test voice port inject-tone** privileged EXEC and to inject a test tone or to end a test tone. A call must be established on the voice port under test. When you are finished testing, be sure to enter the **disable** command to end the test tone. The **disable** keyword is available only if a test condition is already activated.

When you enter the **disable** command, you must enter a direction (either **network** or **local**); however, you can enter either direction, regardless of which direction you entered to inject the test tone.

Examples

The following example injects a 1-kilohertz test tone into voice port 1/1, directed toward the network (far end), on a Cisco MC3810:

```
router# test voice port 1/1 inject-tone network 1khz
```

The following example removes the test tone from port 0/0/1 on a Cisco 3600 series router:

```
router# test voice port 0/0/1 inject-tone network disable
```

or

```
router# test voice port 0/0/1 inject-tone local disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port
test voice port loopback	Performs loopback testing on a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port loopback

To perform loopback testing on a voice port, use the **test voice port loopback** privileged EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

test voice port *slot/subunit/port* **loopback** {**local** | **network** | **disable**}

For the Cisco 2600 and 3600 series with digital voice ports:

test voice port *slot/port:ds0-group* **loopback** {**local** | **network** | **disable**}

For the Cisco MC3810 series with analog voice ports:

test voice port *slot/port* **loopback** {**local** | **network** | **disable**}

For the Cisco MC3810 series with digital voice ports:

test voice port *slot:ds0-group* **loopback** {**local** | **network** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Tests the voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Tests the voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Tests the voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Tests the voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For all platforms:

local	Forces a loopback at the voice port toward the customer premises equipment (CPE)
network	Forces a loopback at the voice port toward network
disable	Ends forced loopback

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **test voice port loopback** privileged EXEC command to initiate or end a loopback at a voice port. A call must be established on the voice port under test. When you are finished testing, be sure to enter the **disable** command to end the forced loopback. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces a loopback toward the CPE on voice port 1/1 on a Cisco MC3810:

```
router# test voice port 1/1 loopback local
```

The following example ends a forced loopback on port 0/0/1 on a Cisco 3600 series router:

```
router# test voice port 0/0/1 loopback disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port.
test voice port inject-tone	Injects a test tone into a voice port.
test voice port relay	Tests relay-related functions on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port relay

To test relay-related functions on a voice port, use the **test voice port relay** privileged EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

test voice port *slot/subunit/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

For the Cisco 2600 and 3600 series with digital voice ports:

test voice port *slot/port:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

For the Cisco MC3810 series with analog voice ports:

test voice port *slot/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

For the Cisco MC3810 series with digital voice ports:

test voice port *slot:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Tests the voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Tests the voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Tests the voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Tests the voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For all platforms:

e-lead	Forces the E&M e-lead relay to the specified state
loop	Forces the FXO loop relay to the specified state
ring-ground	Forces the FXO ring-ground relay to the specified state
battery-reversal	Forces the FXO battery-reversal relay to the specified state
power-denial	Forces the FXS power-denial relay to the specified state
ring	Forces the FXS ringing relay to the specified state
tip-ground	Forces the FXS tip-ground relay to the specified state
on	Forces the selected item to the **on** state
off	Forces the selected item to the **off** state
disable	Ends the forced state for the selected item

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **test voice port relay** privileged EXEC command to force a relay into specific states for testing. For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. When you are finished testing, be sure to enter the **disable** command to end the forced state. The **disable** keyword is available only if a test condition is already activated.

Examples

The following example forces the E&M e-lead relay to the **on** state on port 0/0/1 on a Cisco 3600 series router:

```
router# test voice port 0/0/1 relay e-lead on
```

The following example ends a forced actuation of the battery-reversal relay on an FXS port (0/0/1) on a Cisco 3600 series router:

```
router# test voice port 0/0/1 relay battery-reversal disable
```

Related Commands

Command	Description
test voice port detector	Tests detector-related functions on a voice port
test voice port inject-tone	Injects a test tone into a voice port.
test voice port loopback	Performs loopback testing on a voice port.
test voice port switch	Forces a voice port into fax or voice mode.

test voice port switch

To force a voice port into fax mode, use the **test voice port switch** privileged EXEC command.

For the Cisco 2600 and 3600 series with analog voice ports:

test voice port *slot/subunit/port* **switch** {**fax** | **disable**}

For the Cisco 2600 and 3600 series with digital voice ports:

test voice port *slot/port:ds0-group* **switch** {**fax** | **disable**}

For the Cisco MC3810 series with analog voice ports:

test voice port *slot/port* **switch** {**fax** | **disable**}

For the Cisco MC3810 series with digital voice ports:

test voice port *slot:ds0-group* **switch** {**fax** | **disable**}

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Tests the voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Tests the voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Tests the voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Tests the voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For all platforms:

fax	Forces a switch to fax mode
disable	Ends fax mode; switches back to voice mode

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Use the **test voice port switch** privileged EXEC command to force a voice port into fax mode for testing. If no fax data is detected by the voice port, the voice port remains in fax mode for 30 seconds and then reverts automatically to voice mode. After you enter the **test voice port switch fax** command, you can use the **show voice call** or **show voice call summary** command to check whether the voice port is able to operate in fax mode.

The **disable** command ends the forced mode switch; however, the fax mode ends automatically after 30 seconds. The **disable** keyword is available only while the voice port is in fax mode.

Examples

The following example forces voice port 1/3 on a Cisco MC3810 into fax mode:

```
router# test voice port 1/3 switch fax
```

The following example returns voice port 0/0/1 on a Cisco 3600 series router to voice mode:

```
router# test voice port 0/0/1 switch disable
```

Related Commands

Command	Description
show voice call	Shows the call processing and protocol state-machine information for a voice port.
show voice call summary	Shows a summary of the call processing and protocol state-machine information for a voice port.

timeouts ringing

To configure the timeout value for ringing, use the **timeouts ringing** voice-port configuration command. Use the **no** form of this command to restore the default value.

timeouts ringing {*seconds* | **infinity**}
no timeouts ringing

Syntax Description

seconds	The duration in seconds that a voice port allows ringing to continue if a call is not answered. The range is 5 to 60000.
infinity	Ringing continues until the caller goes on hook.

Defaults

180 seconds

Command Mode

Voice-port configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

This command provides the capability to limit the length of time that a caller can continue ringing a telephone when there is no answer.

Examples

The following example configures voice port 1/1 on a Cisco MC3810 to allow ringing for 600 seconds:

```
router(config)# voice-port 1/1
router(config-voiceport)# timeouts ringing 600
```

The following example configures voice port 0/0/1 on a Cisco 3600 to allow ringing for 600 seconds:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timeouts ringing 600
```

Related Commands

Command	Description
timeouts initial	Configures the initial-digit timeout value for a voice port.
timeouts interdigit	Configures the interdigit timeout value for a voice port.

timeouts wait-release

To configure the delay timeout before the system starts the process for releasing voice ports, use the **timeouts wait-release** voice-port configuration command. Use the **no** form of this command to restore the default value.

timeouts wait-release {*seconds* | **infinity**}
no timeouts wait-release

Syntax Description

seconds	The duration in seconds that a voice port stays in the call-failure state while the Cisco router or concentrator sends a busy tone, reorder tone, or an out-of-service tone to the port. The range is 3 to 3600. The default is 30.
infinity	The voice port is never released as long as the call-failure state remains.

Defaults

30 seconds

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1) MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

Use this command to limit the time a voice port can be held in a call failure state. After the timeout, the release sequence is enabled.

You can also use this command for voice ports with FXS loop-start signaling, to specify the time allowed for a caller to hang up before the voice port goes into the parked state.

Examples

The following example configures voice port 1/1 on a Cisco MC3810 to stay in the call-failure state for 180 seconds while a busy tone, reorder tone, or out-of-service tone is sent to the voice port:

```
router(config)# voice-port 1/1
router(config-voiceport)# timeouts wait-release 180
```

The following example configures voice port 0/0/1 on a Cisco 3600 to stay in the call-failure state for 180 seconds while a busy tone, reorder tone, or out-of-service tone is sent to the voice port:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timeouts wait-release 180
```

Related Commands

Command	Description
timeouts initial	Configures the initial-digit timeout value for a voice port.
timeouts interdigit	Configures the interdigit timeout value for a voice port.

timing guard-out

To specify the guard-out duration of an FXO voice port, use the **timing guard-out** voice-port configuration command. Use the **no** form of this command to restore the default value.

timing guard-out *milliseconds*
no timing guard-out

Syntax Description

milliseconds	Duration in milliseconds of the guard-out period. The range is 300 to 3000. The default is 2000.

Defaults

2000 milliseconds

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1)MA5	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command applies to the Cisco 2600, 3600, and MC3810 platforms.

This command is supported on FXO voice ports only.

Examples

The following example configures the timing guard-out duration on a Cisco MC3810 voice port to 1000 milliseconds:

```
router(config)# voice-port 1/1
router(config-voiceport)# timing guard-out 1000
```

The following example configures the timing guard-out duration on a Cisco 2600 or 3600 voice port to 1000 milliseconds:

```
router(config)# voice-port 1/0/0
router(config-voiceport)# timing guard-out 1000
```

timing percentbreak

To specify the percentage of the break period for dialing pulses for a voice port, use the **timing percentbreak** voice-port configuration command. Use the **no** form of this command to reset the default value.

timing percentbreak *percent*
no timing percentbreak

Syntax Description

percent	Percentage of the break period for dialing pulses. Valid entries are numbers 20 to 80. The default is 50.

Defaults

50 percent

Command Mode

Voice-port configuration

Command History

Release	Modification
11.3(1) MA4	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 series routers.

Usage Guidelines

This command is supported on FXO and E&M voice ports only.

Examples

The following example configures the break period percentage on a Cisco MC3810 voice port to 30 percent:

```
router(config)# voice-port 1/1
router(config-voiceport)# timing percentbreak 30
```

The following example configures the break period percentage on a Cisco 2600 or 3600 voice port to 30 percent:

```
router(config)# voice-port 0/0/1
router(config-voiceport)# timing percentbreak 30
```

Related Commands

Command	Description
timing pulse	Configures the pulse dialing rate for a voice port.
timing pulse-interdigit	Configures the pulse inter-digit timing for a voice port.

vbr-rt

To configure the real-time variable bit rate (VBR) for Voice over ATM connections, use the **vbr-rt** ATM virtual circuit configuration command. Use the **no** form of this command to restore the default.

vbr-rt *peak-rate average-rate burst*
no vbr-rt

Syntax Description

peak-rate	The peak information rate (PIR) of the voice connection in kbps. The range is 56 to 10000.
average-rate	The average information rate (AIR) of the voice connection in kbps. The range is 1 to 56.
burst	Burst size in number of cells. The range is 0 to 65536.

Defaults

No vbr-rt settings are configured.

Command Mode

ATM virtual circuit configuration

Command History

Release	Modification
12.0	This command was introduced on the Cisco MC3810.
12.0(7)XK	Support for this command was extended to the Cisco 3600 series.

Usage Guidelines

The **vbr-rt** command configures traffic shaping between voice and data PVCs. Traffic shaping is required so that the carrier does not discard calls. To configure voice and data traffic shaping, you must configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will be carrying bursty traffic. The peak, average, and burst values are needed so the PVC can effectively handle the bandwidth for the number of voice calls. To calculate the *minimum* peak, average, and burst values for the number of voice calls, use the following calculations:

- Peak value: (2 x the maximum number of calls) x 16 kb
- Average value: (1 x the maximum number of calls) x 16 kb
- Burst value: (4 x the maximum number of calls)

Note When you configure data PVCs that will be traffic shaped with voice PVCs, use the aalsnap encapsulation and calculate the overhead as 1.13 times the voice rate.

Examples

The following example configures the traffic shaping rate for ATM PVC 20 on a Cisco 3600. In the example, the peak, average and burst rates are calculated based on a maximum of 20 calls on the PVC.

```
router(config-if)# pvc 20
router(config-if-atm-pvc)# encapsulation aal5mux voice
router(config-if-atm-pvc)# vbr-rt 640 320 80
```

Related Commands

Command	Description
encapsulation	Configures the ATM adaptation layer (AAL) and encapsulation type for an ATM PVC class

vofr

To enable Voice over Frame Relay (VoFR) on a specific DLCI and to configure specific subchannels on that DLCI, use the **vofr** command from Frame Relay DLCI configuration mode. Use the **no** form of the command to disable VoFR on a specific DLCI.

For switched calls:

vofr [data *cid***] [call-control [***cid***]]**
no vofr [data *cid***] [call-control [***cid***]]**

For switched calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK:

vofr [**cisco**]
no vofr [**cisco**]

For Cisco-trunk permanent calls:

vofr data *cid* **call-control** *cid*
no vofr data *cid* **call-control** *cid*

For Cisco-trunk permanent calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK:

vofr cisco
no vofr cisco

For FRF-11 trunk calls:

vofr [data *cid***] [call-control** *cid***]**
no vofr [data *cid***] [call-control** *cid***]**

Syntax Description

data	(Required for Cisco-trunk permanent calls. Optional for switched calls.) Used to select a subchannel (CID) for data other than the default subchannel, which is 4.
cid	(Optional) Specifies the subchannel to be used for data. Valid values are from 4 to 255; the default is 4. If **data** is specified, enter a valid CID.
call-control	(Optional) Used to specify that a subchannel will be reserved for call-control signaling. This option is not supported on the Cisco MC3810.
cid	(Optional) Specifies the subchannel to be used for call-control signaling. Valid values are from 4 to 255; the default is 5. If you specify **call-control** and you do not enter a CID, the default CID is used.
cisco	(Optional) Cisco proprietary voice encapsulation for VoFR with data is carried on CID 4 and call-control on CID 5. This option is required when configuring switched calls or Cisco trunks to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK. If configuring switched calls or Cisco trunks to Cisco MC3810 concentrators running Cisco IOS release 12.0(7)XK and later releases, do not use this option.

Defaults

Disabled

Command Mode

Frame Relay DLCI

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced.
12.0(7)XK	The use of the **cisco** option was modified. Beginning in this release, use the **cisco** option only when configuring connections to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK.

Usage Guidelines

Table 10 lists the different options of the **vofr** command and which combination of options is used.

Table 10 Combinations of the vofr Command

Type of Call	vofr Command Combination to Use
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** [*cid*]][1]
Switched call (user dialed or auto-ringdown) to a Cisco MC3810 running Cisco IOS releases before 12.0(7)XK	**vofr cisco**[2]
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	**vofr data** *cid* **call-control** *cid*
Cisco-trunk permanent call (private-line) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK	**vofr cisco**
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][3]

1. The recommended use of this command is **vofr data 4 call-control 5**.
2. This command consumes data CID 4 and call-control CID 5.
3. For FRF.111 trunk calls, the call-control option is not required. It is only required if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

If you select the “data” option, enter a numeric value to complete the command. If you select the “call-control” option, you do not enter a numeric value if you wish to accept the default call-control subchannel. See the following examples for clarification.

Usage Restrictions for Cisco IOS Releases Prior to 12.0(7)XK

This section describes restrictions for using the **vofr** command in releases prior to Cisco IOS Release 12.0(7)XK. Beginning in Cisco IOS Release 12.0(7)XK, these restrictions no longer apply.

When you use the **vofr** command without the **cisco** option, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command is entered without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel.

Table 11 describes special conditions and restrictions for the use of the **vofr** command on the Cisco MC3810.

Table 11 Using the vofr Command with the Cisco MC3810

Type of Call	Conditions and Restrictions
FRF.11 trunks	1. Do NOT use **cisco** option or **call-control** option. 2. Use **vofr** or **vofr data** *cid*.
Cisco trunks	1. Must use **vofr cisco**.
switched-vofr	1. Must use **vofr cisco**.

If you select the "data" option, enter a numeric value to complete the command. If you select the "call-control" option, you do not enter a numeric value if you wish to accept the default call-control subchannel. See the following examples for clarification.

When you use the **vofr** command on a Cisco MC3810 without the "cisco" option, switched calls are not permitted. You can only make permanent FRF.11-trunk calls.

Note It is not possible to configure the **call-control** option on a Cisco MC3810. If you configure this option, the setting is ignored.

Examples

The following example shows how to enable VoFR on Serial 1/1, DLCI 100 on a Cisco 2600 series, 3600 series, or 7200 series router or on an MC3810 concentrator, starting from global configuration mode:

```
router(config)# interface serial 1/1
router(config-if)# frame-relay interface-dlci 100
router(config-fr-dlci)# vofr
router(config-fr-dlci)#
```

The above example configures CID 4 for data; no call-control CID is defined.

To configure CID 4 for data and CID 5 for call-control (both defaults), enter the following command:

```
router(config-fr-dlci)# vofr call-control
router(config-fr-dlci)#
```

To configure CID10 for data and CID 15 for call-control, enter the following command:

```
router(config-fr-dlci)# vofr data 10 call-control 15
router(config-fr-dlci)#
```

To configure CID 4 for data and CID 15 for call-control, enter the following command:

```
router(config-fr-dlci)# vofr call-control 15
router(config-fr-dlci)#
```

To configure CID 10 for data and CID 5 for call-control, enter the following command:

```
router(config-fr-dlci)# vofr data 10 call-control
router(config-fr-dlci)#
```

To configure CID 10 for data with no call-control, enter the following command:

```
router(config-fr-dlci)# vofr data 10
router(config-fr-dlci)#
```

To configure a Cisco router or MC3810 for a VoFR application with an older release of the MC3810 (before Release 12.0(3)XG), enter the following command:

```
router(config-fr-dlci)# vofr cisco
router(config-fr-dlci)#
```

Related Commands

Command	Description
frame-relay interface-dlci	Assigns a data link connection identifier (DLCI) to a specified Frame Relay subinterface.
class	Assigns a VC class to a PVC.

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 platforms: • A value from 0 to 3 that identifies the physical slot in the chassis where the voice card is located. On Cisco MC3810 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 platforms, the slot corresponds to the physical slot in the chassis. On the Cisco MC3810, 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.

voice-encap

This command was added in Cisco IOS Release 11.3(1)MA on the Cisco MC3810 for Voice over HDLC. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

voice-group

This command was added in Cisco IOS Release 11.3(1)MA on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

voice local-bypass

To configure local calls to bypass the digital signal processor (DSP), use the **voice local-bypass** global configuration command. Use the **no** form of this command to direct local calls through the DSP.

voice local-bypass
no voice local-bypass

Syntax Description

This command has no arguments or keywords.

Defaults

Local calls bypass the DSP.

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced.

Usage Guidelines

Local calls (calls between voice ports on a router or concentrator) normally bypass the DSP to minimize use of system resources. Use the **no** form of this command if you need to direct local calls through the DSP. Input gain and output attenuation can be configured only if calls are directed through the DSP.

Examples

The following example configures a Cisco MC3810, 2600, or 3600 to pass local calls through the DSP:

```
router(config)# no voice local-bypass
```

Related Commands

Command	Description
input gain	Configures receive gain value for a voice port.
output attenuation	Configures transmit attenuation value for a voice port.

voice vad-time

To change the minimum silence detection time for voice activity detection (VAD), use the **voice vad-time** global configuration command. Use the **no** form of this command to restore the default value.

voice vad-time *milliseconds*
no voice vad-time

Syntax Description

milliseconds	The waiting period in milliseconds before silence detection and suppression of voice-packet transmission. The range is 250 to 65536. The default is 250.

Defaults

250 milliseconds

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600, 3600, and MC3810.

Usage Guidelines

This command affects all voice ports on a router or concentrator, but it does not affect calls already in progress.

You can use this command in transparent CCS applications in which you want VAD to activate when the voice channel is idle, but not during active calls. With a longer silence detection delay, VAD reacts to the silence of an idle voice channel, but not to pauses in conversation.

This command does not affect voice codecs that have ITU-standardized built-in VAD features—for example, G.729B, G.729AB, G.723.1A. The VAD behavior and parameters of these codecs are defined exclusively by the applicable ITU standard.

Examples

The following example configures a 20-second delay before VAD silence detection is enabled:

```
router(config)# voice vad-time 20000
```

Related Commands

Command	Description
vad (dial peer)	Enables voice activity detection on a network dial peer.

Debug Commands

This section documents new, modified and removed commands. All other commands used on these platforms are documented in the Cisco IOS Release 12.0 command reference publications.

- **debug ccfrf11 session**
- **debug ccswvoice voatm-debug**
- **debug ccswvoice voatm-session**
- **debug ccswvoice vofr-debug**
- **debug ccswvoice vofr-session**
- **debug vpm all**
- **debug vtsp all**
- **debug vtsp dsp**
- **debug vtsp error**
- **debug vtsp port**
- **debug vtsp session**
- **debug vtsp stats**
- **debug vtsp vofr subframe**

The following debug commands have been removed in Cisco IOS Release 12.0(7)XK:

- **debug voice all**
- **debug voice cp**
- **debug voice eecm**
- **debug voice protocol**
- **debug voice signaling**
- **debug voice vofr**

debug ccfrf11 session

To display the ccfrf11 function calls during call setup and teardown, use the **debug ccfrf11 session** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccfrf11 session
no debug ccfrf11 session

Syntax Description

This command has no keywords or arguments.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command to display debug information about the various FRF.11 VoFR service provider interface (SPI) functions. Note that this debug command does not display any information regarding the proprietary Cisco switched-VoFR SPI.

This debug is only useful when the session protocol is "frf11-trunk."

Examples

The following example shows sample output from the **debug ccfr11 session** command:

```
router# debug ccfrf11 session
INCOMING CALL SETUP (port setup for answer-mode):
*Mar  6 18:04:07.693:ccfrf11_process_timers:scb (0x60EB6040) timer (0x60EB6098) expired
*Mar  6 18:04:07.693:Setting accept_incoming to TRUE
*Mar  6 18:04:11.213:ccfrf11_incoming_request:peer tag 800:callingNumber=+2602100,
        calledNumber=+3622110
*Mar  6 18:04:11.213:ccfrf11_initialize_ccb:preffered_codec set(-1)(0)
*Mar  6 18:04:11.213:ccfrf11_evhandle_incoming_call_setup_request:calling +2602100,
        called +3622110 Incoming Tag 800
*Mar  6 18:04:11.217:ccfrf11_caps_ind:PeerTag = 800
*Mar  6 18:04:11.217:     codec(preferred) = 4, fax_rate = 2, vad = 2
*Mar  6 18:04:11.217:     cid = 30, config_bitmask = 0, codec_bytes = 20, signal_type=2
*Mar  6 18:04:11.217:     required_bandwidth 8192
*Mar  6 18:04:11.217:ccfrf11_caps_ind:Bandwidth reservation of 8192 bytes succeeded.
*Mar  6 18:04:11.221:ccfrf11_evhandle_call_connect:Entered
```

```
CALL SETUP (MASTER):
5d22h:ccfrf11_call_setup_request:Entered
5d22h:ccfrf11_evhandle_call_setup_request:Entered
5d22h:ccfrf11_initialize_ccb:preffered_codec set(-1)(0)
5d22h:ccfrf11_evhandle_call_setup_request:preffered_codec set(9)(24)
5d22h:ccfrf11_call_setup_trunk:subchannel linking successful
5d22h:ccfrf11_caps_ind:PeerTag = 810
5d22h:     codec(preferred) = 512, fax_rate = 2, vad = 2
5d22h:     cid = 30, config_bitmask = 1, codec_bytes = 24, signal_type=2
5d22h:     required_bandwidth 6500
5d22h:ccfrf11_caps_ind:Bandwidth reservation of 6500 bytes succeeded.

CALL TEARDOWN:
*Mar  6 18:09:14.805:ccfrf11_call_disconnect:peer tag 0
*Mar  6 18:09:14.805:ccfrf11_evhandle_call_disconnect:Entered
*Mar  6 18:09:14.805:ccfrf11_call_cleanup:freeccb 1, call_disconnected 1
*Mar  6 18:09:14.805:ccfrf11_call_cleanup:Setting accept_incoming to FALSE and starting
        incoming timer
*Mar  6 18:09:14.809:timer 2:(0x60EB6098)starts - delay (70000)
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Alive timer stopped
*Mar  6 18:09:14.809:timer 1:(0x60F64104) stops
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Generating Call record
*Mar  6 18:09:14.809:cause=10 tcause=10    cause_text="normal call clearing."
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Releasing 8192 bytes of reserved bandwidth
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:ccb 0x60F6404C, vdbPtr 0x610DB7A4
        freeccb_flag=1, call_disconnected_flag=1
```

Related Commands

Command	Description
debug ccswvoice vofr-debug	Displays the ccswvoice function calls during call setup and teardown.
debug ccswvoice vofr-session	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp session	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug ccswvoice voatm-debug

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice vooatm-debug** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice atm-debug
no debug ccswvoice atm-debug

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 3600 series.

Usage Guidelines

This command should be used when attempting to troubleshoot a VoATM call that uses the "cisco-switched" session protocol.This command provides the same information as the **debug ccswvoice voatm-session** command, but includes additional debugging information relating to the calls.

Examples

The following example shows sample output from the **debug ccswvoice voatm-debug** command:

```
router# debug ccswvoice voatm-debug

2w2d: ccswvoice: callID 529927 pvcid -1 cid -1 state NULL event O/G SETUP
2w2d: ccswvoice_out_callinit_setup: callID 529927 using pvcid 1 cid 15
2w2d: ccswvoice: callID 529927 pvcid 1 cid 15 state O/G INIT event I/C PROC
2w2d: ccswvoice: callID 529927 pvcid 1 cid 15 state O/G PROC event I/C
ALERTccfrf11_caps_ind: codec(preferred) = 1

2w2d: ccswvoice: callID 529927 pvcid 1 cid 15 state O/G ALERT event I/C CONN
2w2d: ccswvoice_bridge_drop: dropping bridge calls src 529927 dst 529926 pvcid 1 cid 15
state ACTIVE
2w2d: ccswvoice: callID 529927 pvcid 1 cid 15 state ACTIVE event O/G REL
2w2d: ccswvoice: callID 529927 pvcid 1 cid 15 state RELEASE event I/C RELCOMP
2w2d: ccswvoatm_store_call_history_entry: cause=10 tcause=10    cause_text=normal call
clearing.
```

Related Commands

Command	Description
debug ccswvoice vofr-session	Displays the ccswvoice function calls during call setup and teardown.

debug ccswvoice voatm-session

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice voatm-session** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice voatm-session
no debug ccswvoice voatm-session

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)MA	This command was introduced on the Cisco MC3810.
12.0(7)XK	This command was first supported on the Cisco 3600 series.

Usage Guidelines

Use this command to show the state transitions of the cisco-switched-voatm state machine as a call is processed. This command should be used when attempting to troubleshoot a VoATM call that uses the “cisco-switched” session protocol.

Examples

The following example shows sample output from the **debug ccswvoice voatm-session** command:

```
router# debug ccswvoice voatm-session

2w2d: ccswvoice: callID 529919 pvcid -1 cid -1 state NULL event O/G SETUP
2w2d: ccswvoice: callID 529919 pvcid 1 cid 11 state O/G INIT event I/C PROC
2w2d: ccswvoice: callID 529919 pvcid 1 cid 11 state O/G PROC event I/C ALERT
2w2d: ccswvoice: callID 529919 pvcid 1 cid 11 state O/G ALERT event I/C CONN
2w2d: ccswvoice: callID 529919 pvcid 1 cid 11 state ACTIVE event O/G REL
2w2d: ccswvoice: callID 529919 pvcid 1 cid 11 state RELEASE event I/C RELCOMP
```

Related Commands

Command	Description
debug ccswvoice vofr-debug	Displays the ccswvoice function calls during call setup and teardown.

debug ccswvoice vofr-debug

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice vofr-debug** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice vofr-debug
no debug ccswvoice vofr-debug

Syntax Description

This command has no arguments or keywords.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command when troubleshooting a VoFR call that uses the "cisco-switched" session protocol. This command provides the same information as the **debug ccswvoice vofr-session** command, but includes additional debugging information relating to the calls.

Examples

The following example shows sample output from the **debug ccswvoice vofr-debug** command:

```
router# debug ccswvoice vofr-debug
CALL TEARDOWN:
3640_vofr(config-voiceport)#
*Mar  1 03:02:08.719:ccswvofr_bridge_drop:dropping bridge calls src 17 dst 16 dlci 100
      cid 9 state ACTIVE
*Mar  1 03:02:08.727:ccswvofr:callID 17 dlci 100 cid 9 state ACTIVE event O/G REL
*Mar  1 03:02:08.735:ccswvofr:callID 17 dlci 100 cid 9 state RELEASE event I/C RELCOMP
*Mar  1 03:02:08.735:ccswvofr_store_call_history_entry:cause=22 tcause=22
      cause_text=no circuit.
3640_vofr(config-voiceport)#

CALL SETUP (outgoing):
*Mar  1 03:03:22.651:ccswvofr:callID 23 dlci -1 cid -1 state NULL event O/G SETUP
*Mar  1 03:03:22.651:ccswvofr_out_callinit_setup:callID 23 using dlci 100 cid 10
*Mar  1 03:03:22.659:ccswvofr:callID 23 dlci 100 cid 10 state O/G INIT event I/C PROC
*Mar  1 03:03:22.667:ccswvofr:callID 23 dlci 100 cid 10 state O/G PROC event I/C CONN
ccfrf11_caps_ind:codec(preferred) = 0
```

Related Commands

Command	Description
debug ccfrf11 session	Displays the ccfrf11 function calls during call setup and teardown.
debug ccswvoice vofr-session	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp session	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug ccswvoice vofr-session

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice vofr-session** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice vofr-session
no debug ccswvoice vofr-session

Syntax Description

This command has no arguments or keywords.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command to show the state transitions of the cisco-switched-vofr state machine as a call is processed, and when attempting to troubleshoot a VoFR call that uses the "cisco-switched" session protocol.

Examples

The following example shows sample output from the **debug ccswvoice vofr-session** command:

```
router# debug ccswvoice vofr-session
CALL TEARDOWN:
3640_vofr(config-voiceport)#
*Mar  1 02:58:13.203:ccswvofr:callID 14 dlci 100 cid 8 state ACTIVE event O/G REL
*Mar  1 02:58:13.215:ccswvofr:callID 14 dlci 100 cid 8 state RELEASE event I/C RELCOMP
3640_vofr(config-voiceport)#

CALL SETUP (outgoing):
*Mar  1 02:59:46.551:ccswvofr:callID 17 dlci -1 cid -1 state NULL event O/G SETUP
*Mar  1 02:59:46.559:ccswvofr:callID 17 dlci 100 cid 9 state O/G INIT event I/C PROC
*Mar  1 02:59:46.567:ccswvofr:callID 17 dlci 100 cid 9 state O/G PROC event I/C CONN
3640_vofr(config-voiceport)#
```

Related Commands

Command	Description
debug ccfrf11 session	Displays the ccfrf11 function calls during call setup and teardown.
debug ccswvoice vofr-debug	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp vofr subframe	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug voice all

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug voice cp

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug voice eecm

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug voice protocol

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug voice signaling

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug voice vofr

This command is no longer supported in Cisco IOS Release 12.0(7)XK.

debug vpm all

Use the **debug vpm all** command to enable all voice port module (VPM) debugging. Use the **no** form of this command to disable all VPM debugging.

debug vpm all
no debug vpm all

Syntax Description

This command has no arguments or keywords.

Defaults

VPM debugging is not enabled.

Command History

Release	Modification
11.3(1)T	This command was introduced for the Cisco 3600 series.
12.0(7)XK	This command was updated for the Cisco 2600, 3600, and MC3810.

Usage Guidelines

Use the **debug vpm all** command to enable the complete set of VPM debugging commands: **debug vpm dsp**, **debug vpm error**, **debug vpm port**, **debug vpm spi**, and **debug vpm trunk_sc**.

Execution of **no debug all** will turn off all port level debugging. It is usually a good idea to turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Examples

For sample outputs, refer to the individual commands in this chapter.

Related Commands

Command	Description
debug vpm port	Limits the **debug vpm all** command to a specified port.
show debug	Shows which debug commands are enabled.
debug vtsp all	Enables the display of trunk conditioning supervisory component trace information.

debug vpm trunk_sc

Use the **debug vpm trunk_sc** privileged EXEC command to enable the display of trunk conditioning supervisory component trace information. The **no** form of this command disables the display of this information.

debug vpm trunk_sc
no debug vpm trunk_sc

Syntax Description

This command has no arguments or keywords.

Defaults

Trunk conditioning supervisory component trace information is not displayed.

Command History

Release	Modification
12.0(7)XK	This command was introduced on the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

Use the **debug vpm port** command with the *slot-number/subunit-number/port* argument to limit the **debug vpm trunk_sc** debug output to a particular port. If you do not use the **debug vpm port** command, the **debug vpm trunk_sc** displays output for all ports.

Execution of **no debug all** will turn off all port level debugging. It is usually a good idea to turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Examples

The following example shows **debug vpm trunk_sc** messages for port 1/0/0 on a Cisco 2600 or 3600 series router:

```
router# debug vpm trunk_sc
router# debug vpm port 1/0/0
```

The following example shows **debug vpm trunk_sc** messages for port 1/1 on a Cisco MC3810:

```
router# debug vpm trunk_sc
router# debug vpm port 1/1
```

The following example turns off **debug vpm trunk_sc** debugging messages:

```
router# no debug vpm trunk_sc
```

Related Commands

Command	Description
debug vpm all	Enables all VPM debugging
debug vpm port	Limits the **debug vpm trunk_sc** command to a specified port.
show debug	Shows which debug commands are enabled.

debug vtsp all

Use the **debug vtsp all** command to show debugging information for all of the **debug vtsp** commands. Use the **no** form of this command to disable debugging output.

debug vtsp all
no debug vtsp all

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp all** command enables the following debug vtsp commands: **debug vtsp session**, **debug vtsp error**, and **debug vtsp dsp**. For more information or sample output, refer to the individual commands in this chapter.

Execution of **no debug vtsp all** will turn off all VTSP-level debugging. You should turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Related Commands

Command	Description
show debug	Shows which debug commands are enabled.
debug vtsp port	Limits vtsp debug output to a specific voice port.

debug vtsp dsp

Use the **debug vtsp dsp** command to show messages from the DSP to the access server. Use the **no** form of this command to disable debugging output.

debug vtsp dsp
no debug vtsp dsp

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp dsp is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600, and MC3810 platforms.

Usage Guidelines

ON AS5300 ACCESS SERVERS

The **debug vtsp dsp** command shows messages from the DSP on the VFC to the router; this command can be useful if you suspect that the VFC is not functional. It is a simple way to check if the VFC is responding to off-hook indications.

ON 2600, 3600, MC3810 PLATFORMS

The **debug vtsp dsp** command shows messages from the DSP to the router.

Sample Display

The following example shows the collection of DTMF digits from the DSP on a Cisco AS5300 access server.

```
*Nov 30 00:44:34.491: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=3
*Nov 30 00:44:36.267: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=1
*Nov 30 00:44:36.571: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=0
*Nov 30 00:44:36.711: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=0
*Nov 30 00:44:37.147: vtsp_process_dsp_message: MSG_TX_DTMF_DIGIT: digit=2
```

Related Commands

Command	Description
debug voice eecm	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp error

Use the **debug vtsp error** command to display processing errors in the voice telephony service provider. Use the **no** form of this command to disable vtsp error debugging.

debug vtsp error
no debug vtsp error

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp errors is not enabled.

Command History

Release	Modification
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp error** command can be used to check for mismatches in interface capabilities.

Sample Display

The following example shows sample output from the **debug vtsp error** command, in which a dialed number is not reachable because it is not configured.

```
router#deb vtsp error
Voice telephony call control error debugging is on

router#
*Mar  1 00:21:48.698:cc_api_call_setup_ind (vdbPtr=0x1575AB0,
callInfo={called=,called_oct3=0x81,calling=9999,calling_oct3=0x0,called_oct3a=0x0,
       fdest=0 peer_tag=1},callID=0x15896A4)
*Mar  1 00:21:48.698:cc_api_call_setup_ind type 3 , prot 0
*Mar  1 00:21:48.706:cc_process_call_setup_ind (event=0x16AD0E0) handed call to app
"SESSION"
*Mar  1 00:21:48.706:sess_appl:ev(23=CC_EV_CALL_SETUP_IND), cid(15), disp(0)
*Mar  1 00:21:48.706:sess_appl:ev(SSA_EV_CALL_SETUP_IND), cid(15), disp(0)
*Mar  1 00:21:48.706:ccCallSetContext (callID=0xF, context=0x1632898)
*Mar  1 00:21:48.706:ccCallSetupAck (callID=0xF)
*Mar  1 00:21:48.706:ccGenerateTone (callID=0xF tone=8)
*Mar  1 00:21:49.710:cc_api_call_digit_begin (vdbPtr=0x1575AB0, callID=0xF, digit=5,
flags=0x1, timestamp=0xB1AE6BC4, expiration=0x0)
*Mar  1 00:21:49.710:sess_appl:ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(15), disp(0)
*Mar  1 00:21:49.710:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_DIGIT_BEGIN)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.714:ssaIgnore cid(15), st(SSA_CS_MAPPING),oldst(0), ev(10)
*Mar  1 00:21:49.778:cc_api_call_digit (vdbPtr=0x1575AB0, callID=0xF, digit=5,
duration=4165,tag 0, callparty 0 )
*Mar  1 00:21:49.778:sess_appl:ev(9=CC_EV_CALL_DIGIT), cid(15), disp(0)
*Mar  1 00:21:49.778:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_CALL_DIGIT)
```

```
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.782:ssaDigit
*Mar  1 00:21:49.782:ssaDigit, callinfo , digit 5, tag 0,callparty 0
*Mar  1 00:21:49.782:ssaDigit, calling 9999,result 1
*Mar  1 00:21:49.915:cc_api_call_digit_begin (vdbPtr=0x1575AB0, callID=0xF, digit=5,
flags=0x1, timestamp=0xB1AF6B6C, expiration=0x0)
*Mar  1 00:21:49.915:sess_appl:ev(10=CC_EV_CALL_DIGIT_BEGIN), cid(15), disp(0)
*Mar  1 00:21:49.915:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_DIGIT_BEGIN)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:49.915:ssaIgnore cid(15), st(SSA_CS_MAPPING),oldst(0), ev(10)
*Mar  1 00:21:49.999:cc_api_call_digit (vdbPtr=0x1575AB0, callID=0xF, digit=5,
duration=95,tag 0, callparty 0 )
*Mar  1 00:21:49.999:sess_appl:ev(9=CC_EV_CALL_DIGIT), cid(15), disp(0)
*Mar  1 00:21:50.003:cid(15)st(SSA_CS_MAPPING)ev(SSA_EV_CALL_DIGIT)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
*Mar  1 00:21:50.003:ssaDigit
*Mar  1 00:21:50.003:ssaDigit, callinfo , digit 55, tag 0,callparty 0
*Mar  1 00:21:50.003:ssaDigit, calling 9999,result -1
*Mar  1 00:21:50.003:ccCallDisconnect (callID=0xF, cause=0x1C tag=0x0)
*Mar  1 00:21:50.003:ccCallDisconnect (callID=0xF, cause=0x1C tag=0x0)
*Mar  1 00:21:50.007:vtsp_process_event():prev_state = 0.4 ,
 state = S_WAIT_RELEASE_NC, event = E_CC_DISCONNECT
 Invalid FSM  Input on channel 1/1:15
*Mar  1 00:21:52.927:vtsp_process_event():prev_state = 0.7 ,
 state = S_WAIT_RELEASE_RESP, event = E_TSP_CALL_FEATURE_IND
 Invalid FSM  Input on channel 1/1:15
*Mar  1 00:21:52.931:cc_api_call_disconnect_done(vdbPtr=0x1575AB0, callID=0xF, disp=0,
tag=0x0)
*Mar  1 00:21:52.931:sess_appl:ev(13=CC_EV_CALL_DISCONNECT_DONE), cid(15), disp(0)
*Mar  1 00:21:52.931:cid(15)st(SSA_CS_DISCONNECTING)ev(SSA_EV_CALL_DISCONNECT_DONE)
oldst(SSA_CS_MAPPING)cfid(-1)csize(0)in(1)fDest(0)
```

Related Commands

Command	Description
debug voice eecm	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp port

To observe the behavior of the VTSP state machine on a specific voice port, use the **debug vtsp port** command. Use the **no** form of the command to turn off the debug function.

For Cisco 2600 and 3600 series with analog voice ports:

debug vtsp port *slot/subunit/port*
no debug vtsp port *slot/subunit/port*

For Cisco 2600 and 3600 series with digital voice ports (with T1 packet voice trunk network modules):

debug vtsp port *slot/port:ds0-group*
no debug vtsp port *slot/port:ds0-group*

For Cisco MC3810 series with analog voice ports:

debug vtsp port *slot/port*
no debug vtsp port *slot/port*

For Cisco MC3810 series with digital voice ports:

debug vtsp port *slot/port*
no debug vtsp port *slot/ds0-group*

Syntax Description

For the Cisco 2600 and 3600 series with analog voice ports:

slot/subunit/port	Debugs the analog voice port you specify with the *slot/subunit/port* designation. *slot* specifies a router slot in which a voice network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *subunit* specifies a voice interface card (VIC) where the voice port is located. Valid entries are 0 and 1. (The VIC fits into the voice network module.) *port* specifies an analog voice port number. Valid entries are 0 and 1.

For the Cisco 2600 and 3600 series with digital voice ports:

slot/port:ds0-group	Debugs the digital voice port you specify with the *slot/port:ds0-group* designation. *slot* specifies a router slot in which the packet voice trunk network module (NM) is installed. Valid entries are router slot numbers for the particular platform. *port* specifies a T1 or E1 physical port in the voice WAN interface card (VWIC). Valid entries are 0 and 1. (One VWIC fits in an NM.) *ds0-group* specifies a T1 or E1 logical port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

For the Cisco MC3810 series with analog voice ports:

slot/port	Debugs the analog voice port you specify with the *slot/port* designation. *slot* is the physical slot in which the analog voice module (AVM) is installed. The *slot* is always 1 for analog voice ports in the Cisco MC3810. *port* specifies an analog voice port number. Valid entries are 1 to 6.

For the Cisco MC3810 series with digital voice ports:

slot:ds0-group	Debugs the digital voice port you specify with the *slot:ds0-group* designation. *slot* specifies the module (and controller). Valid entries are 0 for the MFT (controller 0) and 1 for the DVM (controller 1). *ds0-group* specifies a T1 or E1 logical voice port number. Valid entries are 0 to 23 for T1 and 0 to 30 for E1.

Defaults

Debug vtsp commands are not limited to a specific port.

Command History

Release	Modification
12.0(3)XG	This command was introduced on Cisco 2600 and 3600 series routers.
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco MC3810 series.

Usage Guidelines

Use this command to limit the debug output to a particular voice port. The debug output can be quite voluminous for a single channel. The entire vtsp debug output form a platform with 12 voice ports might create problems. Use this debug with any or all of the other debug modes.

Execution of **no debug vtsp all** will turn off all VTSP-level debugging. It is usually a good idea to turn off all debugging and then enter the debug commands you are interested in one by one. This will help to avoid confusion about which ports you are actually debugging.

Examples

The following example shows sample output from the **debug vtsp port 1/1/0** command:

```
router# debug vtsp port 1/1/0
*Mar  1 03:17:33.691: vtsp_tsp_call_setup_ind (sdb=0x613FD514, tdm_info=0x0,
   tsp_info=0x613FD438, calling_number= called_number= redirect_number=): peer_tag=1110
*Mar  1 03:17:33.691: vtsp_do_call_setup_ind
```

```
*Mar  1 03:17:33.691: dsp_close_voice_channel: [] packet_len=8 channel_id=1
   packet_id=75
*Mar  1 03:17:33.691: dsp_open_voice_channel: [] packet_len=12
   channel_id=1 packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Mar  1 03:17:33.695: dsp_set_playout_delay: [] packet_len=18
   channel_id=1 packet_id=76 mode=1 initial=60 min=4 max=200 fax_nom=300
*Mar  1 03:17:33.695: dsp_echo_canceller_control: [] packet_len=10 channel_id=1
   packet_id=66 flags=0x0
*Mar  1 03:17:33.695: dsp_set_gains: [] packet_len=12 channel_id=1 packet_id=91
   in_gain=0 out_gain=65506
*Mar  1 03:17:33.695: dsp_vad_enable: [] packet_len=10 channel_id=1 packet_id=78
   thresh=-38
*Mar  1 03:17:33.695: vtsp_process_event(): [, 0.S_SETUP_INDICATED, E_CC_PROCEEDING]
*Mar  1 03:17:33.699: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_BRIDGE]act_bridge
*Mar  1 03:17:33.699: vtsp_ring_noan_timer_start: 1185370
*Mar  1 03:17:33.699: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CAPS_IND]act_caps_ind
*Mar  1 03:17:33.699: act_caps_ind: Encap 2, Vad 2, Codec 0x1000, CodecBytes 60,
              FaxRate 2, FaxBytes 30,
              Sub-channel 10, Bitmask 0x0 SignalType 2
*Mar  1 03:17:33.703: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CAPS_ACK]act_caps_ack
*Mar  1 03:17:33.703: dsp_idle_mode: [] packet_len=8 channel_id=1 packet_id=68
*Mar  1 03:17:33.703: vtsp_process_event(): [, 0.S_SETUP_INDICATED,
   E_CC_CONNECT]act_connect
*Mar  1 03:17:33.703: vtsp_ring_noan_timer_stop: 1185370
*Mar  1 03:17:33.911: vtsp_process_event(): [, 0.S_CONNECT, E_DSPRM_PEND_SUCCESS]
   act_pend_codec_success
*Mar  1 03:17:33.911: dsp_close_voice_channel: [] packet_len=8 channel_id=1
   packet_id=75
*Mar  1 03:17:33.911: dsp_open_voice_channel: [] packet_len=12 channel_id=1
   packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Mar  1 03:17:33.911: dsp_set_playout_delay: [] packet_len=18 channel_id=1 packet_id=76
   mode=1 initial=60 min=4 max=200 fax_nom=300
*Mar  1 03:17:33.911: dsp_echo_canceller_control: [] packet_len=10 channel_id=1
   packet_id=66 flags=0x0
*Mar  1 03:17:33.911: dsp_set_gains: [] packet_len=12 channel_id=1 packet_id=91
   in_gain=0 out_gain=65506
*Mar  1 03:17:33.911: dsp_vad_enable: [] packet_len=10 channel_id=1 packet_id=78
   thresh=-38
*Mar  1 03:17:33.911: dsp_encap_config: [] packet_len=24 channel_id=1 packet_id=
 92 TransportProtocol 3 SID_support=0 sequence_number=0 rotate_flag=0 header_bytes 0xA0
*Mar  1 03:17:33.915: dsp_voice_mode: [] packet_len=22 channel_id=1 packet_id=73
 coding_type=14 voice_field_size=60 VAD_flag=1 echo_length=128
 comfort_noise=1 fax_detect=1 digit_relay=0
```

Related Commands

Command	Description
debug voice eecm	Enables all VPM debugging.
show debug	Shows which debug commands are enabled.

debug vtsp session

Use the **debug vtsp session** command to trace how the router interacts with the DSP based on the signaling indications from the signaling stack and requests from the application. Use the **no** form of this command to turn off the debug function.

debug vtsp session
no debug vtsp session

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp session is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp session** command traces how the router interacts with the DSP based on the signaling indications from the signaling stack and requests from the application. This debug command displays information about how each network indication and application request is handled, signaling indications, and DSP control messages.

This debug level shows the internal workings of the voice telephony call state machine.

Sample Display

The following example shows sample output from the **debug vtsp session** command, in which the call has been accepted and the system is checking for incoming dial-peer matches:

```
*Nov 30 00:46:19.535: vtsp_tsp_call_accept_check (sdb=0x60CD4C58,
calling_number=408 called_number=1): peer_tag=0
*Nov 30 00:46:19.535: vtsp_tsp_call_setup_ind (sdb=0x60CD4C58,
tdm_info=0x60B80044, tsp_info=0x60B09EB0, calling_number=408 called_number=1):
peer_tag=1
```

The following example shows sample output from the **debug vtsp session** command, in which a DSP has been allocated to handle the call and has indicated the call to the higher layer code:

```
*Nov 30 00:46:19.535: vtsp_do_call_setup_ind:
*Nov 30 00:46:19.535: dsp_open_voice_channel: [0:D:12] packet_len=12
channel_id=8737 packet_id=74 alaw_ulaw_select=0 transport_protocol=2
*Nov 30 00:46:19.535: dsp_set_playout_delay: [0:D:12] packet_len=18
channel_id=8737 packet_id=76 mode=1 initial=60 min=4 max=200 fax_nom=300
*Nov 30 00:46:19.535: dsp_echo_canceller_control: [0:D:12] packet_len=10
channel_id=8737 packet_id=66 flags=0x0
*Nov 30 00:46:19.539: dsp_set_gains: [0:D:12] packet_len=12 channel_id=8737
packet_id=91 in_gain=0 out_gain=0
*Nov 30 00:46:19.539: dsp_vad_enable: [0:D:12] packet_len=10 channel_id=8737
packet_id=78 thresh=-38
*Nov 30 00:46:19.559: vtsp_process_event: [0:D:12, 0.3, 13] act_setup_ind_ack
```

The following example shows sample output from the **debug vtsp session** command, in which the higher layer code has accepted the call, placed the DSP in DTMF mode, and collected digits:

```
*Nov 30 00:46:19.559: dsp_voice_mode: [0:D:12] packet_len=20 channel_id=8737
packet_id=73 coding_type=1 voice_field_size=160 VAD_flag=0 echo_length=64
comfort_noise=1 fax_detect=1
*Nov 30 00:46:19.559: dsp_dtmf_mode: [0:D:12] packet_len=10 channel_id=8737
packet_id=65 dtmf_or_mf=0
*Nov 30 00:46:19.559: dsp_cp_tone_on: [0:D:12] packet_len=30 channel_id=8737
packet_id=72 tone_id=3 n_freq=2 freq_of_first=350 freq_of_second=440
amp_of_first=4000 amp_of_second=4000 direction=1 on_time_first=65535
off_time_first=0 on_time_second=65535 off_time_second=0
*Nov 30 00:46:19.559: vtsp_timer: 278792
*Nov 30 00:46:22.059: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.059: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.059: vtsp_timer: 279042
*Nov 30 00:46:22.363: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.363: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.363: vtsp_timer: 279072
*Nov 30 00:46:22.639: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.639: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.639: vtsp_timer: 279100
*Nov 30 00:46:22.843: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:22.843: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:22.843: vtsp_timer: 279120
*Nov 30 00:46:23.663: vtsp_process_event: [0:D:12, 0.4, 25] act_dcollect_digit
*Nov 30 00:46:23.663: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.663: vtsp_timer: 279202
```

The following example shows sample output from the **debug vtsp session** command, in which the call proceeded and DTMF was disabled:

```
*Nov 30 00:46:23.663: vtsp_process_event: [0:D:12, 0.4, 15] act_dcollect_proc
*Nov 30 00:46:23.663: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.663: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
```

The following example shows sample output from the **debug vtsp session** command, in which the telephony call leg was conferenced with the packet network call leg, and the telephony call leg has performed capabilities exchange with the network-side call leg:

```
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 17] act_bridge
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 22] act_caps_ind
*Nov 30 00:46:23.699: vtsp_process_event: [0:D:12, 0.5, 23] act_caps_ack
Go into voice mode with codec indicated in caps exchange.
*Nov 30 00:46:23.699: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:23.699: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:23.699: dsp_voice_mode: [0:D:12] packet_len=20 channel_id=8737
packet_id=73 coding_type=6 voice_field_size=20 VAD_flag=1 echo_length=64
comfort_noise=1 fax_detect=1
```

The following example shows sample output from the **debug vtsp session** command in which the call has been connected at remote end:

```
*Nov 30 00:46:23.779: vtsp_process_event: [0:D:12, 0.5, 10] act_connect
```

The following example shows sample output from the **debug vtsp session** command in which disconnect was indicated and passed to upper layer:

```
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 5] act_generate_disc
```

The following example shows sample output from the **debug vtsp session** command, in which the conference was torn down and the disconnect handshake was completed:

```
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 18] act_bdrop
*Nov 30 00:46:30.267: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.267: vtsp_process_event: [0:D:12, 0.11, 20] act_disconnect
*Nov 30 00:46:30.267: dsp_get_error_stat: [0:D:12] packet_len=10 channel_id=0
packet_id=6 reset_flag=1
*Nov 30 00:46:30.267: vtsp_timer: 279862
```

The following example shows sample output from the **debug vtsp session** command, in which the final DSP statistics were retrieved:

```
*Nov 30 00:46:30.275: vtsp_process_event: [0:D:12, 0.17, 30] act_get_error
*Nov 30 00:46:30.275: 0:D:12: rx_dropped=0 tx_dropped=0 rx_control=353
tx_control=338 tx_control_dropped=0 dsp_mode_channel_1=2 dsp_mode_channel_2=0
c[0]=71 c[1]=71 c[2]=71 c[3]=71 c[4]=68 c[5]=71 c[6]=68 c[7]=73 c[8]=83 c[9]=84
c[10]=87 c[11]=83 c[12]=84 c[13]=87 c[14]=71 c[15]=6
*Nov 30 00:46:30.275: dsp_get_levels: [0:D:12] packet_len=8 channel_id=8737
packet_id=89
*Nov 30 00:46:30.279: vtsp_process_event: [0:D:12, 0.17, 34] act_get_levels
*Nov 30 00:46:30.279: dsp_get_tx_stats: [0:D:12] packet_len=10 channel_id=8737
packet_id=86 reset_flag=1
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.17, 31] act_stats_complete
*Nov 30 00:46:30.287: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.287: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:30.287: vtsp_timer: 279864
```

The following example shows sample output from the **debug vtsp session** command, in which the DSP channel was closed and released:

```
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.18, 6] act_wrelease_release
*Nov 30 00:46:30.287: dsp_cp_tone_off: [0:D:12] packet_len=8 channel_id=8737
packet_id=71
*Nov 30 00:46:30.287: dsp_idle_mode: [0:D:12] packet_len=8 channel_id=8737
packet_id=68
*Nov 30 00:46:30.287: dsp_close_voice_channel: [0:D:12] packet_len=8
channel_id=8737 packet_id=75
*Nov 30 00:46:30.287: vtsp_process_event: [0:D:12, 0.16, 42] act_terminate
```

Related Commands

Command	Description
debug voice eecm	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp stats

Use the **debug vtsp stats** command to debug periodic statistical-information-request messages sent and received from the DSP during a call. Use the **no** form of this command to turn off the debug function.

debug vtsp stats
no debug vtsp stats

Syntax Description

This command has no arguments or keywords.

Defaults

Debugging for vtsp stats is not enabled.

Command History

Release	Modification
12.0(3)T	This command was introduced on the Cisco AS5300 platform.
12.0(7)XK	This command was first supported on the Cisco 2600, 3600 and MC3810 platforms.

Usage Guidelines

The **debug vtsp stats** command generates a collection of DSP statistics for generating RTCP packets and a collection of other statistical information.

Sample Display

The following example shows sample **debug vtsp stats** output:

```
*Nov 30 00:53:26.499: vtsp_process_event: [0:D:14, 0.11, 19] act_packet_stats
*Nov 30 00:53:26.499: dsp_get_voice_playout_delay_stats: [0:D:14] packet_len=10
channel_id=8753 packet_id=83 reset_flag=0
*Nov 30 00:53:26.499: dsp_get_voice_playout_error_stats: [0:D:14] packet_len=10
channel_id=8753 packet_id=84 reset_flag=0
*Nov 30 00:53:26.499: dsp_get_rx_stats: [0:D:14] packet_len=10 channel_id=8753
packet_id=87 reset_flag=0
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_VOICE_PLAYOUT_DELAY:
clock_offset=-1664482334 curr_rx_delay_estimate=69 low_water_mark_rx_delay=69
high_water_mark_rx_delay=70
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 28]
act_packet_stats_res
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_VOICE_PLAYOUT_ERROR:
predective_concelement_duration=0 interpolative_concelement_duration=0
silence_concelement_duration=0 retroactive_mem_update=0
buf_overflow_discard_duration=10 num_talkspurt_detection_errors=0
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 29]
act_packet_stats_res
*Nov 30 00:53:26.503: vtsp_process_dsp_message: MSG_TX_GET_RX_STAT:
num_rx_pkts=152 num_early_pkts=-2074277660 num_late_pkts=327892
num_signalling_pkts=0 num_comfort_noise_pkts=0 receive_durtation=3130
voice_receive_duration=2970 fax_receive_duration=0 num_pack_ooseq=0
num_bad_header=0
*Nov 30 00:53:26.503: vtsp_process_event: [0:D:14, 0.11, 32]
act_packet_stats_res
```

Related Commands

Command	Description
debug voice eecm	Enables all VPM debugging.
debug vtsp port	Limits vtsp debug output to a specific voice port.
show debug	Shows which debug commands are enabled.

debug vtsp vofr subframe

To display the first 10 bytes (including header) of selected VoFR subframes for the interface, use the **debug vtsp vofr subframe** command in privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug vtsp vofr subframe *payload* **[from-dsp] [to-dsp]**
no debug vtsp vofr subframe

Syntax Description

payload	Number used to selectively display subframes of a specific payload. The payload types are: 0: Primary Payload - WARNING! This option may cause network instability. 1: Annex-A 2: Annex-B 3: Annex-D 4: All other payloads 5: All payloads - WARNING! This option may cause network instability.
from-dsp	(Optional) Displays only the subframes received from the DSP.
to-dsp	(Optional) Displays only the subframes going to the DSP.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco 2600 and 3600.
12.0(7)XK	Support for this command was extended to the Cisco MC3810.

Usage Guidelines

Each debug output displays the first 10 bytes of the FRF.11 subframe, including header bytes. Use the **from-dsp** and **to-dsp** options to limit the debugs to a single direction. If not specified, debugs are displayed for subframes when they are received from the DSP and before they are sent to the DSP.

Use extreme caution in selecting payload options 0 and 5. These options may cause network instability.

Examples

The following example shows sample output from the **debug vtsp vofr subframe** command:

```
router# debug vtsp vofr subframe 2
vtsp VoFR subframe debugging is enabled for payload 2 to and from DSP 3620_vofr#
*Mar  6 18:21:17.413:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:17.449:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
*Mar  6 18:21:23.969:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:24.005:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
```

Related Commands

Command	Description
debug vtsp all	Enables debugging of all VTSP areas.