

Configuring Voice Ports

This chapter describes how to configure voice ports for both the Cisco 3600 series (for Voice over IP) and for the Cisco MC3810 (for Voice over Frame Relay, Voice over ATM, and Voice over HDLC). For a description of the voice port commands, refer to the “Voice Port Commands” chapter in the *Voice, Video, and Home Applications Command Reference*.

Voice Ports on the Cisco 3600 Series

The Cisco 3600 currently provides only analog voice ports for its implementation of Voice over IP. The type of signaling associated with these analog voice ports depend on the interface module installed into the device. The Cisco 3600 series router supports either a two-port or four-port voice network module (VNM); VNMs can hold either two or four voice interface cards (VICs).

Each VIC is specific to a particular signaling type; therefore, VICs determine the type of signaling for the voice ports on that particular VNM. This means that even though VNMs can hold multiple VICs, each VIC on a VNM must conform to the same signaling type. For more information about the physical characteristics of VNMs and VICs or how to install them, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your VNM.

Voice ports on the Cisco 3600 series support three basic voice signaling types:

- FXO—Foreign Exchange Office interface. The FXO interface is an RJ-11 connector that allows a connection to be directed at the PSTN’s central office (or to a standard PBX interface, if the local telecommunications authority permits). This interface is of value for off-premise extension applications.
- FXS—The Foreign Exchange Station interface. This interface is an RJ-11 connector that allows connection for basic telephone equipment, keysets, PBXs, and supplies ring, voltage, and dial tone.
- E&M—The “Ear and Mouth” interface (or “RecEive and TransMit”) interface. This interface is an RJ-48 connector that allows connection for PBX trunk lines (tie lines). It is a signaling technique for two-wire and four-wire telephone and trunk interfaces.

Configuring Voice Ports on the Cisco 3600 Series

In general, voice port commands define the characteristics associated with a particular voice port signaling type. Under most circumstances, the default voice port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice port values configured, depending on the specifications of the devices in your telephony network.

Voice Ports for the Cisco 3600 Series Configuration Task List

Perform the following tasks to configure the voice ports on the Cisco 3600 series:

- Configuring FXO or FXS Voice Ports
- Fine-Tuning FXO and FXS Voice Ports
- Configuring E&M Voice Ports
- Fine-Tuning E&M Voice Ports
- Activating the Voice Port

Configuring FXO or FXS Voice Ports

Under most circumstances the default voice port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. The first two tasks are required; the third task is optional.

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 Configure the following mandatory voice-port parameters:
 - (a) Dial type (FXO only)
 - (b) Signal type
 - (c) Call progress tone
 - (d) Ring frequency (FXS only)
 - (e) Ring number (FXO only)
- 3 Configure one or more of the following optional voice-port parameters:
 - (a) PLAR connection mode
 - (b) Music-threshold
 - (c) Description
 - (d) Comfort noise (if VAD is activated—VAD is a dial peer command)

To configure FXO and FXS voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot-number/subunit-number/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	dial-type { dtmf pulse }	(For FXO ports only) Select the appropriate dial type for out-dialing.
4	signal { loop-start ground-start }	Select the appropriate signal type for this interface.
5	cptone <i>country</i>	Select the appropriate voice call progress tone for this interface. The default for this command is us . For a list of supported countries, refer to the <i>Voice, Video, and Home Applications Command Reference</i> .

Step	Command	Purpose
6	ring frequency { 25 50 }	(For FXS ports only) Select the appropriate ring frequency (in Hertz) specific to the equipment attached to this voice port.
7	ring number <i>number</i>	(For FXO ports only) Specify the maximum number of rings to be detected before answering a call.
8	connection plar <i>string</i>	(Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The <i>string</i> value specifies the destination telephone number.
9	music-threshold <i>number</i>	(Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from -70 to -30.
10	description <i>string</i>	(Optional) Attach descriptive text about this voice port connection.
11	comfort-noise	(Optional) Specify that background noise will be generated.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice-port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document, *Voice Network Module and Voice Interface Card Configuration Note*, that came with your voice network module.

Fine-Tuning FXO and FXS Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for FXO and FXS voice ports, perform the following tasks:

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 For each of the following parameters, select the appropriate value:
 - (a) Input gain
 - (b) Output attenuation
 - (c) Echo cancel coverage
 - (d) Non-linear processing
 - (e) Initial digit timeouts
 - (f) Interdigit timeouts
 - (g) Timing other than timeouts

To fine-tune FXO or FXS voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot-number/subunit-number/port</i>	Identify the voice-port you want to configure and enter voice-port configuration mode.
3	input gain <i>value</i>	Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14.
4	output attenuation <i>value</i>	Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
5	echo-cancel enable	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	echo-cancel coverage <i>value</i>	Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32.
7	non-linear	Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
8	timeouts initial <i>seconds</i>	Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120.
9	timeouts interdigit <i>seconds</i>	Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120.

Step	Command	Purpose
10	timing digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100. The default is 100.
11	timing inter-digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500. The default is 100.
12	timing pulse-digit <i>milliseconds</i>	(FXO ports only) If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from 10 to 20. The default is 20.
13	timing pulse-inter-digit <i>milliseconds</i>	(FXO ports only) If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000. The default is 500.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Configuring E&M Voice Ports

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your IP network. E&M voice-port values must match those specified by the particular PBX device to which it is connected.

Note E&M voice-port values must match those of the PBX to which it is connected. Refer to the documentation that came with your specific PBX for the appropriate E&M voice-port configuration command values.

To configure an E&M voice port, perform the following tasks. The first two tasks are required; the third task is optional.

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 For each of the following required parameters, select the appropriate parameter value:
 - (a) Dial type
 - (b) Signal type
 - (c) Call progress tone
 - (d) Operation
 - (e) Type
 - (f) Impedance
- 3 Select one or more of the following optional parameters:
 - (a) Connection mode
 - (b) Music-threshold

- (c) Description
- (d) Comfort tone (if VAD is activated)

To configure E&M voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot-number/subunit-number/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	dial-type { dtmf pulse }	Select the appropriate dial type for out-dialing.
4	signal { wink-start immediate delay-dial }	Select the appropriate signal type for this interface.
5	cptone { australia brazil china finland france germany japan northamerica unitedkingdom }	Select the appropriate voice call progress tone for this interface.
6	operation { 2-wire 4-wire }	Select the appropriate cabling scheme for this voice port.
7	type { 1 2 3 5 }	Select the appropriate E&M interface type. Type 1 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground Type 2 indicates the following lead configuration: E—output, relay to SG M—input, referenced to ground SB—feed for M, connected to -48V SG—return for E, galvanically isolated from ground Type 3 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground SB—connected to -48V SG—connected to ground Type 5 indicates the following lead configuration: E—output, relay to ground M—input, referenced to -48V
8	impedance { 600c 600r 900c complex1 complex2 }	Specify a terminating impedance. This value must match the specifications from the telephony system to which this voice port is connected.
9	connection plar <i>string</i>	(Optional) Specify the private line auto ringdown (PLAR) connection, if this voice port is used for a PLAR connection. The <i>string</i> value specifies the destination telephone number.
10	music-threshold <i>number</i>	(Optional) Specify the threshold (in decibels) for on-hold music. Valid entries are from -70 to -30.
11	description <i>string</i>	(Optional) Attach descriptive text about this voice port connection.
12	comfort-noise	(Optional) Specify that background noise will be generated.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice-port command** to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and/or type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the installation document that came with your voice network module.

Fine-Tuning E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain, and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for E&M voice ports, perform the following tasks:

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 For each of the following parameters, select the appropriate value:
 - (a) Input gain
 - (b) Output attenuation
 - (c) Echo cancel coverage
 - (d) Non-linear processing
 - (e) Initial digit timeouts
 - (f) Interdigit timeouts
 - (g) Timing other than timeouts

To fine-tune E&M voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot-number/subunit-number/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	input gain <i>value</i>	Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14.
4	output attenuation <i>value</i>	Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
5	echo-cancel enable	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	echo-cancel coverage <i>value</i>	Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32.
7	non-linear	Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
8	timeouts initial <i>seconds</i>	Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits. Valid entries for this command are from 0 to 120.
9	timeouts interdigit <i>seconds</i>	Specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit. Valid entries for this command are from 0 to 120.
10	timing clear-wait <i>milliseconds</i>	Specify the minimum amount of time between the inactive seizure signal and the call being cleared. Valid entries for clear-wait are from 200 to 2000 milliseconds.
11	timing delay-duration <i>milliseconds</i>	Specify the delay signal duration for delay dial signaling. Valid entries for delay-duration are from 100 to 5000 milliseconds.
12	timing delay-start <i>milliseconds</i>	Specify the minimum delay time from outgoing seizure to outdial address. Valid entries for delay-start are from 20 to 2000 milliseconds.
13	timing dial-pulse min-delay <i>milliseconds</i>	Specify the time between generation of wink-like pulses. Valid entries for dial-pulse min-delay are from 0 to 5000 milliseconds.
14	timing digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF digit signal duration. Valid entries for digit are from 50 to 100 milliseconds.
15	timing inter-digit <i>milliseconds</i>	If the voice-port dial type is DTMF, specify the DTMF inter-digit duration. Valid entries for inter-digit are from 50 to 500 milliseconds.
16	timing pulse <i>pulse-per-second</i>	If the voice-port dial type is pulse, specify the pulse dialing rate. Valid entries for pulse are from 10 to 20 pulses per second.

Step	Command	Purpose
17	timing pulse-inter-digit <i>milliseconds</i>	If the voice-port dial type is pulse, specify the pulse dialing inter-digit timing. Valid entries for pulse-inter-digit are 100 to 1000 milliseconds.
18	timing wink-duration <i>milliseconds</i>	Specify the maximum wink signal duration. Valid entries for wink-duration are from 100 to 400 milliseconds.
19	timing wink-wait <i>milliseconds</i>	Specify the maximum wink-wait duration for a wink start signal. Valid entries for wink-wait are from 100 to 5000 milliseconds.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. In fact it is a good idea to cycle the port—meaning to shut the port down and then bring it online again.

To activate a voice port, use the following commands in voice-port configuration mode:

Command	Purpose
no shutdown	Activate the voice port.

To cycle a voice port, use the following command in voice-port configuration mode:

Step	Command	Purpose
1	shutdown	Deactivate the voice port.
2	voice-port <i>slot-number/subunit-number/port</i>	Identify the voice port you want to activate and enter the voice-port configuration mode.
3	no shutdown	Activate the voice port.

Note If you are not going to use a voice port, shut it down.

Voice Ports on the Cisco MC3810

The Cisco MC3810 hardware features two models, each providing different configuration options for voice ports:

- Six analog voice interfaces— The Cisco MC3810 version with the analog voice module (AVM) supports up to six analog voice personality modules (APMs) with each voice module supporting a single signaling type (see below). Each voice personality module maps to a single analog voice port. The FXO, FXS, and E&M voice modules can be installed in any combination.
- One digital voice module (DVM)— The Cisco MC3810 version with the DVM provides support for up to 24 voice channels, one for each voice port. Depending on whether the controller is T1 or E1, different DS0 voice channels are used. The DVM supports Channel Associated Signaling

(CAS) for the following types: FXO, FXS, and E&M. For E&M signaling, the DVM also supports E1 Mercury Exchange Limited Channel Associated Signaling (MELCAS), a standard used primarily in the United Kingdom.

The Cisco MC3810 voice ports provide support for three basic voice signaling formats:

- **FXO**—The Foreign Exchange Office interface. This interface allows a connection to be directed to the PSTN’s central office. The FXO interface also allows a connection to be directed to a standard PBX interface if the local telecommunications authority permits. This interface is of value for off-premise extension applications.
- **FXS**—The Foreign Exchange Station interface. This interface allows connection for basic telephone equipment and keysets, and supplies ring, voltage, and dial tone.
- **E&M**—The “Ear and Mouth” (or “RecEive and TransMit”) interface allows connection for PBX trunk lines (tie lines). E&M is a signaling technique for two-wire and four-wire telephone and trunk interfaces.

Configuring Voice Ports on the Cisco MC3810

In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data using the Cisco MC3810. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network.

Table 10 lists the valid slot and port numbers for the different voice interfaces.

Table 10 Voice Interface Slot and Port Number

Interface Type	Slot	Valid Port Numbers
Analog voice module (AVM)	1	1–6
Digital voice module (DVM)	1	Digital T1: 1–24 Digital E1: 1–15 and 17–31
Multiflex Trunk (MFT)	0	Digital T1: 1–24 Digital E1: 1–15 and 17–31

Note The voice-port number designations start with 1. Unlike serial port interfaces and interfaces on other Cisco products, there is no *port 0* for voice ports.

Voice Ports for the Cisco MC3810 Configuration Task List

Perform the following tasks to configure the voice ports on the Cisco MC3810:

- Configuring FXO or FXS Voice Ports
- Fine-Tuning FXO and FXS Voice Ports
- Configuring E&M Voice Ports
- Fine-Tuning E&M Voice Ports
- Activating the Voice Port

Configuring FXO or FXS Voice Ports

Under most circumstances the default voice-port values are adequate for both FXO and FXS voice ports. If you need to change the default configuration for these voice ports, perform the following tasks. The first two tasks are required; the third task is optional.

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 Configure the following mandatory voice-port parameters:
 - (a) Connection
 - (b) Dial type
 - (c) Signal type
 - (d) CODEC
 - (e) Call progress tone
- 3 Configure one or more of the following optional voice-port parameters:
 - (a) Description
 - (b) VAD
 - (c) Comfort noise (if VAD is activated)

To configure FXO and FXS voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	voice-port <i>slot/port</i>	Enter voice-port configuration mode. The slot number for analog voice ports on the Cisco MC3810 is always 1. There is no port 0 for voice ports.
2	connection { plar tie-line plar-opx } <i>string</i>	Configure the voice-port connection mode type and the destination telephone number. The plar value is used for Private Line Auto Ringdown (PLAR) connections. The tie-line value is used for a tie-line connection to a PBX. The plar-opx value is used for PLAR Off-Premises eXtension, to allow the local voice port to provide a local response before the remote voice port receives an answer.
3	dial-type { pulse dtmf }	Configure the voice-port dial-type. The default is dtmf (FXO only).
4	signal { loop-start ground-start }	Configure the signaling type for analog FXO and FXS voice ports. The default is loop-start.

Step	Command	Purpose
5	<code>codec {g729r8 g729ar8 g726r32 g711alaw g711ulaw}</code>	<p>Configure the voice-port compression mode. The <code>g729ar8</code> value is the default and is recommended.</p> <p>The <code>g729ar8</code> compression mode can support a maximum of 24 simultaneously active on-net voice calls while the <code>g729r8</code> value can only support a maximum of 12. The <code>g729</code> compression modes have a nominal data rate of 8 kbps.</p>
6	<code>compand-type {u-law a-law}</code>	Configure the companding standard used to convert between analog and digital signals in PCM systems. This command applies to digital voice-ports only.
7	<code>cptone country</code>	<p>Configure the appropriate call progress tone for the local region.</p> <p>The default for this command is northamerica. For a list of supported countries, refer to the <i>Voice, Video, and Home Applications Command Reference</i>.</p>
8	<code>description string</code>	(Optional) Enter a string description for the voice port. The string describes the voice port in displays. You can use the description command to note the voice port's location or use.
9	<code>vad</code>	(Optional) Enable voice activity detection (VAD).
10	<code>voice confirmation-tone</code>	(Optional) If the voice port is configured for connection plar-opx for Off-Premises eXtension, disable the two-beep confirmation tone that a caller hears when picking up the handset.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice port** command to verify that the data configured is correct.
- Use the **show voice dsp** command to verify the current status of all DSP voice channels.
- Use the **show voice call** summary command to verify the call status for all voice ports.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- Check the dial-peer configuration.
- Check the Frame Relay, ATM, or HDLC configuration.
- Check to see if the voice network module has been correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*.

Fine-Tuning FXO and FXS Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain and output attenuation for FXO or FXS voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning applicable for both FXO and FXS voice ports, perform the following tasks:

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 For each of the following parameters, select the appropriate value:
 - (a) Input gain
 - (b) Output attenuation
 - (c) Echo cancel coverage
 - (d) Initial digit timeouts
 - (e) Interdigit timeouts
 - (f) Timing other than timeouts
 - (g) Impedance (FXO voice ports only)
 - (h) Ring number (FXO voice ports only)
 - (i) Ring frequency (FXS voice port only)
 - (j) Ring cadence (FXS voice port only)

To fine-tune FXO or FXS voice ports, perform the following steps beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	input gain <i>value</i>	Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14.
4	output attenuation <i>value</i>	Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
5	echo-cancel enable	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	echo-cancel coverage <i>value</i>	Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32.
7	timeouts initial <i>seconds</i>	Configure the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds.
8	timeouts interdigit <i>seconds</i>	Configure the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds.
9	timing digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100.
10	timing inter-digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500 milliseconds. The default is 100.
11	timing pulse-digit <i>milliseconds</i>	If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from 10 to 20 milliseconds. The default is 20.
12	timing pulse-inter-digit <i>milliseconds</i>	If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000 milliseconds. The default is 500.
13	impedance { 600r 600c 900r 900c }	(For FXO ports only) Configure the impedance. The default is 600 ohms real.
14	ring number <i>number</i>	(For FXO ports only) Configure the number of rings detected before a connection is closed on the FXO port.

Step	Command	Purpose
15	ring frequency <i>number</i>	(For FXS ports only) Specify the local ring frequency for the FXS voice port. The <i>number</i> value should be set to either 20 or 30.
16	ring cadence [on1 off1] [on2 off2] [on3 off3] [on4 off4] [on5 off5] [on6 off6]	(For FXS only) Specify the local ring cadence for the FX voice port. Using this command, specify the on and off pulses for the ring. The ring cadence differs depending on the local region. The units are in 100-millisecond units.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Configuring E&M Voice Ports

Unlike FXO and FXS voice ports, the default E&M voice-port parameters most likely will not be sufficient to enable voice data transmission over your network.

Note E&M voice-port values must match those of the PBX to which it is connected. Refer to the documentation that came with your specific PBX for the appropriate E&M voice-port configuration command values.

To configure an E&M voice port on the Cisco MC3810, perform the following tasks. The first two tasks are required; the third task is optional.

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 Configure the appropriate value for each of the following required parameters:
 - (a) Connection
 - (b) Dial type
 - (c) Cabling scheme
 - (d) Interface type
 - (e) Signal type
 - (f) CODEC
 - (g) Call progress tone
- 3 Configure one or more of the following optional parameters:
 - (a) Description
 - (b) VAD
 - (c) Comfort noise (if VAD is activated)

To configure E&M voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	connection { plar tie-line plar-opx } <i>string</i>	Configure the voice-port connection mode type and the destination telephone number. The plar value is used for Private Line Auto Ringdown (PLAR) connections. The tie-line value is used for a tie-line connection to a PBX. The plar-opx value is used for PLAR Off-Premises eXtension, to allow the local voice port to provide a local response before the remote voice port receives an answer.
4	dial-type dtmf	Select the appropriate dial type for out-dialing. For E&M voice ports, the only available choice is DTMF.
5	operation { 2-wire 4-wire }	Select the appropriate cabling scheme for this voice port.
6	type { 1 2 3 5 }	Select the appropriate E&M interface type. Type 1 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground Type 2 indicates the following lead configuration: E—output, relay to SG M—input, referenced to ground SB—feed for M, connected to -48V SG—return for E, galvanically isolated from ground Type 3 indicates the following lead configuration: E—output, relay to ground M—input, referenced to ground SB—connected to -48V SG—connected to ground Type 5 indicates the following lead configuration: E—output, relay to ground M—input, referenced to -48V.
7	signal { wink-start immediate delay-dial }	Configure the signaling type for E&M voice ports. The default is wink-start.
8	codec { g729r8 g729ar8 }	Configure the voice-port compression mode. The g729ar8 value is the default and is recommended. The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls while the g729r8 value can only support a maximum of 12. Both compression modes have a nominal data rate of 8 kbps.
9	compand-type { u-law a-law }	Configure the companding standard used to convert between analog and digital signals in PCM systems. This command applies to digital voice-posts only.

Step	Command	Purpose
10	cptone <i>country</i>	Configure the appropriate call progress tone for the local region. The default for this command is northamerica . For a list of supported countries, refer to the <i>Voice, Video, and Home Applications Command Reference</i> .
11	description <i>string</i>	(Optional) Attach descriptive text about this voice port connection.
12	vad	(Optional) Enable voice activity detection (VAD).
13	voice confirmation-tone	(Optional) If the voice port is configured for connection plar-opx for Off-Premises eXtension, disable the two-beep confirmation tone that a caller hears when picking up the handset.

Validation Tips

You can check the validity of your voice-port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and check for a dial tone.
- If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.
- Use the **show voice port** command to verify that the data configured is correct.
- Use the **show voice dsp** command to verify the current status of all DSP voice channels.
- Use the **show voice call** summary command to verify the call status for all voice ports.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.
- Use the **show voice port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- Make sure that the values pertaining to your specific PBX setup, such as timing and type, are correct.
- Check to see if the voice network module has been correctly installed. For more information, refer to the Cisco MC3810 *Multiservice Concentrator Hardware Installation Guide*.

Fine-Tuning E&M Voice Ports

Depending on the specifics of your particular network, you may need to adjust voice parameters involving timing, input gain and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice-port tuning commands.

Note In most cases, the default values for voice-port tuning commands will be sufficient.

To configure voice-port tuning for E&M voice ports, perform the following tasks:

- 1 Identify the voice port and enter the voice-port configuration mode.
- 2 Select the appropriate value for each of the following parameters:
 - (a) Input gain
 - (b) Output attenuation
 - (c) Echo cancel coverage
 - (d) Non-linear processing
 - (e) Initial digit timeouts
 - (f) Interdigit timeouts
 - (g) Timing other than timeouts

To fine-tune E&M voice ports, use the following commands beginning in privileged EXEC mode:

Step	Command	Purpose
1	configure terminal	Enter global configuration mode.
2	voice-port <i>slot/port</i>	Identify the voice port you want to configure and enter voice-port configuration mode.
3	input gain <i>value</i>	Specify (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from -6 to 14.
4	output attenuation <i>value</i>	Specify (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
5	echo-cancel enable	Enable echo-cancellation of voice that is sent out the interface and received back on the same interface.
6	echo-cancel coverage <i>value</i>	Adjust the size (in milliseconds) of the echo-cancel. Acceptable values are 16, 24, and 32.
7	non-linear	Enable non-linear processing, which shuts off any signal if no near-end speech is detected. (Non-linear processing is used with echo-cancellation.)
8	timeouts initial <i>seconds</i>	Configure the initial timeout value. The initial timeout value specifies the number of seconds the system waits for the caller to input the first digit of the dialed digits. The default is 10 seconds.
9	timeouts interdigit <i>seconds</i>	Configure the interdigit timeout value. The timeouts interdigit value specifies the number of seconds the system waits (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. The default is 10 seconds.
10	timeouts wait-release { <i>value</i> infinity }	Configure the timeout value for releasing voice ports. This command limits the duration that a voice port stays in the call failure state while the Cisco MC3810 sends a busy tone, reorder tone or out-of-service tone to the port.

Step	Command	Purpose
11	timing digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF digit signal duration. The range of the DTMF digit signal duration is from 50 to 100 milliseconds. The default is 100.
12	timing inter-digit <i>milliseconds</i>	If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration. The range of the DTMF inter-digit signal duration is from 50 to 500 milliseconds. The default is 100.
13	timing pulse-digit <i>milliseconds</i>	If the voice-port dial type is pulse, configure the pulse digit signal duration. The range of the pulse digit signal duration is from 10 to 20 milliseconds. The default is 20.
14	timing pulse-inter-digit <i>milliseconds</i>	If the voice-port dial type is pulse, configure the pulse inter-digit signal duration. The range of the pulse inter-digit signal duration is from 100 to 1000 milliseconds. The default is 500.
15	timing wink-duration <i>milliseconds</i>	Configure the timing wink-duration value. This value sets the wink signal duration for a wink-start signal. This value applies only if the signal command is set to "wink-start." The range is from 100 to 400 milliseconds and the default is 200.
16	timing wink-wait <i>milliseconds</i>	Configure the timing wink-wait value. This value sets the wink wait duration for a wink-start signal. This value applies only if the signal command is set to "wink-start." The range is from 100 to 5000 milliseconds and the default is 200.
17	timing clear-wait <i>milliseconds</i>	Configure the timing clear-wait value. This value sets the amount of time between the inactive seizure signal and the call being cleared. The range is from 100 to 2000 milliseconds and the default is 400.
18	timing delay-duration <i>milliseconds</i>	Configure the timing delay-duration value. This value sets the delay signal duration for delay dial signaling. This value applies only if the signal command is set to "delay-dial." The range is from 100 to 5000 milliseconds and the default is 140.
19	timing delay-start <i>milliseconds</i>	Configure the timing delay-start value. This value sets the delay interval between the generation of the delay-start signal from incoming detection seizure. This value applies only if the signal command is set to "delay-dial." The range is from 100 to 290 milliseconds and the default is 150.
20	timing percentbreak <i>percent</i>	Configure the timing percent-break value. This value sets the percentage of the break period for a dialing pulse. The default is 50 percent.

Note After you change any voice-port command, it is a good idea to cycle the port by using the **shutdown** and **no shutdown** commands.

Activating the Voice Port

After you have configured the voice port, you need to activate the voice port to bring it online. In fact it is a good idea to cycle the port—meaning to shut the port down and then bring it online again.

To activate a voice port, use the following command in voice-port configuration mode:

Command	Purpose
no shutdown	Activate the voice port.

To cycle a voice port, use the following commands in voice-port configuration mode:

Step	Command	Purpose
1	shutdown	Deactivate the voice port.
2	voice-port <i>slot-number/subunit-number/port</i> or voice-port <i>slot/port</i>	Identify the voice port you want to activate and enter the voice-port configuration mode.
3	no shutdown	Activate the voice port.

Note If you are not going to use a voice port, shut it down.
