



## Voice-Related Commands

This chapter documents voice-related commands used to configure Voice over IP, Voice over Frame Relay, Voice over HDLC, and Voice over ATM. Commands in this section are listed alphabetically. For this release, Voice over IP is specific to the Cisco 3600 series; Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking are specific to the Cisco MC3810. For information on how to configure Voice over IP, Voice over Frame Relay, Voice over ATM, or Voice over HDLC, refer to the *Voice, Video, and Home Applications Configuration Guide*.

Table 1 describes which commands apply to each protocol.

**Table 1** Voice-Related Commands Per Protocol

Commands	Voice over IP	Voice over Frame Relay	Voice over ATM	Voice over HDLC	Frame Relay-ATM Interworking
acc-qos	X				
alt-dial		X	X	X	
answer-address	X				
atm compression			X		
cablelength long		X	X	X	
cablelength short		X	X	X	
channel-group		X	X	X	
clear voice port		X	X	X	
clock rate line		X	X	X	
clock rate network-clock		X	X	X	
clock source		X	X	X	
codec	X				
compress			X		
cross-connect		X	X	X	X
destination-pattern	X	X	X	X	
dial-control-mib	X				
dial-peer terminator	X				
dial-peer voice	X	X	X	X	X
encapsulation			X		X
expect-factor	X				

Table 1 Voice-Related Commands Per Protocol (continued)

Commands	Voice over IP	Voice over Frame Relay	Voice over ATM	Voice over HDLC	Frame Relay-ATM Interworking
fax-rate	X				
forward-digits		X	X	X	
frame-relay interface-dlci		X			
frame-relay route		X			
fr-atm connect dlci					X
icpif	X				
interface fr-atm					X
ip precedence	X				
ip udp checksum	X				
loopback			X		
loop-detect		X	X	X	
mode			X		
network-clock base-rate		X	X	X	X
network-clock-select		X	X	X	X
network-clock-switch		X	X	X	X
num-exp	X				
port	X	X	X	X	
preference		X	X	X	
prefix	X	X	X	X	
req-qos	X				
session protocol	X				
session target	X	X	X	X	
show call active voice	X				
show call history voice	X				
show dial-peer voice	X	X	X	X	
show dialplan incall number	X				
show dialplan number	X	X	X	X	
show network-clocks		X	X	X	
show num-exp	X				
show voice call		X	X	X	
show voice dsp		X	X	X	
show voice port	X	X	X	X	
shutdown (dial-peer)	X	X	X	X	
shutdown (DS 1 link)		X	X	X	

**Table 1** *Voice-Related Commands Per Protocol (continued)*

Commands	Voice over IP	Voice over Frame Relay	Voice over ATM	Voice over HDLC	Frame Relay-ATM Interworking
snmp enable peer-trap poor-qov	X	X	X	X	
snmp-server enable traps	X	X	X	X	
tdm-group		X	X	X	
vad	X				
vbr-rt			X		
voice-encap				X	
voice-group		X	X	X	
voice local-bypass		X	X	X	

## acc-qos

To generate an SNMP event if the quality of service for a dial peer drops below a specified level, use the `acc-qos dial-peer` configuration command. Use the `no` form of this command to restore the default value.

```
acc-qos {best-effort | controlled-load | guaranteed-delay}
```

```
no acc-qos
```

## Syntax Description

<b>best-effort</b>	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation. This is the default.
<b>controlled-load</b>	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to ensure that preferential service is received even when the bandwidth is overloaded.
<b>guaranteed-delay</b>	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

## Default

```
best-effort
```



## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Cisco MC3810 POTS, VoFR, VoATM, and VoHDLC dial peers.

The **alt-dial** command is used for the on-net-to-off-net alternative dialing function. The string replaces the destination-pattern string for dialing out.

## Example

The following example configures an alternate dial-out string of 9,5559871:

```
alt-dial 9,5559871
```

## answer-address

To specify the full E.164 telephone number to be used to identify the dial peer of an incoming call, use the **answer-address** dial-peer configuration command. Use the **no** form of this command to disable the configured telephone number.

```
answer-address [+]string[t]
```

```
no answer-address
```

## Syntax Description

<b>+</b>	(Optional) Character indicating an E.164 standard number.
<i>string</i>	Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are: <ul style="list-style-type: none"><li>• Digits 0 through 9, letters A through D, pound sign (#), and asterisk (*), which represent specific digits that can be entered.</li><li>• Comma (,), which inserts a pause between digits.</li><li>• Period (.), which matches any entered digit.</li></ul>
<b>t</b>	(Optional) Control character indicating that the <b>answer-address</b> value is a variable length dial-string.

## Default

The default value is enabled with a null string.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable to both Cisco 3600 series VoIP and POTS dial peers.

Use the **answer-address** command to identify the origin (or dial peer) of incoming calls from the IP network. Cisco IOS software identifies the dial peers of a call in one of two ways: either by identifying the interface through which the call is received or through the telephone number configured with the **answer-address** command. In the absence of a configured telephone number, the peer associated with the interface will be associated with the incoming call.

For calls coming in from a POTS interface, the **answer-address** command is not used to select an incoming dial peer. The incoming POTS dial peer is selected on the basis of the port configured for that dial peer.

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. Use the optional control character **t** to indicate that a particular **answer-address** value is a variable-length dial-string. In this case, the system will not match the dialed numbers until the interdigit timeout value has expired.



The Cisco IOS software does not check the validity of the E.164 telephone number; it accepts any series of digits as a valid number.

## Example

The following example configures the E.164 telephone number, “555-9626” as the dial peer of an incoming call:

```
dial-peer voice 10 pots
  answer-address +5559626
```

## Related Commands

**destination-pattern**  
**port**  
**prefix**  
**timeouts interdigit**

## atm compression

To specify the software compression mode on the Cisco MC3810, use the **atm compression** interface configuration command. Use the **no** form of this command to remove the compression mode setting.

```
atm compression {per-packet | per-interface | per-vc}
no atm compression {per-packet | per-interface | per-vc}
```

## Syntax Description

<i>per-packet</i>	Specifies packet-by-packet compression mode (no history). This is the default.
<i>per-interface</i>	Specifies one context per interface (with history).

per-vc Specifies one context for every virtual circuit (with history).

## Default

per-packet

## Command Mode

Interface configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to ATM configuration on the Cisco MC3810.

## Example

The following example configures per-packet ATM compression:

```
interface atm0
 atm compression per-packet
```

## clear voice port

To clear voice port calls in progress on the Cisco MC3810, use the **clear voice port** privileged EXEC command.

```
clear voice port [slot/port]
```

## Syntax Description

*slot/port* (Optional) The voice port slot number and port number. If you do not specify a voice port, all calls on all voice ports are cleared.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

If you do not specify a voice port, all calls on all voice ports are cleared. A confirmation prompt is displayed.

## Example

The following example clears all calls on voice port 1/2 on the Cisco MC3810:

```
clear voice port 1/2
```

## clock rate line

To configure the line clock rate for serial ports 0 or 1 in DTE mode on the Cisco MC3810, use the **clock rate line** interface configuration command. Use the **no** form of this command to cancel the clock rate line value.

```
clock rate line rate
```

```
no clock rate line rate
```

## Syntax Description

*rate* Network clock line rate in bits per second. The range is from 56 kbps to 2048 kbps. The value entered should be a multiple of 8,000 of the value set for the **network-clock base-rate** command. There is no default rate.

## Default

No clock rate is set.

## Command Mode

Interface configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command specifies the rate of the incoming clock so that the appropriate internal clock scaling can be performed.

To configure the clock rate for a serial port in DTE mode, use the **clock rate network-clock** command.

## Example

The following configures the clock rate on serial 1 in DTE mode:

```
interface serial 1
  clock rate line 2048
```

## Related Commands

clock rate network-clock  
clock source  
network-clock base-rate

## clock rate network-clock

To configure the network clock speed for serial ports 0 or 1 in DCE mode on the Cisco MC3810, use the `clock rate network-clock` interface configuration command. Use the `no` form of this command to cancel the network clock speed value.

`clock rate network-clock rate`  
`no clock rate network-clock rate`

## Syntax Description

*rate* Network clock speed in bits per second. The range is from 56 kbps to 2048 kbps. The value entered should be a multiple of the value set for the `network-clock base-rate` command. There is no default rate.

## Default

No clock rate is set.

## Command Mode

Interface configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command uses a synchronized clock on the serial port. The use of this command allows the clock on the serial port to be synchronized with the clock source of controller T1 0.

To configure the clock rate for a serial port in DTE mode, use the `clock rate line` command.

## Example

The following configures the clock rate on serial 1 in DCE mode:

```
interface serial 1
  clock rate network-clock 2048
```

## Related Commands

clock rate line  
clock source  
network-clock base-rate

## codec (dial-peer)

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

```
codec {g711alaw | g711ulaw | g729r8}
no codec
```

## Syntax Description

<code>g711alaw</code>	G.711 A-Law 64,000 bits per second (bps).
<code>g711ulaw</code>	G.711 u-Law 64,000 bps.
<code>g729r8</code>	G.729 8000 bps. This is the default CODEC.

## Default

`g729r8`

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is only applicable to Cisco 3600 series VoIP peers.

Use the `codec` command to define a specific voice coder rate of speech for a dial peer.

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

If `codec` values for the VoIP peers of a connection do not match, the call will fail.



The CODEC value for the Cisco MC3810 is configured on the voice port.

## Example

The following example configures a voice coder rate that provides toll quality and uses a relatively high amount of bandwidth:

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

## compress

To configure whether compression is applied to the voice connection on the ATM permanent virtual circuit (PVC) for the Cisco MC3810, use the **compress** ATM virtual circuit configuration command. Use the **no** form of this command to turn off compression.

```
compress
no compress
```

## Syntax Description

This command has no arguments or keywords.

## Default

No compression

## Command Mode

ATM virtual circuit configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 12.0.

This command applies to the ATM configuration on the Cisco MC3810.

## Example

The following example configures ATM PVC 20 to support voice compression:

```
pvc 20
  compress
```

## cross-connect

To cross-connect two groups of digital signal level 0s (DS0s) from two controllers on the Cisco MC3810, or to cross-connect the Universal I/O (UIO) serial port for pass-through traffic to a trunk controller, use the **cross-connect** global configuration command. Use the **no** form of this command to remove the cross-connect function for the given controller.

For pass-through between two controllers, use the following commands:

```
cross-connect id controller-1 tdm-group-no-1 controller-2 tdm-group-no-2
no cross-connect id controller-1 tdm-group-no-1 controller-2 tdm-group-no-2
```

For pass-through traffic from a UIO serial port to a trunk controller, use the following commands:

```
cross-connect id interface-serial controller tdm-group-no
no cross-connect id interface-serial controller tdm-group-no
```



**Note** The UIO serial port is either serial port 0 or 1.

## Syntax Description

For pass-through between two controllers:

<i>id</i>	Unique ID assigned to this cross-connection. The valid range is from 0 to 31.
<i>controller-1</i>	Type of the first controller (T1 0, T1 1, or E1)
<i>tdm-group-no-1</i>	TDM group number assigned to the first controller.
<i>controller-2</i>	Type of the second controller (T1, E1 0, or E1 1).
<i>tdm-group-no-2</i>	TDM group number assigned to the second controller.

For pass-through from a UIO serial port to a controller

<i>id</i>	Unique ID assigned to this cross connection.
<i>interface-serial</i>	Number of the serial port, either 0 or 1.
<i>controller</i>	Type of the controller. Enter one of the following: T1 0, T1 1, E1 0, or E1 1.
<i>tdm-group-no</i>	TDM group number assigned to the controller.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking on the Cisco MC3810.

## Example

The following example configures a pass-through cross-connect from serial port 0 to controller T1 1 on TDM group 20.

```
cross-connect 10 serial0 T1 1 20
```

## Related Commands

tdm-group

## destination-pattern

To specify either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer, use the **destination-pattern** dial-peer configuration command. Use the **no** form of this command to disable the configured prefix or telephone number.

```
destination-pattern [+]string[t]
```

```
no destination-pattern
```

## Syntax Description

- |               |   |
|---------------|---|
| <b>+</b>      | (Optional) Character indicating an E.164 standard number. The plus sign (+) is not supported on the Cisco MC3810.   |
| <i>string</i> | Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: <ul style="list-style-type: none"><li>• The asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. On the Cisco 3600 only, these characters cannot be used as leading characters in a string (for example, *650).</li><li>• Comma (,), which inserts a pause between digits.</li><li>• Period (.), which matches any entered digit (this character is used as a wildcard). On the Cisco 3600, the period cannot be used as a leading character in a string (for example, .650).</li></ul> |
| <b>t</b>      | Control character indicating that the <b>destination-pattern</b> value is a variable length dial-string.  |

## Default

enabled with a null string

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable to both VoIP and POTS dial peers on all platforms.

Use the **destination-pattern** command to define the E.164 telephone number for this dial peer.

This pattern is used to match dialed digits to a dial peer. The dial peer is then used to complete the call. When a router receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the voice-telephony peer. The router then strips out the left-justified numbers corresponding to the destination pattern. If you have configured a prefix, the prefix is appended to the front of the remaining numbers, creating a dial string, which the router then dials. If all numbers in the destination pattern are stripped-out, the user receives a dial tone.

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. Use the optional control character **t** to indicate that a particular **destination-pattern** value is a variable-length dial-string. In this case, the system does not match the dialed numbers until the interdigit timeout value has expired.



The Cisco IOS software does not check the validity of the E.164 telephone number; it accepts any series of digits as a valid number.

## Example

The following example configures the E.164 telephone number, “555-7922,” for a dial peer:

```
dial-peer voice 10 pots
 destination-pattern +5557922
```

## Related Commands

answer-address  
prefix  
timeouts interdigit

## dial-control-mib

To specify attributes for the call history table, use the **dial-control-mib** global configuration command.

```
dial-control-mib {max-size number | retain-timer number}
```

## Syntax Description

<b>max-size</b> <i>number</i>	Specifies the maximum size of the call history table. Valid entries are from 0 to 500 table entries. A value of 0 prevents any history from being retained.
<b>retain-timer</b> <i>number</i>	Specifies the length of time, in minutes, for entries in the call history table. Valid entries are from 0 to 2147483647 minutes. A value of 0 prevents any history from being retained.

## Default

The default call history table length is 50 table entries. The default retain timer is 15 minutes.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

## Example

The following example configures the call history table to hold 400 entries, with each entry remaining in the table for 10 minutes:

```
configure terminal
dial-control-mib max-size 400
dial-control-mib retain-timer 10
```

## dial-peer terminator

To designate a special character to be 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 disable the designated terminating character.

```
dial-peer terminator character
no dial-peer terminator character
```

## Syntax Description

<i>character</i>	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.
------------------	--

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 12.0.

This command applies only to the Cisco 3600 series.

There are certain areas in the world (for example, in certain European countries) where valid telephone numbers can vary in length. When a dialed-number string has been identified as a variable length dialed-number, the system waits until the configured value for the `timeouts interdigits` command has expired before placing the call. To avoid waiting until the interdigit timeout value has expired, you can designate a special character as a terminator—meaning that when you dial that character, the system no longer waits for any additional digits and places the call. Use the `dial-peer terminator` global configuration command to designate a particular character as a terminator.

## Example

The following example configures # as the special terminating character for variable-length dialed-numbers:

```
configure terminal
dial-peer terminator #
```

## Related Commands

`answer-address`  
`destination-pattern`  
`timeouts interdigit`

## dial-peer voice

To enter dial-peer configuration mode (and specify the method of voice-related encapsulation), use the `dial-peer voice` global configuration command.

**For the Cisco 3600 series:**

```
dial-peer voice number {voip | pots}
```

**For the Cisco MC3810:**

```
dial-peer voice tag-number {pots | voatm | vofr | vohdlc}
```

## Syntax Description

**For the Cisco 3600 series Voice over IP:**

<i>number</i>	Digit(s) defining a particular dial peer. Valid entries are from 1 to 2147483647.
<code>voip</code>	Indicates that this is a VoIP peer using voice encapsulation on the POTS network.
<code>pots</code>	Indicates that this is a POTS peer using Voice over IP encapsulation on the IP backbone.

**For the Cisco MC3810:**

<i>tag-number</i>	Digit(s) defining a particular dial peer. Defines the dial peer and assigns the protocol type to it. Valid entries are from 1 to 10000.
<b>pots</b>	Indicates that this is a POTS peer using basic telephone service.
<b>voatm</b>	Indicates that this is a Voice over ATM peer using the real-time AAL5 voice encapsulation on the ATM backbone network.
<b>vofr</b>	Indicates that this is a Voice over Frame Relay peer using encapsulation on the Frame Relay backbone network.
<b>vohdlc</b>	Indicates that this is a Voice over HDLC peer using Cisco serial encapsulation (HDLC) for voice.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to all voice applications on both the Cisco MC3810 and the Cisco 3600 series.

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.

## Example

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

```
configure terminal
dial-peer voice 10 pots
```

## Related Commands

**voice-port**

## expect-factor

To specify when the router will generate an alarm to the network manager, indicating that the expected quality of voice has dropped, use the **expect-factor** dial-peer configuration command. Use the **no** form of this command to reset the default value.

```
expect-factor value
no expect-factor value
```

## Syntax Description

*value*                      Integers that represent the ITU specification for quality of voice as described in G.113. Valid entries are from 0 to 20, with 0 representing toll quality.

## Default

10

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Cisco 3600 series VoIP peers.

Voice over IP monitors the quality of voice received over the network. Use the **expect-factor** command to specify when the router will generate an SNMP trap to the network manager.

## Example

The following example configures toll quality of voice when connecting to a dial peer:

```
dial-peer voice 10 voip
  expect-factor 0
```

## fax-rate

To establish the rate at which a fax will be sent to the specified dial peer, use the **fax-rate** dial-peer configuration command. Use the **no** form of this command to reset the default value.

```
fax-rate {2400 | 4800 | 7200 | 9600 | 14400 | disable | voice}
no fax-rate
```

## Syntax Description

2400	Specifies a fax transmission speed of 2400 bits per second (bps).
4800	Specifies a fax transmission speed of 4800 bps.
7200	Specifies a fax transmission speed of 7200 bps.
9600	Specifies a fax transmission speed of 9600 bps.
14400	Specifies a fax transmission speed of 14,400 bps.

<code>disable</code>	Disables fax relay transmission capability.
<code>voice</code>	Specifies the highest possible transmission speed allowed by voice rate.

## Default

`voice`

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is only applicable to Cisco 3600 series VoIP peers.

Use the `fax-rate` command to specify the fax transmission rate to the specified dial peer.

The values for this command apply only to the fax transmission speed and do not affect the quality of the fax itself. The higher values provide a faster transmission speed but monopolize a significantly larger portion of the available bandwidth. Slower transmission speeds use less bandwidth.

If the `fax-rate` command is set above the codec rate in the same dial peer, the data sent over the network for fax transmission will be above the bandwidth reserved for RVSP. Because more network bandwidth will be monopolized by the fax transmission, we do not recommend setting the `fax-rate` value higher than the `codec` value. If the `fax-rate` value is set lower than the `codec` value, faxes will take longer to transmit but will use less bandwidth.

## Example

The following example configures a transmission speed of 9600 bps for faxes sent to a dial peer:

```
dial-peer voice 10 voip
  fax-rate 9600
```

## Related Commands

`codec`

## forward-digits

To configure forward digits for voice calls on the Cisco MC3810, use the `forward-digits dial-peer` configuration command. Use the `no` form of this command to restore the default value.

```
forward-digits {num-digit | all}
no forward-digits {num-digit | all}
```

## Syntax Description

<i>num-digit</i>	The number of digits to be forwarded. If the number of digits is longer than the length of a destination phone number, the length of the destination number is used.
all	Forward all digits. If “all” is used, the full length of the destination pattern will be used.

## Default

No digits are forwarded.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies only to the Cisco MC3810.

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

## Example

The following example configures forward digits for a POTS dial peer on a Cisco MC3810:

```
dial-peer voice 1 pots
  destination-pattern 8...
  forward-digits all
```

## Related Commands

destination-pattern

## fr-atm connect dlci

To map a Frame Relay DLCI to an ATM virtual circuit descriptor (VCD) for the FRF.5 Frame Relay-ATM interworking function on the Cisco MC3810, use the **fr-atm connect dlci** interface configuration command. The encapsulation type of the current interface must be Frame Relay or Frame Relay 1490 (IETF). Use the **no** form of this command to remove the DLCI to VCD map.

```
fr-atm connect dlci dlci atm-interface pvc [name | [vpi/] vci]
no fr-atm connect dlci dlci atm-interface pvc [name | [vpi/] vci]
```

## Syntax Description

<i>dlci</i>	The Frame Relay DLCI number.
<i>atm-interface</i>	The ATM interface mapped to the DLCI. The ATM interface must be a serial interface with ATM encapsulation. On the Cisco MC3810, the interface must be ATM 0.
<i>pvc</i>	Specifies the ATM PVC.
<i>name</i>	The ATM PVC name.
<i>vpi</i>	The ATM virtual path identifier (VPI).
<i>vci</i>	The ATM virtual channel identifier (VCI).

## Default

No Frame Relay-ATM mapping is configured.

## Command Mode

Interface configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Frame Relay-ATM Interworking on the Cisco MC3810.

**Note**

The Cisco MC3810 provides only *network interworking* (FRF.5). The Cisco MC3810 can be used with *service interworking* (FRF.8), which is provided by the carrier's ATM network equipment.

## Example

The following example configures a Frame Relay-ATM Interworking connection on FR-ATM interface 20, in which Frame Relay DLCI 100 is mapped to ATM VCI/VPI 100/200 for ATM interface 0.

```
interface fr-atm 20
 fr-atm connect dlci 100 atm0 100/200
```

## icpif

To specify the 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.

**icpif** *number*

**no icpif** *number*

## Syntax Description

*number* Integer, expressed in equipment impairment factor units, specifying the ICPIF value. Valid entries are 0 to 55. The default is 30.

## Default

30

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable only to Cisco 3600 series VoIP peers.

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

## Example

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

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

## interface fr-atm

To create a Frame Relay-ATM Interworking interface on the Cisco MC3810 and to enter Frame Relay-ATM Interworking configuration mode, use the **interface fr-atm** global configuration command. Use the **no** form of this command to delete the Frame Relay-ATM Interworking interface.

```
interface fr-atm number
no interface fr-atm number
```

## Syntax Description

*number* The Frame Relay-ATM Interworking interface number. Valid range is from 0 to 20.

## Default

Frame Relay-ATM Interworking interface 20 is configured by default.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Frame Relay-ATM Interworking on the Cisco MC3810 only.

Use the **interface fr-atm** command to enter Frame Relay-ATM interworking interface configuration mode. When you enter this command for the first time, an interface number is created dynamically. You can configure up to 21 Frame Relay-ATM interworking interfaces.

**Note**

The Cisco MC3810 provides only *network interworking* (FRF.5). The Cisco MC3810 can be used with *service interworking* (FRF.8), which is provided by the carrier's ATM network equipment.

## Example

The following example configures Frame Relay-ATM Interworking interface number 20:

```
interface fr-atm 20
```

## Related Command

fr-atm connect dlci

## ip udp checksum

To calculate the UDP checksum for voice packets transmitted by the dial peer, use the **ip udp checksum dial-peer** configuration command. Use the **no** form of this command to disable this feature.

```
ip udp checksum
```

```
no ip udp checksum
```

## Syntax Description

This command has no arguments or keywords.

## Default

disabled

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable to Cisco 3600 series VoIP peers.

Use the **ip udp checksum** command to enable UDP checksum calculation for each of the outbound voice packets. This command is disabled by default to speed up the transmission of the voice packets. If you suspect that the connection has a high error rate, you should enable **ip udp checksum** to prevent bad voice packets forwarded to the DSP.

## Example

The following example calculates the UDP checksum for voice packets transmitted by this dial peer:

```
dial-peer voice 10 voip
 ip udp checksum
```

## Related Commands

loop-detect

## loop-detect

To enable loop detection for T1, use the **loop-detect** controller configuration command. Use the **no** form of the command to cancel the loop detect operation.

```
loop-detect
no loop-detect
```

## Syntax Description

This command has no arguments or keywords.

## Default

Loop detection is disabled.

## Command Mode

Controller configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

## Example

The following example configures loop detection for controller T1 0:

```
controller t1 0
 loop-detect
```

## Related Commands

loopback

## mode (Voice over ATM)

To set the mode of the T1/E1 controller and enter specific configuration commands for each mode type, use the **mode** controller configuration command. Use the **no** form of this command to restore the default mode of the controller.

```
mode {atm | cas}
```

```
no mode {atm | cas}
```

## Syntax Description

atm	<p>Sets the controller into ATM mode and creates an ATM interface (ATM 0) on the Cisco MC3810. When ATM mode is enabled, no channel groups, CAS groups, CCS groups, or clear channels are allowed because ATM occupies all the DS0s on the T1/E1 trunk.</p> <p>When you set the controller to ATM mode, the controller framing is automatically set to ESF for T1 or CRC4 for E1. The linecode is automatically set to B8ZS for T1 or HDB3 for E1. When you remove ATM mode by entering the <b>no mode atm</b> command, ATM interface 0 is deleted.</p> <p>ATM mode is supported only on controller 0 (T1 or E1 0).</p>
cas	<p>Sets the controller into channel associated signaling (CAS) mode, which allows you to create channel groups, CAS groups, and clear channels (both data and CAS modes).</p> <p>CAS mode is supported on both controllers 0 and 1.</p>

## Default

No mode is configured.

## Command Mode

Controller configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to the Cisco MC3810 with the digital voice module (DVM) installed.

When no mode is selected, channel groups and clear channels (data mode) can be created using the **channel group** and **tdm-group** commands, respectively.

On the Cisco MC3810, some DS0s are used exclusively for different signaling modes. The DS0 channels have the following limitations when mixing different applications (such as voice and data) on the same network trunk:

- On E1 controllers, DS0 16 is used exclusively for either CAS or CCS, depending on which mode is configured.
- On T1 controllers, DS0 24 is used exclusively for CCS.

## Examples

The following example configures ATM mode on controller T1 0. This step is required for Voice over ATM:

```
controller T1 0
 mode atm
```

The following example configures CAS mode on controller T1 1:

```
controller T1 1
 mode cas
```

## Related Commands

```
channel-group
tdm-group
voice-group
```

## network-clock base-rate

To configure the network clock base rate for universal I/O serial ports 0 and 1 on the Cisco MC3810, use the `network-clock base-rate` global configuration command. Use the `no` form of this command to disable the current network clock base rate.

```
network-clock base-rate {56k | 64k}
no network-clock base-rate {56k | 64k}
```

## Syntax Description

56k	Sets the network clock base rate to 56 kilobits per second (kbps).
64k	Sets the network clock base rate to 64kbps.

## Default

56 kbps

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking on the Cisco MC3810.

## Example

The following example sets the network clock base-rate to 64 kbps:

```
network-clock base-rate 64k
```

## Related Commands

`network-clock-select (MC3810)`  
`network-clock-switch`

## network-clock-select (MC3810)

To use the network clock source to provide timing to the system backplane pulse code modulation (PCM) bus, use the `network-clock-select` global configuration command. Use the `no` form of this command to cancel the network clock selection.

```
network-clock-select priority [serial 0 | system | controller]  
no network-clock-select priority [serial 0 | system | controller]
```

## Syntax Description

<i>priority</i>	Specifies the priority of the clock source. Valid entries are from 1 to 4. You can configure up to four clock sources. The higher the number of the clock source, the higher the priority. For example, clock source 1 has higher priority than clock source 2. When the higher priority clock source fails, after the delay specified using the <code>network-clock-switch</code> command, the next higher priority clock source is selected.
<code>serial 0</code>	(Optional) Specifies serial interface 0 as the clock source.
<code>system</code>	(Optional) Specifies the system clock as the clock source.
<i>controller</i>	(Optional) Specifies which controllers is the clock source. You can specify either the trunk controller (T1/E1 0) or the digital voice module (T1/E1/ 1).

## Default

No network clock source is specified.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking on the Cisco MC3810.

## Example

The following example sets the priority of four network clock sources. When the clock source with the highest priority (controller T1 0) fails, the Cisco MC3810 switches the clock source to the second highest priority (controller T1 1).

```
network-clock-select 1 T1 0
network-clock-select 2 T1 1
network-clock-select 3 serial 0
network-clock-select 4 System
```

## Related Commands

network-clock-switch

# network-clock-switch

To configure the switch delay time to the next priority network clock source when the current network clock source fails, use the **network-clock-switch** global configuration command. Use the **no** form of this command to cancel the network clock delay time selection.

```
network-clock-switch [switch-delay | never] [restore-delay | never]
no network-clock-switch delay
```

## Syntax Description

<i>switch-delay</i>	The delay time before the next priority network clock source is used when the current network clock source fails. The range is from 0 to 99 seconds. The default is 10 seconds.
<b>never</b>	(Optional) Indicates no delay time before the current network clock source recovers.
<i>restore-delay</i>	(Optional) The delay time before the current network clock source recovers. The range is from 0 to 99 seconds.
<b>never</b>	(Optional) Indicates no delay time before the next priority network clock source is used when the current network clock source fails.

## Default

10 seconds

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking on the Cisco MC3810.

## Example

The following command switches the network clock source after 20 seconds and sets the delay time before the current network clock source recovers to 20 seconds:

```
network-clock-switch 20 20
```

## Related Commands

network-clock-select (MC3810)

## num-exp

To define how to expand an extension number into a particular destination pattern, use the **num-exp** global configuration command. Use the **no** form of this command to cancel the configured number expansion.

```
num-exp extension-number expanded-number
```

```
no num-exp extension-number expanded-number
```

## Syntax Description

*extension-number* Digit(s) defining an extension number for a particular dial peer.

*expanded-number* Digit(s) defining the expanded telephone number or destination pattern for the extension number listed.

## Default

No number expansion is defined.

## Command Mode

Global configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

Use the **num-exp** global configuration command to define how to expand a particular set of numbers (for example, an extension number) into a particular 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—meaning that if you want to replace four numbers in an extension with wildcards, type in four periods.

## Examples

The following example expands the extension number 55541 to be expanded to 1408555541:

```
num-exp 65541 1408555541
```

The following example expands all five-digit extensions beginning with 5 to append the following numbers at the beginning of the extension number 1408555:

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

## port (voice)

To associate a dial peer with a specific voice port, use the **port dial-peer** configuration command. Use the **no** form of this command to cancel this association.

**For the Cisco 3600 series:**

```
port slot-number/subunit-number/port  
no port
```

**For the Cisco MC3810:**

```
port slot/port  
no port
```

## Syntax Description

<i>slot</i>	(Cisco MC3810) Slot number where the voice interface card is installed. Valid entries are 1 or 0.
<i>slot-number</i>	(Cisco 3600 series) Slot number in the router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
<i>subunit-number</i>	(Cisco 3600 series) Subunit on the voice interface card in the router where the voice port is located. Valid entries are 0 or 1.

*port* (Cisco 3600 series) Voice port number. Valid entries are 0 or 1.  
(Cisco MC3810) Voice port number. Valid entries are the following:

- Analog voice ports: 1-6
- Digital voice ports:
  - T1: 1-24
  - E1: 1-15, and 16-31

## Default

No port is configured.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable only to POTS peers on both the Cisco 3600 series and the Cisco MC3810.

Use the **port** configuration command to associate the designated voice port with the selected dial peer.

This command is used for calls incoming from a telephony interface to select an incoming dial peer and for calls coming from the VoIP network to match a port with the selected outgoing dial peer.

## Examples

The following example associates a Cisco 3600 series dial peer with voice port 1, which is located on subunit 0, and accessed through port 0:

```
dial-peer voice 10 pots
  port 1/0/0
```

The following example associates a Cisco MC3810 dial peer with voice port 1/1:

```
port 1/1
```

## preference

To indicate the preference order of the dial peer on the Cisco MC3810, use the **preference** dial-peer configuration command. Setting the preference is useful in selecting the desired dial peer when multiple dial peers are matched for a dial string. Use the **no** form of this command to remove the preference value on the voice port.

**preference** *value*

**no preference** *value*

## Syntax Description

*value* An integer value from 0 to 10, where the lower the number, the higher the preference. The default value is 0 (highest preference).

## Default

0 (highest preference)

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Cisco MC3810 POTS, Voice over Frame Relay, Voice over ATM, and Voice over HDLC dial peers.

**Note**

If POTS and voice-network peers are mixed in the same hunt group, the POTS dial peers must have priority over the voice-network peers.

## Examples

The following example configures POTS dial peer 10 to a preference of 1 and VoFR dial peer 20 to a preference of 2:

```
dial-peer voice 10 pots
  preference 1
```

```
dial-peer voice 20 vofr
  preference 2
```

## prefix

To specify the prefix of the dialed digits for this dial peer, use the **prefix** dial-peer configuration command. Use the **no** form of this command to disable this feature.

**prefix** *string*

**no prefix**

## Syntax Description

*string* Integers representing the prefix of the telephone number associated with the specified dial peer. Valid numbers are 0 through 9, and a comma (,). Use a comma to include a pause in the prefix.

## Default

Null string

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Cisco 3600 series and Cisco MC3810 POTS peers.

Use the **prefix** command to specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix string** value is sent to the telephony interface first, before the telephone number associated with the dial peer.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

## Example

The following example specifies a prefix of “9” and then a pause:

```
dial-peer voice 10 pots
  prefix 9,
```

## Related Commands

**answer-address**  
**destination-pattern**

## req-qos

To specify the desired quality of service to be used in reaching a specified dial peer, use the **req-qos dial-peer configuration** command. Use the **no** form of this command to restore the default value for this command.

```
req-qos {best-effort | controlled-load | guaranteed-delay}
no req-qos
```

## Syntax Description

<b>best-effort</b>	Indicates that Resource Reservation Protocol (RSVP) makes no bandwidth reservation.
<b>controlled-load</b>	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.

**guaranteed-delay** Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

## Default

best-effort. The **no** form of this command restores the default value.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable only to Cisco 3600 series VoIP peers.

Use the **req-qos** command to request a specific quality of service to be used in reaching a dial peer. Like **acc-qos**, when you issue this command, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. Cisco IOS software uses Resource Reservation Protocol (RSVP) to request quality of service guarantees from the network.

## Example

The following example configures **guaranteed-delay** as the desired (requested) quality of service to a dial peer:

```
dial-peer voice 10 voip
  req-qos guaranteed-delay
```

## Related Commands

**acc-qos**

## session protocol

To establish a session protocol for calls between the local and remote routers via the packet network, use the **session protocol dial-peer** configuration command. Use the **no** form of this command to reset the default value for this command.

**session protocol cisco**

**no session protocol**

## Syntax Description

**cisco** Specifies Cisco Session Protocol.

## Default

cisco

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

For Cisco IOS Release 12.0, Cisco Session Protocol (`cisco`) is the only applicable session protocol. This command is applicable only to Cisco 3600 series VoIP peers.

## Example

The following example selects Cisco Session Protocol as the session protocol:

```
dial-peer voice 10 voip
 session protocol cisco
```

## Related Commands

`session target`

## session target

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

### For the Cisco 3600 series Voice over IP:

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

### For the Cisco MC3810 Voice over Frame Relay:

```
session target interface {FR-ATM interface | Serial interface dlci}
no session target interface {FR-ATM interface | Serial interface dlci}
```

### For Cisco MC3810 Voice over ATM:

```
session target interface ATM interface pvc [name | vpi/vci | vci]
no session target interface ATM interface pvc [name | vpi/vci | vci]
```

### For Cisco MC3810 Voice over HDLC:

```
session target interface serial-port-number
no session target interface serial-port-number
```

## Syntax Description

### For the Cisco 3600 series Voice over IP:

<i>ipv4:destination-address</i>	IP address of the dial peer.
<i>dns:host-name</i>	<p>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.</p> <p>(Optional) You can use one of the following three wildcards with this keyword when defining the session target for VoIP peers:</p> <ul style="list-style-type: none"> <li>• <b>\$s\$</b>.—Indicates that the source destination pattern will be used as part of the domain name.</li> <li>• <b>\$d\$</b>.—Indicates that the destination number will be used as part of the domain name.</li> <li>• <b>\$e\$</b>.—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.</li> <li>• <b>\$u\$</b>.—Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.</li> </ul>
<i>loopback:rtp</i>	Indicates that all voice data will be looped back to the originating source. This is applicable for VoIP peers.
<i>loopback:compressed</i>	Indicates that all voice data will be looped back in compressed mode to the originating source. This is applicable for POTS peers.
<i>loopback:uncompressed</i>	Indicates that all voice data will be looped-back in uncompressed mode to the originating source. This is applicable for POTS peers.

### For the Cisco MC3810 Voice over Frame Relay:

<i>interface</i>	Specifies the interface number.
FR-ATM	Specifies a logical Frame Relay-ATM interface on the Cisco MC3810. The valid range for the Frame Relay-ATM <i>interface</i> is 0–20.
Serial	Specifies a serial interface on the Cisco MC3810. The valid range for <i>interface</i> is 0 to 1.
<i>dldci</i>	Specifies the Frame Relay DLCI. The valid range is from 16 to 1007.

### For the Cisco MC3810 Voice over ATM:

<i>interface</i>	Specifies the interface number.
ATM <i>interface</i>	Specifies the ATM interface number on the Cisco MC3810. The only valid number is 0.

<i>name</i>	The PVC name.
<i>vpilvci</i>	The ATM network virtual path identifier (VPI) and virtual channel identifier (VCI) of this PVC.
<i>vci</i>	The ATM network virtual channel identifier (VCI) of this PVC.

**For the Cisco MC3810 Voice over HDLC:**

<i>interface</i>	Specifies the interface number.
<i>serial-port-number</i>	Specifies the serial port number on the Cisco MC3810. The valid range is 0 to 1.

## Default

The default for this command is enabled with no IP address or domain name defined.

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

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 Frame Relay on a Cisco MC3810 with a session target on serial port1 and a DLCI of 200:

```
dial-peer voice 11 vofr
 destination-pattern 13102221111
 session target Serial1 200
```

The following example configures a session target for Voice over ATM on a Cisco MC3810. The session target is sent to ATM interface 0, and for a PVC with a VCI of 20.

```
dial-peer voice 12 voatm
 destination-pattern 13102221111
 session target atm0 pvc 20
```

The following example configures a session target on serial port 0 for Voice over HDLC on a Cisco MC3810:

```
dial-peer voice 13 vohdlc
 destination-pattern 13102221111
 session target serial0
```

## Related Commands

```
destination-pattern
session protocol
```

## show call active voice

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

```
show call active voice
```

## Syntax Description

This command has no arguments or keywords.

## Command Mode

Privileged EXEC

## Usage Guidelines

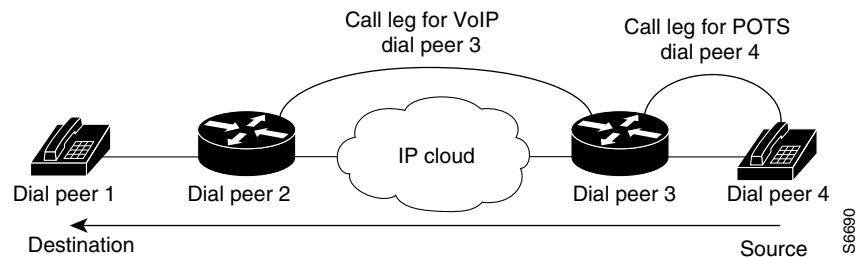
This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

Use the **show call active voice** privileged EXEC command to display the contents of the active call table, which shows all of the calls currently connected through the router.

For each call, there are two call legs, usually a POTS call leg and a VoIP call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. Each dial peer creates a call leg, as shown in Figure 1.

**Figure 1 Call Legs Example**



These two call legs are associated by the connection ID. The connection ID is global across the voice network, so that you can associate two call legs on one router with two call legs on another router, thereby providing an end-to-end view of a call.

## 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= OnTimeRvPayout=0 GapFillWithSilence=0 GapFillWithPrediction=600
GapFillWithInterpolation=0 GapFillWithRedundancy=0 HiWaterPayoutDelay=110
LoWaterPayoutDelay=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 2 provides an alphabetical listing of the fields in this output and a description of each field.

**Table 2** Show Call Active Voice Field Descriptions

Field	Description
ACOM Level	Current ACOM level for the call. This value is 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.
HiWaterPayoutDelay	High water mark Voice Payout 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.
LoWaterPayoutDelay	Low water mark Voice Payout FIFO Delay during the call.
NoiseLevel	Active noise level for the call.
OnTimeRvPayout	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 OnTimeRvPayout 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.

**Table 2** Show Call Active Voice Field Descriptions (continued)

Field	Description
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

```
show call history voice
show dial-peer voice
show num-exp
show voice-port
```

## show call history voice

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

```
show call history voice last number
```

## Syntax Description

`last number` Displays the last calls connected, where the number of calls displayed is defined by the argument *number*. Valid entries for the argument *number* is any number from 1 to 2147483647.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

Use the **show call history voice** privileged EXEC command to display the call history table. The call history table contains a listing of all calls connected through this router in descending time order since Voice over IP was enabled. 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*.

## Example

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

```
router# show call history voice
GENERIC: SetupTime=20405 Index=0 PeerAddress= PeerSubAddress= PeerId=0
PeerIfIndex=0 LogicalIfIndex=0 DisconnectCause=NORMAL DisconnectText= ConnectTime=0
DisconnectTime=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 3 provides an alphabetical listing of the fields in this output and a description of each field.

**Table 3** Show Call History Voice Field Descriptions

Field	Description
ACOMLevel	Average ACOM level for this call. This value is 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.

**Table 3 Show Call History Voice Field Descriptions (continued)**

Field	Description
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.
HiWaterPayoutDelay	High water mark Voice Payout 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.
LoWaterPayoutDelay	Low water mark Voice Payout FIFO Delay during the voice call.
NoiseLevel	Average noise level for this call.
OnTimeRvPayout	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 OnTimeRvPayout 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.
RemoteIPAddress	Remote system IP address for the call.
RemoteUDPPort	Remote system UDP listener port to which voice packets for this call are transmitted.

**Table 3** Show Call History Voice Field Descriptions (continued)

Field	Description
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

```
show call active voice
show dial-peer voice
show num-exp
show voice-port
```

## show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** privileged EXEC command.

### For the Cisco 3600 series:

```
show dial-peer voice [number]
```

### For the Cisco MC3810:

```
show dial-peer voice [number] [summary]
```

## Syntax Description

### For the Cisco 3600 series:

*number* (Optional) A specific dial peer. This option displays configuration for the dial peer identified by the argument *number*. Valid entries are any integers that identify a specific dial peer, from 1 to 32767.

**For the Cisco MC3810:**

<i>number</i>	(Optional) A specific dial peer. This option displays configuration information for a single dial peer identified by the argument <i>number</i> . Valid entries are any integers that identify a specific dial peer, from 1 to 32767.
summary	(Optional) Displays a summary of all voice dial peers.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

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

Use the **show dial-peer voice** privileged EXEC command to display the configuration for all VoIP and POTS dial peers configured for the router. To show configuration information for only one specific dial peer, use the argument *number* to identify the dial peer.

## Example

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

```
router# show dial-peer voice 1
VoiceEncapPeer1
  tag = 1, dest-pat = '+14085291000',
  answer-address = '',
  group = 0, Admin state is up, Operation state is down
  Permission is Both,
  type = pots, prefix = '',
  session-target = '', voice port =
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Accepted Calls = 0, Refused Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
```

The following is sample output from the `show dial-peer voice` command for a VoIP dial peer:

```
router# show dial-peer voice 10
VoiceOverIpPeer10
  tag = 10, dest-pat = `',
  incall-number = `+14087',
  group = 0, Admin state is up, Operation state is down
  Permission is Answer,
  type = voip, session-target = `',
  sess-proto = cisco, req-qos = bestEffort,
  acc-qos = bestEffort,
  fax-rate = voice, codec = g729r8,
  Expect factor = 10, Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Accepted Calls = 0, Refused Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
```

Table 4 explains the fields contained in both of these examples.

**Table 4** Show Dial Peer Voice Field Descriptions

Field	Description
Accepted Calls	Number of calls from this peer accepted since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
Charged Units	Total number of charging units applying to this peer since system startup. The unit of measure for this field is in hundredths of seconds.
codec	Default voice coder rate of speech for this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of measure for this field is in hundredths of seconds.
dest-pat	Destination pattern (telephone number) for this peer.
Expect factor	User-requested Expectation Factor of voice quality for calls via this peer.
fax-rate	Fax transmission rate configured for this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
group	Group number associated with this peer.
ICPIF	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.
Operation state	Operational state of this peer.
Permission	Configured permission level for this peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.

**Table 4 Show Dial Peer Voice Field Descriptions (continued)**

Field	Description
Refused Calls	Number of calls from this peer refused since system startup.
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote router via the IP backbone.
Successful Calls	Number of completed calls to this peer.
tag	Unique dial peer ID number.
VAD	Whether or not voice activation detection (VAD) is enabled for this dial peer.

## Related Commands

show call active voice  
 show call-history voice  
 show num-exp  
 show voice-port

## show dialplan incall number

To pair different voice ports and telephone numbers together for troubleshooting, use the **show dialplan incall number** privileged EXEC command.

**show dialplan incall** *slot-number/subunit-number/port number dial string*

## Syntax Description

<i>slot-number</i>	Slot number in the Cisco router where the voice network module is installed. Valid entries are from 0 to 3, depending on the voice interface card you have installed.
<i>subunit-number</i>	Subunit on the voice network module where the voice port is located. Valid entries are 0 or 1.
<i>port</i>	Voice port. Valid entries are 0 or 1.
<i>dial string</i>	Particular destination pattern (telephone number).

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

Occasionally, an incoming call cannot be matched to a dial peer in the dial peer database. One reason this might occur is that the specified destination cannot be reached via the voice interface through which the incoming call came. Use the **show dialplan incall number** command as a troubleshooting method to resolve the call destination by pairing voice ports and telephone numbers together until there is a match.

## Example

The following example tests whether the telephone extension 57681 can be reached through voice port 1/0/1:

```
show dialplan incall 1/0/1 number 57681
```

## Related Commands

You can use the master indexes or search online to find documentation of related commands.

**show dialplan number**

# show dialplan number

To show which dial peer is reached when a particular telephone number is dialed, use the **show dialplan number** privileged EXEC command.

```
show dialplan number dial string
```

## Syntax Description

*dial string*            Particular destination pattern (telephone number).

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

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

The **show dialplan number** command is used to test if the dial-plan configuration is valid and working as expected.

## Example

The following example displays the dial peer associated with the destination pattern of 54567:

```
router# show dialplan number 51234

Macro Exp.: 14085551234
VoiceOverIpPeer1004
  tag = 1004, destination-pattern = `+1408555....',
  answer-address = `',
  group = 1004, Admin state is up, Operation state is up
  type = voip, session-target = `ipv4:1.13.24.0',
  ip precedence: 0          UDP checksum = disabled
  session-protocol = cisco, req-qos = best-effort,
  acc-qos = best-effort,
  fax-rate = voice, codec = g729r8,
  Expect factor = 10, Icpif = 30,
  VAD = enabled, Poor QOV Trap = disabled
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Accepted Calls = 0, Refused Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
Matched: +14085551234  Digits: 7
Target: ipv4:172.13.24.0
```

Table 5 explains the fields contained in this example.

**Table 5 Show DialPlan Number Field Descriptions**

Field	Description
Macro Exp.	Expected destination pattern for this dial peer.
VoiceOverIpPeer	Identifies the dial peer associated with the destination pattern entered.
tag	Unique dial peer identifying number
destination-pattern	Destination pattern (telephone number) configured for this dial peer
answer-address	Answer address configured for this dial peer.
Admin state	Describes the administrative state of this dial peer.
Operation state	Describes the operational state of the dial peer.
type	Type of dial peer (POTS or VoIP).
session-target	Displays the configured session target (IP address or host name) for this dial peer.
ip precedence	Displays the numeric value for the IP Precedence configured for this dial peer.
UDP checksum	Indicates the status of the UDP checksum feature.
session-protocol	Session protocol to be used for Internet calls between local and remote router via the IP backbone.
req-qos	Configured requested quality of service for calls for this dial peer.
acc-qos	Configures acceptable quality of service for calls for this dial peer.
fax-rate	Configured facsimile transmission speed for with this dial peer.
codec	CODEC type configured for this dial peer.
Expect factor	Configured value at which the system will generate an SMTP message alerting that the voice quality has dropped.
Icpif	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
VAD	Whether or not voice activation detection (VAD) is enabled for this dial peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Connect Time	Unit of measure indicating the call connection time associated with this dial peer.
Charged Units	Number of call units charged to this dial peer.
Successful Calls	Number of completed calls to this peer since system startup.
Failed Calls	Number of uncompleted (failed) calls to this peer since system startup.
Accepted Calls	Number of calls from this peer accepted since system startup.
Refused Calls	Number of calls from this peer refused since system startup.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.

**Table 5** Show DialPlan Number Field Descriptions (continued)

Field	Description
Matched	Destination pattern matched for this dial peer.
Target	Matched session target (IP address or host name) for this dial peer.

## Related Commands

show dialplan incall number

## 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

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

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 6 explains the fields in the sample output.

**Table 6** *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

show call active voice  
 show call history voice  
 show dial-peer voice  
 show voice-port

## show voice call

To show the call status for all voice ports on the Cisco MC3810, use the **show voice call** privileged EXEC command.

**show voice call** [summary]

## Syntax Description

**summary** (Optional) Specifies to show a summary of the status instead of the full detailed report.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

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

- Tandem switch
- End-to-end call manager
- Call processing state machine
- Protocol state machine

## Example

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

```
router# show voice call summary
1/1 (orig): eecm = ST_DIGIT_COLLECT, LFXS= call_progress, CPD= failure_cont
1/2 ( ): eecm = IDLE, LFXS= idle, CPD= idle
1/3 ( ): eecm = IDLE, LFXS= idle, CPD= idle
1/4 ( ): eecm = IDLE, LFXO= idle, CPD= idle
1/5 ( ): eecm = IDLE, LEM= idle, CPD= idle
1/6 ( ): eecm = IDLE, LEM= idle, CPD= idle
```

Table 7 explains the fields in the sample output.

**Table 7 Show Voice Call Field Descriptions**

Field	Description
(orig)	Indicates the call is originating on the voice port.
eecm	Status of the End-to-End Call Manager.
LFXS	Status of the FXS line.
CPD	Status of the Call Processing Data.
LFXO	Status of the FXO line.
LEM	Status of the E&M line.

## Related Commands

```
show dial-peer voice
show voice dsp
show voice-port
```

## show voice dsp

To show the current status of all DSP voice channels, use the **show voice dsp** privileged EXEC command.

```
show voice dsp
```

## Syntax Description

This command has no arguments or keywords.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

## Example

The following is sample output from the `show voice dsp` command for the Cisco MC3810:

```
router# show voice dsp
DSP# 0, channel# 0 G729A BUSY
DSP# 0, channel# 1 G729A BUSY
DSP# 1, channel# 2 FAX IDLE
DSP# 1, channel# 3 FAX IDLE
DSP# 2, channel# 4 NONE BAD
DSP# 2, channel# 5 NONE BAD
DSP# 3, channel# 6 NONE BAD
DSP# 3, channel# 7 NONE BAD
DSP# 4, channel# 8 NONE BAD
DSP# 4, channel# 9 NONE BAD
DSP# 5, channel# 10 NONE BAD
DSP# 5, channel# 11 NONE BAD
```

Table 8 explains the fields in the sample output.

**Table 8** Show Voice DSP Field Descriptions

Field	Description
DSP	Number of the DSP
Channel	Number of the channel and its status.

## Related Commands

show dial-peer voice  
 show voice call summary  
 show voice-port

## show voice port

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

**For the Cisco 3600 series:**

```
show voice port slot-number/subunit-number/port
```

**For the Cisco MC3810:**

```
show voice port [slot/port] [summary]
```

## Syntax Description

### For the Cisco 3600 series:

<i>slot-number</i>	Slot number in the Cisco router where the voice interface card is installed. Valid entries are from 0 to 3, depending on the slot where it has been installed.
<i>subunit-number</i>	Subunit on the voice interface card where the voice port is located. Valid entries are 0 or 1.
<i>port</i>	Voice port number. Valid entries are 0 or 1.

### For the Cisco MC3810:

<i>slot/port</i>	(Optional) Displays information for only the voice port you specify with the <i>slot/port</i> designation.  <i>slot</i> specifies the slot number in the Cisco router where the voice interface card is installed. The only valid entry is 1.  <i>port</i> specifies the voice port number. Valid ranges are as follows: Analog voice ports: from 1 to 6. Digital voice port: Digital T1: from 1 to 24. Digital E1: from 1 to 15, and from 17 to 31.
<i>summary</i>	(Optional) Display a summary of all voice ports.

## Command Mode

Privileged EXEC

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

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

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` command for an E&M voice port on the Cisco 3600 series:

```
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 voice port on the Cisco 3600 series:

```
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 dB
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 voice port on the 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 summary` command for all voice ports on a Cisco MC3810 with an analog voice module (AVM):

```
router# show voice port summary
```

PORT	SIG-TYPE	ADMIN	OPER	IN-STATUS	OUT-STATUS	CODEC	VAD	IN GAIN	OUT ATTN	ECHO CANCEL
1/1	fxs-ls	up	up	on-hook	idle	729a	n	0	0	y
1/2	fxs-ls	up	up	on-hook	idle	729a	n	0	0	y
1/3	e&m-wnk	up	up	idle	idle	729a	n	0	0	y
1/4	e&m-wnk	up	up	idle	idle	729a	n	0	0	y
1/5	fxo-ls	up	up	idle	on-hook	729a	n	0	0	y
1/6	fxo-ls	up	up	idle	on-hook	729a	n	0	0	y

Table 9 explains the fields in the sample output.

**Table 9 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.

**Table 9 Show Voice Port Field Descriptions (continued)**

Field	Description
Nominal Playout Delay	The amount of time the Cisco MC3810 DSP waits before starting to play out the voice packets from the DSP buffer.
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 that 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

```

show call active voice
show call history voice
show dial-peer voice
show num-exp

```

## shutdown (dial-peer)

To change the administrative state of the selected dial peer from up to down, use the **shutdown** dial-peer configuration command. Use the **no** form of this command to change the administrative state of this dial peer from down to up.

```
shutdown
no shutdown
```

### Syntax Description

This command has no arguments or keywords.

### Default

```
no shutdown
```

### Command Mode

Dial-peer configuration

### Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

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

When a dial peer is shut down, you cannot initiate calls to that peer. This command is applicable to both Cisco 3600 series and Cisco MC3810 VoIP and POTS peers.

### Example

The following example changes the administrative state of voice telephony dial peer 10 to down:

```
configure terminal
dial-peer voice 10 pots
shutdown
```

## shutdown (DS1 Link)

To shutdown a DS1 link (send a Blue Alarm), use the **shutdown** controller configuration command. Use the **no** form of the command to activate the DS1 (cancel the sending of the Blue Alarm).

```
shutdown
no shutdown
```

### Syntax Description

This command has no arguments or keywords.

## Default

no shutdown

## Command Mode

Controller configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

## Example

The following example shuts down a DS 1 link on controller T1 0:

```
controller T1 0
 shutdown
```

# snmp enable peer-trap poor-qov

To generate poor quality of voice notification for applicable calls associated with VoIP dial peers, use the **snmp enable peer-trap poor-qov** dial-peer configuration command. Use the **no** form of this command to disable this notification.

```
snmp enable peer-trap poor-qov
no snmp enable peer-trap poor-qov
```

## Syntax Description

This command has no arguments or keywords.

## Default

Disabled

## Command Mode

Dial-peer configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command is applicable only to VoIP peers on the Cisco 3600 series; VoFR, VoATM, VoHDLC peers on the Cisco MC3810.

Use the **snmp enable peer-trap poor-qov** command to generate poor quality of voice notifications for applicable calls associated with this dial peer. If you have an SNMP manager that will use SNMP messages when voice quality drops, you might want to enable this command. Otherwise, you should disable this command to reduce unnecessary network traffic.

## Example

The following example enables poor quality of voice notifications for calls associated with VoIP dial peer 10:

```
dial-peer voice 10 voip
 snmp enable peer-trap poor-qov
```

## Related Commands

```
snmp-server enable trap voice poor-qov
snmp-server enable traps
snmp trap link-status
```

## tdm-group

To configure a list of time slots for creating clear channel groups (pass-through) for Time Division Multiplexing (TDM) cross-connect, use the **tdm-group** controller configuration command. Use the **no** form of this command to delete a clear channel group.

```
tdm-group tdm-group-no timeslot timeslot-list [type {e&m | fxs [loop-start | ground-start] | fxo [loop-start | ground-start] | fxs-melcas | fxo-melcas | e&m-melcas}]
no tdm-group tdm-group-no
```

## Syntax Description

<i>tdm-group-no</i>	Time Division Multiplexing (TDM) group number.
<i>timeslot</i>	Timeslot number.
<i>timeslot-list</i>	Timeslot list. The valid range is from 1-24 for T1, and from 1-15 and 17-31 for E1.
<i>type</i>	(Optional) (Valid only when the <b>mode cas</b> command is enabled.) Specifies the voice signaling type of the voice port. If configuring a TDM group for data traffic only, do not specify the <b>type</b> option. Choose from one of the following options:

- **e&m**—for E&M signaling
- **fxo**—for Foreign Exchange Office signaling (optionally, you can also specify loop-start or ground-start)
- **fxs**—for Foreign Exchange Station signaling (optionally, you can also specify loop-start or ground-start)
- **e&m-melcas**—for E&M Mercury Exchange Limited (MEL) Channel Associated Signaling
- **fxs-melcas**— for Foreign Exchange Station Mercury Exchange Limited (MEL) Channel Associated Signaling
- **fxo-melcas**—for Foreign Exchange Office Mercury Exchange Limited (MEL) Channel Associated Signaling

The melcas options apply only to E1 lines and are used primarily in the United Kingdom.

## Default

No TDM group is configured.

## Command Mode

Controller configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.



### Note

Channel groups, CAS voice groups, and TDM groups all use group numbers. All group numbers configured for channel groups, CAS voice groups and TDM groups must be unique on the local Cisco MC3810 concentrator. For example, you cannot use the same group number for a channel group and for a TDM group.

## Example

The following example configures TDM group number 20 on controller T1 1 to support FXO ground-start:

```
controller T1 1
 mode cas
  tdm-group 20 20 type fxs ground-start
```

## Related Commands

mode

## vad (dial peer)

To enable voice activity detection (VAD) for the calls using this dial peer, use the `vad dial-peer` configuration command. Use the `no` form of this command to disable VAD.

```
vad
no vad
```

### Syntax Description

This command has no arguments or keywords.

### Default

Enabled

### Command Mode

Dial-peer configuration

### Usage Guidelines

This command first appeared in Cisco IOS Release 11.3(1)T.

This command applies to Voice over IP on the Cisco 3600 series.

Use the `vad` command to enable voice activity detection. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality is slightly degraded but the connection monopolizes much less bandwidth. If you use the `no` form of this command, VAD is disabled and voice data is continuously transmitted to the IP backbone.



**Note**

On the Cisco MC3810, VAD is assigned to the voice port using the `vad voice-port` configuration command.

### Example

The following example enables VAD:

```
dial-peer voice 10 voip
vad
```

### Related Commands

`comfort-noise`

## voice-encap

To define the data segmentation size on an HDLC interface to support voice over HDLC, use the `voice-encap` interface configuration command. Use the `no` form of this command to delete the setting.

```
voice-encap size
```

`no voice-encap size`

## Syntax Description

*size* The size of the data segmentation. The valid range is from 8 to 1600.

## Default

No data segmentation size is defined.

## Command Mode

Interface configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over HDLC on the Cisco MC3810.

## Example

The following example configures serial interface 1 to support a data segmentation size of 64 for Voice over HDLC:

```
interface serial 1
 voice-encap 64
```

## voice-group

To configure a list of timeslots for voice channel associated signaling (CAS) on the T1/E1 controller on the Cisco MC3810, use the **voice-group** controller configuration command. Use the **no** form of this command to delete the CAS group.

```
voice-group voice-group-no timeslots timeslot-list type {e&m-immediate | e&m-delay |
e&m-wink | e&m-melcas | fxs-ground-start | fxs-loop-start | fxs-melcas |
fxo-ground-start | fxo-loop-start | fxo-melcas}
no voice-group voice-group-no timeslots timeslot-list
```

## Syntax Description

*voice-group-no* Number to identify the voice group. The valid range is from 0 to 23.

*timeslot timeslot-list* A list of timeslots that makes up the CAS group. The valid range is from 1 to 24 for T1, and from 1 to 15 and 17 to 31 for E1.

<b>type</b>	<p>The type of voice interface:</p> <ul style="list-style-type: none"><li>• <b>e&amp;m-immediate</b>—for E&amp;M immediate</li><li>• <b>e&amp;m-delay</b>—for E&amp;M delay</li><li>• <b>e&amp;m-wink</b>—for E&amp;M wink</li><li>• <b>e&amp;m-melcas</b>—for E&amp;M Mercury Exchange Limited (MEL) Channel Associated Signaling (equivalent to CEPT)</li><li>• <b>fxs-ground-start</b>—for Foreign Exchange Station ground-start</li><li>• <b>fxs-loop-start</b>—for Foreign Exchange Station loop-start</li><li>• <b>fxs-melcas</b>— for Foreign Exchange Station Mercury Exchange Limited (MEL) Channel Associated Signaling</li><li>• <b>fxo-ground-start</b>—for Foreign Exchange Office.ground-start</li><li>• <b>fxo-loop-start</b>—for Foreign Exchange Office.loop-start</li><li>• <b>fxo-melcas</b>—for Foreign Exchange Office Mercury Exchange Limited (MEL) Channel Associated Signaling</li></ul> <p>The melcas options are used primarily in the United Kingdom.</p>
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## Default

No voice group is configured.

## Command Mode

Controller configuration

## Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, and Voice over HDLC on the Cisco MC3810.

This command is only available if the **mode cas** command is enabled.

## Example

The following example configures a voice group on controller T1 0 on a Cisco MC3810:

```
controller T1 0
 mode cas
 voice-group 10 timeslots10 64
```

## Related Commands

**channel-group**  
**mode (Voice over ATM)**  
**tdm-group**

## voice local-bypass

To directly cross-connect local POTS calls without going through a Digital Signal Processor (DSP), use the `voice local-bypass` global configuration command. Use the `no` form of this command to cancel the voice local-bypass disable operation.

```
voice local-bypass
no voice local-bypass
```

### Syntax Description

This command has no arguments or keywords.

### Default

Disabled

### Command Mode

Global configuration

### Usage Guidelines

This command first appeared in Cisco IOS Release 11.3 MA.

This command applies to Voice over Frame Relay, Voice over ATM, Voice over HDLC, and Frame Relay-ATM Interworking on the Cisco MC3810.

This command allows you to pass uncompressed voice traffic for local POTS calls.

### Example

The following example configures the Cisco MC3810 to directly cross-connect local calls without going through a Digital Signal Processor (DSP):

```
voice local-bypass
```