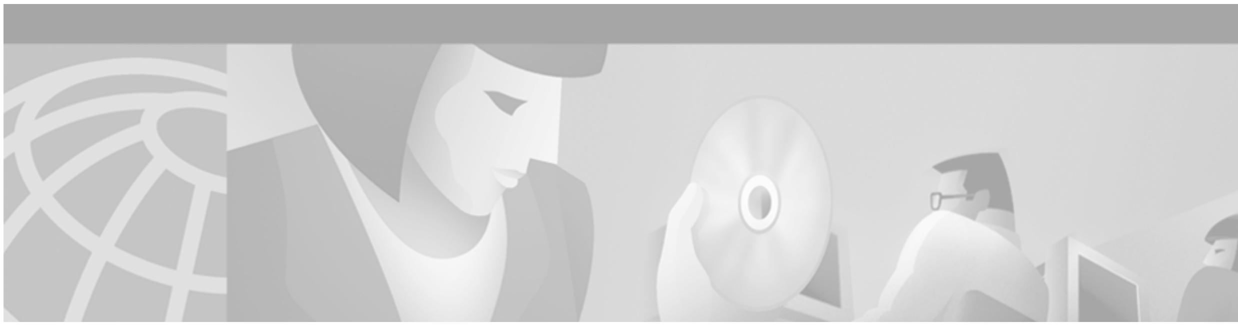


Configuring Voice Ports

Voice ports are found at the intersections of packet-based networks and traditional telephony networks, and they facilitate the passing of voice and call signals between the two networks. Physically, voice ports connect a router or access server to a line from a circuit-switched telephony device in a PBX or the public switched telephone network (PSTN).

Basic software configuration for voice ports describes the type of connection being made and the type of signaling to take place over this connection. Additional commands provide fine-tuning for voice quality, enable special features, and specify parameters to match those of proprietary PBXs.

This chapter includes the following sections:

Not all voice-port commands are covered in this chapter. Some are described in the "Configuring Trunk Connections and Conditioning Features" chapter or the "Configuring ISDN Interfaces for Voice" chapter in this configuration guide. The voice-port configuration commands included in this chapter are fully documented in the *Cisco IOS Voice, Video, and Fax Command Reference*.

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

Voice Port Configuration Overview

Voice ports on routers and access servers emulate physical telephony switch connections so that voice calls and their associated signaling can be transferred intact between a packet network and a circuit-switched network or device.

For a voice call to occur, certain information must be passed between the telephony devices at either end of the call, such as the devices' on-hook status, the line's availability, and whether an incoming call is trying to reach a device. This information is referred to as signaling, and to process it properly, the devices at both ends of the call segment (that is, those directly connected to each other) must use the same type of signaling.

The devices in the packet network must be configured to convey signaling information in a way that the circuit-switched network can understand. They must also be able to understand signaling information received from the circuit-switched network. This is accomplished by installing appropriate voice hardware in the router or access server and by configuring the voice ports that connect to telephony devices or the circuit-switched network.

The illustrations below show examples of voice port usage.

- In Figure 10, one voice port connects a telephone to the wide-area network (WAN) through the router.
- In Figure 11, one voice port connects to the PSTN and another to a telephone; the router acts like a small PBX.
- Figure 12 shows how two PBXs can be connected over a WAN to provide toll bypass.

Figure 10 Telephone to WAN

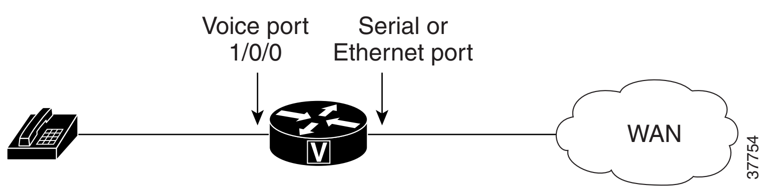


Figure 11 Telephone to PSTN

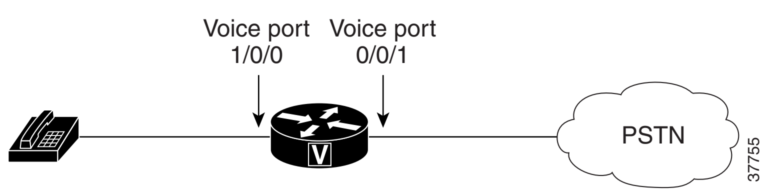


Figure 12 PBX-to-PBX over a WAN

Cisco provides a variety of Cisco IOS commands for flexibility in programming voice ports to match the physical attributes of the voice connections that are being made. Some of these connections are made using analog means of transmission, while others use digital transmission. Table 4 shows the analog and digital voice-port connection support of the router platforms discussed in this chapter.

Table 4 Analog and Digital Voice-port Support on Cisco Routers and Access Servers

Platform	Analog	Digital
Cisco 803 and 804	Yes	No
Cisco 1750	Yes	No
Cisco 2600 series	Yes	Yes
Cisco 3600 series	Yes	Yes
Cisco MC3810	Yes	Yes
Cisco AS5300	No	Yes
Cisco AS5800	No	Yes
Cisco 7200 series	No	Yes
Cisco 7500 series	No	Yes

Telephony Signaling Interfaces

Voice ports on routers and access servers physically connect the router or access server to telephony devices such as telephones, fax machines, PBXs, and PSTN central office (CO) switches. These devices may use any of several types of signaling interfaces to generate information about on-hook status, ringing, and line seizure.

The router's voice-port hardware and software need to be configured to transmit and receive the same type of signaling being used by the device with which they are interfacing so that calls can be exchanged smoothly between the packet network and the circuit-switched network.

The signaling interfaces discussed in this chapter include foreign exchange office (FXO), foreign exchange station (FXS), and receive and transmit (E&M), which are types of analog interfaces. Some digital connections emulate FXO, FXS, and E&M interfaces, and they are discussed in the second half of this chapter. It is important to know which signaling method the telephony side of the connection is using, and to match the router configuration and voice interface hardware to that signaling method.

The next three illustrations show how the different signaling interfaces are associated with different uses of voice ports. In Figure 13, FXS signaling is used for end-user telephony equipment, such as a telephone or fax machine. Figure 14 shows an FXS connection to a telephone and an FXO connection to the PSTN at the far side of a WAN; this might be a telephone at a local office going over a WAN to a router at headquarters that connects to the PSTN. In Figure 15, two PBXs are connected across a WAN by E&M interfaces. This illustrates the path over a WAN between two geographically separated offices in the same company.

Figure 13 FXS Signaling Interfaces

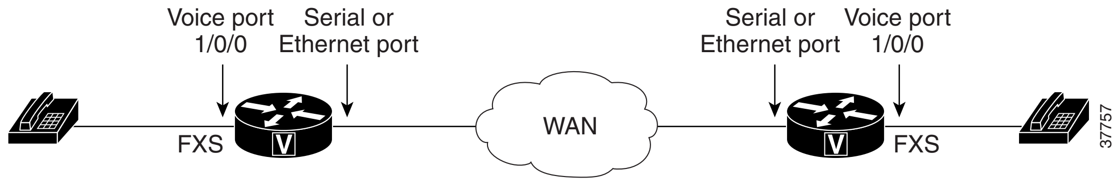


Figure 14 FXS and FXO Signaling Interfaces

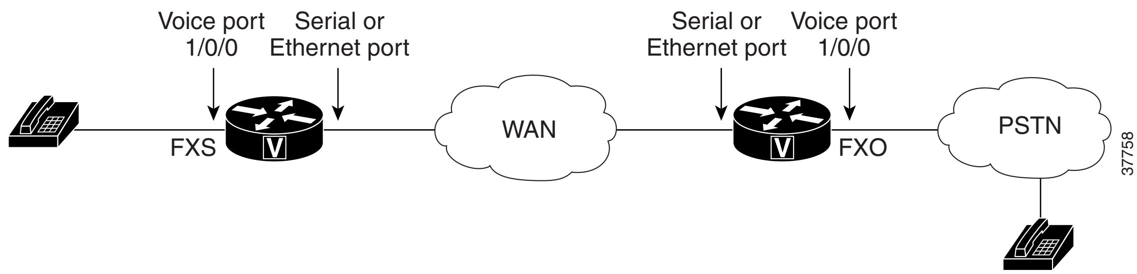
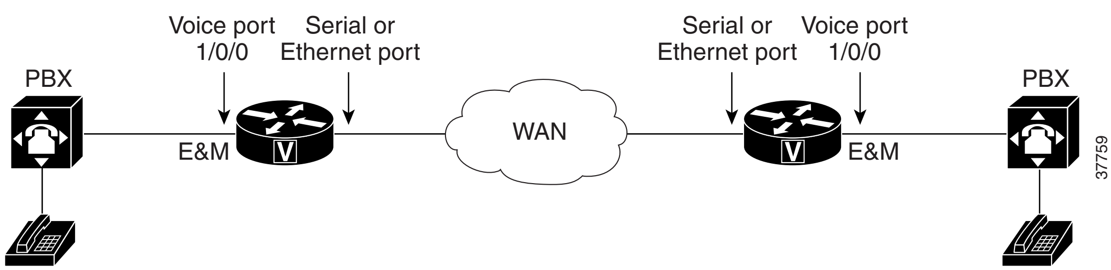


Figure 15 E&M Signaling Interfaces



FXS and FXO Interfaces

An FXS interface connects the router or access server to end-user equipment such as telephones, fax machines, or modems. The FXS interface supplies ring, voltage, and dial tone to the station and includes an RJ-11 connector for basic telephone equipment, keysets, and PBXs.

An FXO interface is used for trunk, or tie line, connections to a PSTN CO or to a PBX that does not support E&M signaling (when local telecommunications authority permits). This interface is of value for off-premise station applications. A standard RJ-11 modular telephone cable connects the FXO voice interface card to the PSTN or PBX through a telephone wall outlet.

FXO and FXS interfaces indicate on-hook or off-hook status and the seizure of telephone lines by one of two access signaling methods: loop start or ground start. The type of access signaling is determined by the type of service from the CO; standard home telephone lines use loop start, but business telephones can order ground start lines instead.

Loop-start is the more common of the access signaling techniques. When a handset is picked up (the telephone goes off-hook), this action closes the circuit that draws current from the telephone company CO and indicates a change in status, which signals the CO to provide dial tone. An incoming call is signaled from the CO to the handset by sending a signal in a standard on/off pattern, which causes the telephone to ring.

Loop-start has two disadvantages, however, that usually are not a problem on residential telephones but that become significant with the higher call volume experienced on business telephones. Loop-start signaling has no means of preventing two sides from seizing the same line simultaneously, a condition known as *glare.* Also, loop start signaling does not provide switch-side disconnect supervision for FXO calls. The telephony switch (the connection in the PSTN, another PBX, or key system) expects the router's FXO interface, which looks like a telephone to the switch, to hang up the calls it receives through its FXO port. However, this function is not built into the router for received calls; it only operates for calls originating from the FXO port.

Another access signaling method used by FXO and FXS interfaces to indicate on-hook or off-hook status to the CO is ground start signaling. It works by using ground and current detectors that allow the network to indicate off-hook or seizure of an incoming call independent of the ringing signal and allow for positive recognition of connects and disconnects. For this reason, ground start signaling is typically used on trunk lines between PBXs and in businesses where call volume on loop start lines can result in glare. See the "Disconnect Supervision Commands" section on page 82 and "FXO Supervisory Disconnect Tone Commands" section on page 84 for voice port commands that configure additional recognition of disconnect signaling.

In most cases, the default voice port command values are sufficient to configure FXO and FXS voice ports.

E&M Interfaces

Trunk circuits connect telephone switches to one another; they do not connect end-user equipment to the network. The most common form of analog trunk circuit is the E&M interface, which uses special signaling paths that are separate from the trunk's audio path to convey information about the calls. The signaling paths are known as the *E-lead* and the *M-lead*. The name *E&M* is thought to derive from the phrase *Ear* and *Mouth* or *rEceive* and *transMit* although it could also come from *Earth* and *Magnet*. The history of these names dates back to the days of telegraphy, when the CO side had a key that grounded the E circuit, and the other side had a sounder with an electromagnet attached to a battery. Descriptions such as *Ear* and *Mouth* were adopted to help field personnel determine the direction of a signal in a wire. E&M connections from routers to telephone switches or to PBXs are preferable to FXS/FXO connections because E&M provides better answer and disconnect supervision.

Like a serial port, an E&M interface has a data terminal equipment/data communications equipment (DTE/DCE) type of reference. In the telecommunications world, the *trunking* side is similar to the DCE, and is usually associated with CO functionality. The router acts as this side of the interface. The other side is referred to as the *signaling* side, like a DTE, and is usually a device such as a PBX. Five distinct physical configurations for the signaling part of the interface (Types I-V) use different methods to signal on-hook/off-hook status, as shown in Table 5. Cisco voice implementation supports E&M Types I, II, III, and V.

The physical E&M interface is an RJ-48 connector that connects to PBX trunk lines, which are classified as either two-wire or four-wire. This refers to whether the audio path is full duplex on one pair of wires (two-wire) or on two pair of wires (four-wire). A connection may be called a four-wire E&M circuit although it actually has six to eight physical wires. It is an analog connection although an analog E&M circuit may be emulated on a digital line. For more information on digital voice port configuration of E&M signaling, see the "DS0 Groups on Digital T1/E1 Voice Ports" section on page 70.

PBXs built by different manufacturers can indicate on-hook/off-hook status and telephone line seizure on the E&M interface by using any of three types of access signaling that are as follows:

- Immediate-start is the simplest method of E&M access signaling. The calling side seizes the line by going off-hook on its E-lead and sends address information as dual-tone multifrequency (DTMF) digits (or as dialed pulses on Cisco 2600 series routers and Cisco 3600 series routers) following a short, fixed-length pause.
- Wink-start is the most commonly used method for E&M access signaling, and is the default for E&M voice ports. Wink-start was developed to minimize glare, a condition found in immediate-start E&M, in which both ends attempt to seize a trunk at the same time. In wink-start, the calling side seizes the line by going off-hook on its E-lead, then waits for a short temporary off-hook pulse, or "wink," from the other end on its M-lead before sending address information. The switch interprets the pulse as an indication to proceed and then sends the dialed digits as DTMF or dialed pulses.
- In delay-dial signaling, the calling station seizes the line by going off-hook on its E-lead. After a timed interval, the calling side looks at the status of the called side. If the called side is on-hook, the calling side starts sending information as DTMF digits; otherwise, the calling side waits until the called side goes on-hook and then starts sending address information.

Table 5 E&M Wiring and Signaling Methods

E&M Type	E-Lead Configuration	M-Lead Configuration	Signal Battery Lead Configuration	Signal Ground Lead Configuration
I	Output, relay to ground	Input, referenced to ground	—	—
II	Output, relay to SG	Input, referenced to ground	Feed for M, connected to –48V	Return for E, galvanically isolated from ground
III	Output, relay to ground	Input, referenced to ground	Connected to –48V	Connected to ground
V	Output, relay to ground	Input, referenced to –48V	—	—

Analog Voice Ports Configuration Task List

Analog voice port interfaces connect routers in packet-based networks to analog two-wire or four-wire analog circuits in telephony networks. Two-wire circuits connect to analog telephone or fax devices, and four-wire circuits connect to PBXs. Typically, connections to the PSTN CO are made with digital interfaces.

This section describes how to configure analog voice ports and covers the following topics:

- Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules, page 45
- Configuring Basic Parameters on Analog FXO, FXS, or E&M Voice Ports, page 46
- Configuring Analog Telephone Connections on Cisco 803 and 804 Routers, page 50

Three other sections later in the chapter provide help with fine-tuning and troubleshooting:

- Fine-Tuning Analog and Digital Voice Ports, page 78
- Verifying Analog and Digital Voice-Port Configurations, page 96
- Troubleshooting Analog and Digital Voice Port Configurations, page 107

Prerequisites for Configuring Analog Voice Ports

- Obtain two- or four-wire line service from your service provider or from a PBX.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.
- Install at least one other network module or WAN interface card to provide the connection to the network LAN or WAN.
- Establish a working IP and Frame Relay or ATM network. For more information about configuring IP, refer to the *Cisco IOS IP Configuration Guide*, Release 12.2.
- Install appropriate voice processing and voice interface hardware on the router. See the "Configuring Platform-Specific Analog Voice Hardware" section on page 43.

Preparing to Configure Analog Voice Ports

Before configuring an analog voice port, assemble the following information about the telephony connection that the voice port will be making. If connecting to a PBX, it is important to understand the PBX's wiring scheme and timing parameters. This information should be available from your PBX vendor or the reference manuals that accompany your PBX.

- Telephony signaling interface: FXO, FXS, or E&M
- Locale code (usually the country) for call progress tones
- If FXO, type of dialing: DTMF (touch-tone) or pulse
- If FXO, type of start signal: loop-start or ground-start
- If E&M, type: I, II, III, or V
- If E&M, type of line: two-wire or four-wire
- If E&M, type of start signal: wink, immediate, delay-dial

Table 6 should help you determine which hardware and configuration instructions are appropriate for your situation. Table 7 on page 42 shows slot and port numbering, which differs for each of the voice-enabled routers. More current information may be available in the release notes that accompany the Cisco IOS software you are using.

Table 6 Analog Voice Port Configurations

Telephony Signaling Interface	Router Platform	Voice Hardware Required	Section Containing Voice Port Configuration Instructions
End user: telephone or fax	Cisco 803 Cisco 804	—	"Configuring Analog Telephone Connections on Cisco 803 and 804 Routers"
FXO	Cisco 1750 Cisco 2600 series Cisco 3600 series	VIC-2FXO, VIC-2FXO-EU	"Configuring Basic Parameters on Analog FXO, FXS, or E&M Voice Ports"
	Cisco MC3810	MC3810-AVM6 MC3810-APM-FXO	
FXS	Cisco 1750 Cisco 2600 series Cisco 3600 series	VIC-2FXS	
	Cisco MC3810	MC3810-AVM6 MC3810-APM-FXS	
E&M	Cisco 1750 Cisco 2600 series Cisco 3600 series	VIC-2E/M	
	Cisco MC3810	MC3810-AVM6 MC3810-APM-EM	

Table 7 Analog Voice Slot/Port Designations

Router Platform	Voice Hardware	Chassis Slot Numbers	Voice NM Slot Numbers	Voice Port Numbers
Cisco 803, 804	Analog POTS	—	—	—
Cisco 1750	Analog VIC	0 to 1	—	0 to 1
Cisco 2600 series	Voice/fax network module with two-port VIC	Varies, based on router	1	0 to 1
Cisco 3600 series	Voice/fax network module with two-port voice over interface cards (VICs)	1	3620: 0 to 1 3640: 0 to 3 3660: 1 to 6	0 to 1
Cisco MC3810	Analog voice module (AVM)	1	—	1 to 6

Configuring Platform-Specific Analog Voice Hardware

This section describes the general types of analog voice port hardware available for the router platforms included in this chapter:

- Cisco 800 Series Routers, page 43
- Cisco 1750 Modular Router, page 43
- Cisco 2600 Series and Cisco 3600 Series Routers, page 44
- Cisco MC3810 Multiservice Concentrator, page 44

Note For current information about supported hardware, see the release notes for the platform and Cisco IOS release being used.

Cisco 800 Series Routers

Cisco 803 and Cisco 804 routers support data and voice applications. The data applications on these routers are implemented through the ISDN port, and the voice applications are implemented with ISDN Basic Rate Interface (BRI) through the telephone ports. If a Cisco 803 or 804 router is being used, connect two devices, such as an analog touch-tone telephone, fax machine, or modem through two fixed telephone ports, the gray PHONE 1 and PHONE 2 ports that have RJ-11 connectors. Each device is connected to basic telephone services through the ISDN line.

For more information, refer to the *Cisco 800 Series Routers Hardware Installation Guide*.

Cisco 1750 Modular Router

The Cisco 1750 modular router provides Voice over IP (VoIP) functionality and can carry voice traffic (for example, telephone calls and faxes) over an IP network. To make a voice connection, the router must have a supported VIC installed. The Cisco 1750 router supports two slots for either WAN interface cards (WICs) or VICs and supports one VIC-only slot. For analog connections, two-port VICs are available to support FXO, FXS, and E&M signaling. VICs provide direct connections to telephone equipment (analog phones, analog fax machines, key systems, or PBXs) or to a PSTN.

For more information, refer to the *Cisco 1750 Voice-over-IP Quick Start Guide*.

Cisco 2600 Series and Cisco 3600 Series Routers

The Cisco 2600 and 3600 series routers are modular, multifunction platforms that combine dial access, routing, local area network-to-local area network (LAN) services, and multiservice integration of voice, video, and data in the same device.

Voice network modules installed in Cisco 2600 series or Cisco 3600 series routers convert telephone voice signals into data packets that can be transmitted over an IP network. The voice network modules have no connectors; VICs installed in the network modules provide connections to the telephone equipment or network. VICs work with existing telephone and fax equipment and are compatible with H.323 standards for audio and video conferencing.

The Cisco 2600 series router can house one network module. In the Cisco 3600 series, the Cisco 3620 router has slots for up to two network modules; the Cisco 3640 router has slots for up to four network modules; and the Cisco 3660 router has slots for up to six network modules. (Typically, one of the slots is used for LAN connectivity.)

For analog telephone connections, low-density voice/fax network modules that contain either one or two VIC slots are installed in the network module slots. Each VIC is specific to a particular telephone signaling interface (FXS, FXO, or E&M); therefore, the VIC determines the type of signaling on that module.

For more information, refer to the following:

- *Cisco 2600 Series Hardware Installation Guide*
- *Cisco 3600 Series Hardware Installation Guide*
- *Cisco Network Module Hardware Installation Guide*

Cisco MC3810 Multiservice Concentrator

To support analog voice circuits, a Cisco MC3810 multiservice concentrator must be equipped with an AVM, which supports six analog voice ports. By installing specific signaling modules known as analog personality modules (APMs), the analog voice ports may be equipped for the following signaling types in various combinations: FXS, FXO, and E&M. For FXS, the analog voice ports use an RJ-11 connector interface to connect to analog telephones or fax machines (two-wire) or to a key system (four-wire). For FXO, the analog voice ports use an RJ-11 physical interface to connect to a CO trunk. For E&M connections, the analog voice ports use an RJ-1CX physical interface to connect to an analog PBX (two-wire or four-wire).

Optional high-performance voice compression modules (HCMs) can replace standard voice compression modules (VCMs) to operate according to the voice compression coding algorithm (codec) specified when the Cisco MC3810 concentrator is configured. The HCM2 provides four voice channels at high codec complexity and eight channels at medium complexity. The HCM6 provides 12 voice channels at high complexity and 24 channels at medium complexity. One or two HCMs can be installed in a Cisco MC3810 multiservice concentrator, but an HCM may not be combined with a VCM in one chassis.

For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide.*

Note For current information about supported hardware, see the release notes for the platform and Cisco IOS release being used.

Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules

The term *codec* stands for *coder-decoder*. A codec is a particular method of transforming analog voice into a digital bit stream (and vice versa) and also refers to the type of compression used. Several different codecs have been developed to perform these functions, and each one is known by the number of the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) standard in which it is defined. For example, two common codecs are the G.711 and the G.729 codecs. The various codecs use different algorithms to encode analog voice into digital bit-streams and have different bit rates, frame sizes, and coding delays associated with them. The codecs also differ in the amount of perceived voice quality they achieve. Specialized hardware and software in the digital signal processors (DSPs) perform codec transformation and compression functions, and different DSPs may offer different selections of codecs.

Select the same type of codec as the one that is used at the other end of the call. For instance, if a call was coded with a G.729 codec, it must be decoded with a G.729 codec. Codec choice is configured on dial peers. For more information, see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide.

Codec complexity refers to the amount of processing power that a codec compression technique requires: some require more processing power than others. Codec complexity affects call density, which is the number of calls that can take place on the DSP interfaces, which can be HCMs, port adapter DSP farms, or voice cards, depending on the type of router (in this case, the Cisco MC3810 multiservice concentrator). The greater the codec complexity, the fewer the calls that can be handled.

Codec complexity is either medium or high. The difference between medium- and high-complexity codecs is the amount of CPU power necessary to process the algorithm and, therefore, the number of voice channels that can be supported by a single DSP. All medium-complexity codecs can also be run in high-complexity mode, but fewer (usually half as many) channels will be available per DSP.

For details on the number of calls that can be handled simultaneously using each of the codec standards, refer to the entries for the **codec** and **codec complexity** commands in the *Cisco IOS Voice, Video, and Fax Command Reference*.

On a Cisco MC3810 concentrator, only a single codec complexity setting is used, even when two HCMs are installed. The value that is specified in this task affects the choice of codecs available when the **codec** dial-peer configuration command is configured. See the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide.

Note On the Cisco MC3810 with high-performance compression modules, check the DSP voice channel activity with the **show voice dsp** command. If any DSP voice channels are in the busy state, the codec complexity cannot be changed. When all the DSP channels are in the idle state, changes can be made to the codec complexity selection.

To configure codec complexity on the Cisco MC3810 multiservice concentrator using HCMs, use the following commands beginning in privileged EXEC mode:

	Command	Purpose
Step 1	`Router# show voice dsp`	Checks the DSP voice channel activity. If any DSP voice channels are in the busy state, the codec complexity cannot be changed. When all the DSP channels are in the idle state, continue to Step 2.
Step 2	`Router# configure terminal`	Enters global configuration mode.
Step 3	`Router(config)# voice-card 0`	Enters voice-card configuration mode and specifies voice card 0.
Step 4	`Router(config-voicecard)# codec complexity {high | medium}`	(For analog voice ports) Specifies codec complexity based on the codec standard being used. This setting restricts the codecs available in dial peer configuration. All voice cards in a router must use the same codec complexity setting. The keywords are as follows: • **high**—Specifies two voice channels encoded in any of the following formats: G.711ulaw, G.711alaw, G.723.1(r5.3), G.723.1 Annex A(r5.3), G.723.1(r6.3), G.723.1 Annex A(r6.3), G.726(r16), G.726(r24), G.726(r32), G.728, G.729, G.729 Annex B, and fax relay. • **medium**—(default) Specifies four voice channels encoded in any of the following formats: G.711ulaw, G.711alaw, G.726(r16), G.726(r24), G.726(r32), G.729 Annex A, G.729 Annex B with Annex A, and fax relay. **Note** If two HCMs are installed, this command configures both HCMs at once.

Configuring Basic Parameters on Analog FXO, FXS, or E&M Voice Ports

This section describes commands for basic analog voice port configuration. All the data recommended in the "Preparing to Configure Analog Voice Ports" section on page 41 should be gathered before starting this procedure.

If configuring a Cisco MC3810 multiservice concentrator that has HCMs, codec complexity should also be configured, following the steps in the "Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules" section on page 45.

Note If you have a Cisco MC3810 multiservice concentrator or Cisco 3660 router, the **compand-type a-law** command must be configured on the analog ports only. The Cisco 2660, 3620, and 3640 routers do not require the configuration of th **compand-type a-law** command, however, if you request a list of commands, the **compand-type a-law** command will display.

In addition to the basic voice port parameters described in this section, there are commands that allow voice port configurations to be fine tuned. In most cases, the default values for fine-tuning commands are sufficient for establishing FXO and FXS voice port configurations. E&M voice ports are more likely to require some configuration. If it is necessary to change some of the voice port values to improve voice quality or to match parameters on proprietary PBXs to which you are connecting, use the commands in the current section and also in the "Fine-Tuning Analog and Digital Voice Ports" section on page 78.

After the voice-port has been configured, make sure that the ports are operational by following the steps described in the following sections:

- Verifying Analog and Digital Voice-Port Configurations, page 96
- Troubleshooting Analog and Digital Voice Port Configurations, page 107

For more information on these and other voice port commands, see the *Cisco IOS Voice, Video, and Fax Command Reference*.

Note The commands, keywords, and arguments that you are able to use may differ slightly from those presented here, based on your platform, Cisco IOS release, and configuration. When in doubt, use Cisco IOS command help (**command ?**) to determine the syntax choices that are available.

To configure basic analog voice port parameters on Cisco 1750, Cisco 2600 series, Cisco 3600 series, and Cisco MC3810 routers, use the following commands beginning in global configuration mode:

	Command	Purpose

Step 1	**Cisco 1750 and MC3810**
	`Router(config)# voice-port slot/port`
	
	Cisco 2600 and 3600 series
	`Router(config)# voice-port slot/subunit/port`
		Enters voice-port configuration mode.
		The arguments are as follows:
		• *slot*—Specifies the number of the router slot where the voice network module is installed (Cisco 2600 and Cisco 3600 series routers) or the router slot number where the analog voice module is installed (Cisco MC3810 multiservice concentrator).
		• *port*—Indicates the voice port. Valid entries are 0 or 1.
		• *subunit*—Specifies the location of the VIC.
		Note The slash must be entered between *slot* and *port.*
		Valid entries vary by router platform; see Table 7 on page 42 or enter the **show voice port summary** command for available values.

Step 2	**FXO or FXS** `Router(config-voiceport)# signal {loop-start | ground-start}`	Selects the access signaling type to match that of the telephony connection you are making. The keywords are as follows: • **loop-start**—(default) Uses a closed circuit to indicate off-hook status; used for residential loops. • **ground-start**—Uses ground and current detectors; preferred for PBXs and trunks.

	Command	Purpose
	E&M `Router(config-voiceport)# signal {wink-start | immediate-start | delay-dial}`	The keywords are as follows: • **wink-start**—(default) Indicates that the calling side seizes the line, then waits for a short off-hook *wink* from the called side before proceeding. • **immediate-start**—Indicates that the calling side seizes the line and immediately proceeds; used for E&M tie trunk interfaces. • **delay-dial**—Indicates that the calling side seizes the line and waits, then checks to determine whether the called side is on-hook before proceeding; if not, it waits until the called side is on-hook before sending digits. Used for E&M tie trunk interfaces. **Note** Configuring the **signal** keyword for one voice port on a Cisco 2600 or 3600 series router VIC changes the signal value for both ports on the VIC.
Step 3	`Router(config-voiceport)# cptone locale`	Selects the two-letter locale for the voice call progress tones and other locale-specific parameters to be used on this voice port. Cisco routers comply with the ISO 3166 locale name standards. To see valid choices, enter a question mark (?) following the **cptone** command. The default is **us**.
Step 4	`Router(config-voiceport)# dial-type {dtmf | pulse}`	(FXO only) Specifies the dialing method for outgoing calls.
Step 5	`Router(config-voiceport)# operation {2-wire | 4-wire}`	(E&M only) Specifies the number of wires used for voice transmission at this interface (the audio path only, not the signaling path). The default is 2-wire.
Step 6	`Router(config-voiceport)# type {1 | 2 | 3 | 5}`	(E&M only) Specifies the type of E&M interface to which this voice port is connecting. See Table 5 on page 40 for an explanation of E&M types. The default is 1.

Step 7	**Cisco 1750 Router and 2600 and 3600 Series Routers**
	`Router(config-voiceport)# ring frequency {25 | 50}`
	Cisco MC3810 Multiservice Concentrator
	`Router(config-voiceport)# ring frequency {20 | 30}`
		(FXS only) Selects the ring frequency, in hertz, used on the FXS interface. This number must match the connected telephony equipment and may be country-dependent. If not set properly, the attached telephony device may not ring or it may buzz.
		The keyword default is 25 on the Cisco 1750 router, 2600 and 3600 series routers; and 20 on the Cisco MC3810 multiservice concentrator.

	Command	Purpose
Step 8	`Router(config-voiceport)#` **`ring number`** *`number`*	(FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router. The default is 1.
Step 9	`Router(config-voiceport)#` **`ring cadence`** `{[`**`pattern01`** `|` **`pattern02`** `|` **`pattern03`** `|` **`pattern04`** `|` **`pattern05`** `|` **`pattern06`** `|` **`pattern07`** `|` **`pattern08`** `|` **`pattern09`** `|` **`pattern10`** `|` **`pattern11`** `|` **`pattern12`**`] | [`**`define`** *`pulse interval`*`]}`	(FXS only) Specifies an existing pattern for ring, or it defines a new one. Each pattern specifies a ring-pulse time and a ring-interval time. The keywords and arguments are as follows: • **pattern01** through **pattern12** name pre-set ring cadence patterns. Enter **ring cadence ?** to see ring pattern explanations. • **define** *pulse interval* specifies a user-defined pattern: *pulse* is a number (one or two digits, from 1 to 50) specifying ring pulse (on) time in hundreds of milliseconds, and *interval* is a number (one or two digits from 1 to 50) specifying ring interval (off) time in hundreds of milliseconds. The default is the pattern specified by the cptone locale that has been configured.
Step 10	`Router(config-voiceport)#` **`description`** *`string`*	Attaches a text string to the configuration that describes the connection for this voice port. This description appears in various displays and is useful for tracking the purpose or use of the voice port. The *string* argument is a character string from 1 to 255 characters in length. The default is that there is no text string (describing the voice port) attached to the configuration.
Step 11	`Router(config-voiceport)#` **`no shutdown`**	Activates the voice port. If a voice port is not being used, shut the voice port down with the **shutdown** command.

Configuring Analog Telephone Connections on Cisco 803 and 804 Routers

Multiple devices (analog telephone, fax machine, or modem) can be connected to a Cisco 803 or 804 telephone port. The number of devices that can be connected depends on the ringer equivalent number (REN) of each device that is to be connected. (The REN can usually be found on the bottom of a device.) The REN of the router telephone port is 5, so if the REN of each device to be connected is 1, a maximum of five devices can be connected to that particular telephone port.

These routers support touch-tone analog telephones only; they do not support rotary telephones.

To configure standard features for analog telephone connections on Cisco 803 and 804 routers, use the following commands in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# pots country country`	Specifies the country to use for country-specific default settings for physical characteristics. Enter **pots country ?** for a list of supported countries and the codes to enter. A default country is not defined.
Step 2	`Router(config)# pots line-type {type1 | type2 | type3}`	(Optional) Specifies the impedance of telephones, fax machines, or modems connected to a Cisco 800 series router. The keywords are as follows: • **type1**—Specifies the resistance used for the POTS connection, typically 600 ohms. • **type2**—Specifies the resistance used for the POTS connection, typically 900 ohms. • **type3**—Specifies the resistance used for the POTS connection, typically 300/400 ohms. The default depends on the country chosen in the **pots country** command.
Step 3	`Router(config)# pots dialing-method {overlap | enblock}`	(Optional) Specifies how the router collects and sends digits dialed on connected telephones, fax machines, or modems. The keywords are as follows: • **overlap—**Tells the router to send each digit dialed in a separate message. • **enblock—**Tells the router to collect all digits dialed and to send the digits in one message. The default depends on the country chosen in the **pots country** command.

	Command	Purpose
Step 4	`Router(config)#` **`pots disconnect-supervision`** `{`**`osi`** `|` **`reversal`**`}`	(Optional) Specifies how the router notifies the connected telephones, fax machines, or modems when the calling party has disconnect. The keywords are as follows: • **osi**—(open switching interval) Specifies the duration for which DC voltage applied between tip and ring conductors of a telephone port is removed. • **reversal**—Specifies the polarity reversal of the tip and ring conductors of a telephone port. The default depends on the country chosen in the **pots country** command.
Step 5	`Router(config)#` **`pots encoding`** `{`**`alaw`** `|` **`ulaw`**`}`	(Optional) Specifies the pulse code modulation (PCM) encoding scheme for telephones, fax machines, or modems connected to a Cisco 800 series router. The keywords are as follows: • **alaw**—Specifies the ITU-T PCM encoding scheme used to represent analog voice samples as digital values. • **ulaw**—Specifies the North American PCM encoding scheme used to represent analog voice samples as digital values. The default depends on the country chosen in the **pots country** command.
Step 6	`Router(config)#` **`pots tone-source`** `{`**`local`** `|` **`remote`**`}`	(Optional) Specifies the source of dial, ringback, and busy tones for telephones, fax machines, or modems connected to a Cisco 800 series router. The keywords are as follows: • **local**—(default) Specifies that the router supplies the tones. • **remote**—Specifies that the telephone switch supplies the tones.
Step 7	`Router(config)#` **`pots ringing-freq`** `{`**`20Hz`** `|` **`25Hz`** `|` **`50Hz`**`}`	(Optional) Specifies the frequency at which telephones, fax machines, or modems connected to a Cisco 800 series router ring. The keywords are as follows: • **20Hz**—Indicates that connected devices ring at 20 Hz. • **25Hz**—Indicates that connected devices ring at 25 Hz. • **50Hz**—Indicates that connected devices ring at 50 Hz. The default depends on the country chosen in the **pots country** command.

	Command	Purpose
Step 8	`Router(config)# pots disconnect-time interval`	(Optional) Specifies the interval at which the disconnect method is applied if connected telephones, fax machines, or modems fail to detect that a calling party has disconnected. The *interval* argument is the number of milliseconds of the interval and ranges from 50 to 2000. The default depends on the country chosen in the **pots country** command.
Step 9	`Router(config)# pots silence-time seconds`	(Optional) Specifies the interval of silence after a calling party disconnects. The *seconds* argument is the number of seconds of the interval and ranges from 0 to 10. The default depends on the country chosen in the **pots country** command.
Step 10	`Router(config)# pots distinctive-ring-guard-time milliseconds`	(Optional) Specifies the delay after which a telephone port can be rung after a previous call is disconnected. The *milliseconds* argument is the number of milliseconds of the delay and ranges from 0 to 1000. The default depends on the country chosen in the **pots country** command.

Verifying Analog Telephone Connections on Cisco 803 and 804 Routers

After configuring analog telephone connections, perform the following steps to verify proper operation:

Step 1 Pick up the handset of an attached telephony device and check for a dial tone.

Step 2 Review the configuration using the **show pots status** command, which displays settings of physical characteristics and other information on telephone interfaces.

```
Router# show pots status

POTS Global Configuration:
 Country: United States
  Dialing Method: Overlap, Tone Source: Remote, CallerId Support: YES
  Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
  Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 msec
  Disconnect timer: 1000 msec, Disconnect Silence timer: 5 sec
  TX Gain: 6dB, RX Loss: -6dB,
  Filter Mask: 6F
  Adaptive Cntrl Mask: 0
POTS PORT: 1
  Hook Switch Finite State Machine:
   State: On Hook, Event: 0
   Hook Switch Register: 10, Suspend Poll: 0
  CODEC Finite State Machine
   State: Idle, Event: 0
  Connection: None, Call Type: Two Party, Direction: Rx only
  Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
  Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 msec
  Disconnect timer: 1000 msec, Disconnect Silence timer: 5 sec
  TX Gain: 6dB, RX Loss: -6dB,
```

```
  Filter Mask: 6F
  Adaptive Cntrl Mask: 0
  CODEC Registers:
   SPI Addr: 2, DSLAC Revision: 4
   SLIC Cmd: 0D, TX TS: 00, RX TS: 00
   Op Fn: 6F, Op Fn2: 00, Op Cond: 00
   AISN: 6D, ELT: B5, EPG: 32 52 00 00
   SLIC Pin Direction: 1F
  CODEC Coefficients:
   GX: A0 00
   GR: 3A A1
   Z: EA 23 2A 35 A5 9F C2 AD 3A AE 22 46 C2 F0
   B: 29 FA 8F 2A CB A9 23 92 2B 49 F5 37 1D 01
   X: AB 40 3B 9F A8 7E 22 97 36 A6 2A AE
   R: 01 11 01 90 01 90 01 90 01 90 01 90
   GZ: 60
  ADAPT B: 91 B2 8F 62 31
 CSM Finite State Machine:
   Call 0 - State: idle, Call Id: 0x0
    Active: no
   Call 1 - State: idle, Call Id: 0x0
    Active: no
   Call 2 - State: idle, Call Id: 0x0
    Active: no
 POTS PORT: 2
   Hook Switch Finite State Machine:
    State: On Hook, Event: 0
    Hook Switch Register: 20, Suspend Poll: 0
   CODEC Finite State Machine:
    State: Idle, Event: 0
    Connection: None, Call Type: Two Party, Direction: Rx only
    Line Type: 600 ohm, PCM Encoding: u-law, Disc Type: OSI,
    Ringing Frequency: 20Hz, Distinctive Ring Guard timer: 0 mse
  Disconnect timer: 1000msec,Disconnect Silence timer: 5 sec
   TX Gain: 6dB, RX Loss: -6dB,
   Filter Mask: 6F
   Adaptive Cntrl Mask: 0
 CODEC Registers:
   SPI Addr: 3, DSLAC Revision: 4
   SLIC Cmd: 0D, TX TS: 00, RX TS: 00
   Op Fn: 6F, Op Fn2: 00, Op Cond: 00
   AISN: 6D, ELT: B5, EPG: 32 52 00 00
   SLIC Pin Direction: 1F
 CODEC Coefficients:
   GX: A0 00
   GR: 3A A1
   Z: EA 23 2A 35 A5 9F C2 AD 3A AE 22 46 C2 F0
   B: 29 FA 8F 2A CB A9 23 92 2B 49 F5 37 1D 01
   X: AB 40 3B 9F A8 7E 22 97 36 A6 2A AE
   R: 01 11 01 90 01 90 01 90 01 90 01 90
   GZ: 60
  ADAPT B: 91 B2 8F 62 31
 CSM Finite State Machine:
   Call 0 - State: idle, Call Id: 0x0
    Active: no
   Call 1 - State: idle, Call Id: 0x0
    Active: no
   Call 2 - State: idle, Call Id: 0x0
    Active: no
Time Slot Control: 0
```

Troubleshooting Tip for Cisco 803 and 804 Routers

Check to ensure that all cables are securely connected.

Configuring Digital Voice Ports

The digital voice port commands discussed in this section configure channelized T1 or E1 connections; for information on ISDN connections, see "Configuring ISDN Interfaces for Voice" in this configuration guide.

The T1 or E1 lines that connect a telephony network to the digital voice ports on a router or access server contain channels for voice calls; a T1 line contains 24 full-duplex channels or *timeslots*, and an E1 line contains 30. The signal on each channel is transmitted at 64 kbps, a standard known as digital signal 0 (DS0); the channels are known as DS0 channels. The **ds0-group** command creates a logical voice port (a DS0 group) from some or all of the DS0 channels, which allows you to address those channels easily, as a group, in voice-port configuration commands.

Digital voice ports are found at the intersection of a packet voice network and a digital, circuit-switched telephone network. The digital voice port interfaces that connect the router or access server to T1 or E1 lines pass voice data and signaling between the packet network and the circuit-switched network.

Signaling is the exchange of information about calls and connections between two ends of a communication path. For instance, signaling communicates to the call's end points whether a line is idle or busy, whether a device is on-hook or off-hook, and whether a connection is being attempted. An end point can be a CO switch, a PBX, a telephony device such as a telephone or fax machine, or a voice-equipped router acting as a gateway. There are two aspects to consider about signaling on digital lines: one aspect is the actual information about line and device states that is transmitted, and the second aspect is the method used to transmit the information on the digital lines.

The actual information about line and device states is communicated over digital lines using signaling methods that emulate the methods used in analog circuit-switched networks: FXS, FXO, and E&M.

The method used to transmit the information describes the way that the emulated analog signaling is transmitted over digital lines, which may be *common-channel signaling* (CCS) or *channel-associated signaling* (CAS). CCS sends signaling information down a dedicated channel and CAS takes place within the voice channel itself. This chapter describes CAS signaling, which is sometimes called *robbed-bit signaling* because user bandwidth is *robbed* by the network for signaling. A bit is taken from every sixth frame of voice data to communicate on- or off-hook status, wink, ground start, dialed digits, and other information about the call.

In addition to setting up and tearing down calls, CAS provides the receipt and capture of dialed number identification (DNIS) and automatic number identification (ANI) information, which are used to support authentication and other functions. The main disadvantage of CAS signaling is its use of user bandwidth to perform these signaling functions.

For signaling to pass between the packet network and the circuit-switched network, both networks must use the same type of signaling. The voice ports on Cisco routers and access servers can be configured to match the signaling of most COs and PBXs, as explained in this chapter.

This section discusses the following topics:

Prerequisites for Configuring Digital Voice Ports

Digital T1 or E1 packet voice capability requires specific service, software, and hardware:

- Obtain T1 or E1 service from the service provider or from your PBX.
- Create your company's dial plan.
- Establish a working telephony network based on your company's dial plan.
- Establish a connection to the network LAN or WAN.
- Set up a working IP and Frame Relay or ATM network. For more information about configuring IP, refer to the *Cisco IOS IP Configuration Guide*, Release 12.2.
- Install appropriate voice processing and voice interface hardware on the router. See the "Platform-Specific Digital Voice Hardware" section on page 58.
- (Cisco 2600 and 3600 series routers) For digital T1 packet voice trunk network modules, install Cisco IOS Release 12.0(5)XK, 12.0(7)T, 12.2(1), or a later release. The minimum DRAM memory requirements are as follows:
  - 32 MB, with one or two T1 lines
  - 48 MB, with three or four T1 lines
  - 64 MB, with five to ten T1 lines
  - 128 MB, with more than ten T1 lines

  The memory required for high-volume applications may be greater than that listed. Support for digital T1 packet voice trunk network modules is included in Plus feature sets. The IP Plus feature set requires 8 MB of Flash memory; other Plus feature sets require 16 MB.
- (Cisco 2600 and 3600 series routers) For digital E1 packet voice trunk network modules, install Cisco IOS Release 12.1(2)T, 12.2(1), or a later release. The minimum DRAM memory requirements are:
  - 48 MB, with one or two E1s
  - 64 MB, with three to eight E1s
  - 128 MB, with 9 to 12 E1s

  For high-volume applications, the memory required may be greater than these minimum values. Support for digital E1 packet voice trunk network modules is included in Plus feature sets. The IP Plus feature set requires 16 MB of Flash memory.
- (Cisco MC3810 concentrators) HCMs require Cisco IOS Release 12.0(7)XK or 12.1(2)T, 12.2(1), or a later release.
- (Cisco 7200 and 7500 series routers) For digital T1/E1 voice port adapters, install Cisco IOS Release 12.0(5)XE, 12.0(7)T, 12.2(1), or a later release. The minimum DRAM memory requirement to support T1/E1 high-capacity digital voice port adapters is 64 MB.

The memory required for high-volume applications may be greater than that listed. Support for T1/E1 high-capacity digital voice port adapters is included in Plus feature sets. The IP Plus feature set requires 16 MB of Flash memory.

Preparing Information to Configure Digital Voice Ports

Gather the following information about the telephony network connection that the voice port will be making:

- Line interface: T1 or E1
- Signaling interface: FXO, FXS, or E&M. If the interfaces are Primary Rate Interface (PRI) or BRI, see the "Configuring ISDN Interfaces for Voice" chapter in this configuration guide and *Cisco IOS Terminal Services Configuration Guide*.
- Line coding: AMI or B8ZS for T1, and AMI or HDB3 for E1
- Framing format: SF (D4) or ESF for T1, and CRC4 or no-CRC4 for E1
- Number of channels

Table 8 describes voice-port hardware configurations for various platforms. After the controllers have been configured, the **show voice port summary** command can also be used to determine available voice port numbers. If the **show voice port** command and a specific port number is entered, the default voice-port configuration for that port displays.

Table 8 Digital Voice Slot/Port Designations

Router Platform	Voice Hardware	Slot Number	Port Number
Cisco 2600 series	Digital T1/E1 Packet Voice Trunk Network Module (NM-HDV with VWIC-1MFT or VWIC-2MFT) One network module can be installed in a Cisco 2600 series router.	*slot* is the router location of the voice module. 1	*port* is the VWIC location in the network module. 0 to 1
Cisco 3600 series	Digital T1/E1 Packet Voice Trunk Network Module (NM-HDV with VWIC-1MFT or VWIC-2MFT) One network module can be installed in a Cisco 3620 router. A Cisco 3640 router can support three modules, and as many as six can be installed in a Cisco 3660 router.	*slot* is the router location of the voice module. 3620: 0 to 1 3640: 0 to 3 3660: 0 to 5	*port* is the VWIC location in the network module. 0 to 1

Table 8 Digital Voice Slot/Port Designations (continued)

Router Platform	Voice Hardware	Slot Number	Port Number
Cisco MC3810	• Digital voice module (DVM) • Voice compression module (VCM3 or VCM6) or • High-compression module (HCM2 or HCM6) VCM3 and VCM6 do not support codec complexity options.	1	—
Cisco AS5300	One Octal T1/E1 feature card (eight ports) or one Quad T1/E1 feature card (four ports) and one or two VFCs for voice and fax features.	—	*controller* is : Octal: 0 to 7 Quad: 0 to 3
Cisco AS5800	Up to four 12-port T1/E1 trunk cards and up to eight VFCs	*shelf* is 1 *slot* is 0 to 5	0 to 11
Cisco 7200 series	• Two-port T1/E1 enhanced digital voice port adapters • PA-VXC (high-capacity) • PA-VXB (moderate capacity) Port adapter slot 0 is reserved for the Fast Ethernet port on the I/O controller (if present).	Port adapter slot: from 1 to 4, or from 1 to 6	Interface port: 0 to 1
Cisco 7500 series	PA-VXB and PA-VXC on a VIP2 or VIP4 in Cisco 7500 series routers If the VIP is inserted in interface processor slot 3 and port adapter slot 0, then the addresses of the PA-VXB or PA-VXC are 3/0/0 or 3/0/1 (interface processor slot 3, port adapter slot 0, and interfaces 0 and 1).	Interface processor slot: 0 to 12 (depends on the number of slots in the router)	Port adapter slot: always 0 or 1 Interface port: 0 or 1

The following is **show voice port summary** sample output for a Cisco MC3810 multiservice concentrator:

```
Router# show voice port summary

IN       OUT
PORT   CH SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
0:17   18 fxo-ls     down  down idle     on-hook  y
0:18   19 fxo-ls     up    dorm idle     on-hook  y
0:19   20 fxo-ls     up    dorm idle     on-hook  y
0:20   21 fxo-ls     up    dorm idle     on-hook  y
0:21   22 fxo-ls     up    dorm idle     on-hook  y
0:22   23 fxo-ls     up    dorm idle     on-hook  y
0:23   24 e&m-imd    up    dorm idle     idle     y
```

Platform-Specific Digital Voice Hardware

This section briefly describes digital voice hardware on the following platforms:

- Cisco 2600 series and Cisco 3600 series routers
- Cisco MC3810 multiservice concentrator
- Cisco AS5300 universal access server
- Cisco AS5800 universal access server
- Cisco 7200 series and Cisco 7500 series routers

Note For current information about supported hardware, see the release notes for the platform and Cisco IOS release you are using.

Cisco 2600 Series and Cisco 3600 Series Routers

Digital voice hardware on Cisco 2600 series and Cisco 3600 series modular access routers includes the high-density voice (HDV) network module and the multiflex trunk (MFT) voice/WAN interface card (VWIC). When an HDV is used in conjunction with an MFT and packet voice DSP modules (PVDMs), the HDV module is also called a *digital packet voice trunk network module*. The digital T1 or E1 packet voice trunk network module supports T1 or E1 applications, including fractional use. The T1 version integrates a fully managed data service unit/channel service unit (DSU/CSU), and the E1 version includes a fully managed DSU. The digital T1 or E1 packet voice trunk network module provides per-channel T1 or E1 data rates of 64 or 56 kbps for WAN services (Frame Relay or leased line).

Digital T1 or E1 packet voice trunk network modules for Cisco 2600 and 3600 series routers allow enterprises or service providers, using the voice-equipped routers as customer premise equipment (CPE), to deploy digital voice and fax relay. These network modules receive constant bit-rate telephony information over T1 or E1 interfaces and convert that information to a compressed format so that it can be sent over a packet network. The digital T1 or E1 packet voice trunk network modules can connect either to a PBX (or similar telephony device) or to a CO to provide PSTN connectivity. One digital T1 or E1 packet voice trunk network module can be installed in a Cisco 2600 series router or in a Cisco 3620 router. A Cisco 3640 router can support three network modules, and a Cisco 3660 router can support up to six network modules.

The MFT VWICs that are used in the packet voice trunk network modules are available in one- and two-port configurations for T1 and for E1, and in two-port configurations with drop-and-insert capability for T1 and E1. MFTs support the following kinds of traffic:

- Data. As WICs for T1 or E1 applications, including fractional data line use, the T1 version includes a fully managed DSU/CSU, and the E1 version includes a fully managed DSU.
- Packet voice. As VWICs included with the digital T1 or E1 packet voice trunk network module to provide connections to PBXs and COs, the MFTs enable packet voice applications.
- Multiplexed voice and data. Some two-port T1 or E1 VWICs can provide drop-and-insert multiplexing services with integrated DSU/CSUs. For example, when used with a digital T1 packet voice trunk network module, drop-and-insert allows 64-kbps DS0 channels to be taken from one T1 and digitally cross-connected to 64-kbps DS0 channels on another T1. Drop and insert, sometimes called TDM cross-connect, uses circuit switching rather than the DSPs that VoIP technology employs. (Drop-and-insert is described in the “Configuring Trunk Connections and Trunk Conditioning Features” chapter in this configuration guide.)

The digital T1 or E1 packet voice trunk network module contains five 72-pin Single In-line Memory Module (SIMM) sockets or banks, numbered 0 through 4, for PVDMs. Each socket can be filled with a single 72-pin PVDM, and there must be at least one packet voice data module (PVDM-12) in the network module to process voice calls. Each PVDM holds three digital signal processors (DSPs), so with five PVDM slots populated, a total of 15 DSPs are provided. High-complexity codecs support two simultaneous calls on each DSP, and medium-complexity codecs support four calls on each DSP. A digital T1 or E1 packet voice trunk network module can support the following numbers of channels:

- When the digital T1 or E1 packet voice trunk network module is configured for high-complexity codec mode, up to six voice or fax calls can be completed per PVDM-12, using the following codecs: G.711, G.726, G.729, G729 Annex A (E1), G.729 Annex B, G.723.1, G723.1 Annex A (T1), G.728, and fax relay.
- When the digital T1 or E1 packet voice trunk network module is configured for medium-complexity codec mode, up to 12 voice or fax calls can be completed per PVDM-12, using the following codecs: G.711, G.726, G.729 Annex A, G.729 Annex B with Annex A, and fax relay.

For more information, refer to the following publications:

- *Cisco 2600 Series Hardware Installation Guide*
- *Cisco 3600 Series Hardware Installation Guide*
- *Cisco Network Module Hardware Installation Guide*
- Cisco IOS Release 12.0(7)T online document *Configuring 1- and 2-Port T1/E1 Multiflex Voice/WAN Interface Cards on Cisco 2600 and 3600 Series Routers*

Cisco MC3810 Multiservice Concentrator

To support a T1 or E1 digital voice interface, the Cisco MC3810 multiservice concentrator must be equipped with a digital voice interface card (DVM). The DVM interfaces with a digital PBX, channel bank, or video codec. It supports up to 24 channels of compressed digital voice at 8 kbps, or it can cross-connect channelized data from user equipment directly onto the router’s trunk port for connection to a carrier network.

The DVM is available with a balanced interface using an RJ-48 connector or with an unbalanced interface using Bayonet-Neill-Concelman (BNC) connectors.

Optional HCMs can replace standard VCMs to operate according to the voice compression coding algorithm (codec) specified when the Cisco MC3810 multiservice concentrator is configured. The HCM2 provides 4 voice channels at high codec complexity and 8 channels at medium complexity. The

HCM6 provides 12 voice channels at high complexity and 24 channels at medium complexity. You can install one or two HCMs in a Cisco MC3810, but an HCM can not be combined with a VCM in the same chassis.

For more information, refer to the following publications:

- *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*
- *Overview of the Cisco MC3810 Series*
- *Configuring Cisco MC3810 Series Concentrators to Use High-Performance Compression Modules*

Cisco AS5300 Universal Access Server

The Cisco AS5300 Universal Access Server includes three expansion slots. One slot is for either an Octal T1/E1/PRI feature card (eight ports) or a Quad T1/E1/PRI feature card (four ports), and the other two can be used for voice/fax or modem feature cards. Because a single voice/fax feature card (VFC) can support up to 48 (T1) or 60 (E1) voice calls, the Cisco AS5300/Voice Gateway system can support a total of 96 or 120 simultaneous voice calls. The use of VFCs requires Cisco IOS release 12.0.2XH or later.

Cisco AS5300 VFCs are coprocessor cards, each with a powerful reduced instruction set computing (RISC) engine and dedicated, high-performance DSPs to ensure predictable, real-time voice processing. The design couples this coprocessor with direct access to the Cisco AS5300 routing engine for streamlined packet forwarding.

For more information, refer to the following publications:

- *Cisco AS5300 Chassis Installation Guide*
- *Cisco AS5300 Module Installation Guide*

Cisco AS5800 Universal Access Server

The Cisco AS5800 Universal Access Server consists of two primary system components: the Cisco 5814 dial shelf (DS), which holds channelized trunk cards and connects to the PSTN, and the Cisco 7206 router shelf (RS), which holds port adapters and connects to the IP backbone.

The dial shelf acts as the access concentrator by accepting and consolidating all types of remote traffic, including voice, dial-in analog and digital ISDN data, and industry-standard WAN and remote connection types. The dial shelf also contains controller cards voice feature cards, modem feature cards, trunk cards, and dial shelf interconnect cards.

One or two dial shelf controllers (DSCs) provide clock and power control to the dial shelf cards. Each DSC contains a block of logic that is referred to as the common logic and system clocks. This block of logic can use a variety of sources to generate the system timing, including an E1 or T1/T3 input signal from the BNC connector on the DSC's front panel. The configuration commands for the master clock specify the various clock sources and a priority for each source (see the "Clock Sources on Digital T1/E1 Voice Ports" section on page 66).

The Cisco AS5800 voice feature card is a multi-DSP coprocessing board and software package that adds VoIP capabilities to the Cisco AS5800 platform. The Cisco AS5800 voice feature card, when used with other cards such as LAN/WAN and modem cards, provides a gateway for up to 192 packetized voice/fax calls and 360 data calls per card. A Cisco AS5800 can support up to 1,344 voice calls in split-dial-shelf configuration with two 7206VXR router shelves.

For more information, refer to the following publications:

- *Cisco AS5800 Universal Access Server Operation, Administration, Maintenance, and Provisioning Guide*
- *Cisco AS5800 Access Server Hardware Installation Guide*

Cisco 7200 and Cisco 7500 Series Routers

Cisco 7200 and Cisco 7500 series routers support multimedia routing and bridging with a wide variety of protocols and media types. The Cisco 7000 family versatile interface processor (VIP) is based on a RISC engine optimized for I/O functions. To this engine are attached one or two port adapters or daughter boards, which provide the media-specific interfaces to the network. The network interfaces provide connections between the routers' peripheral component interconnect (PCI) buses and external networks. Port adapters can be placed in any available port adapter slot, in any desired combination.

T1/E1 high-capacity digital voice port adapters for Cisco 7200 and Cisco 7500 series routers allow enterprises or service providers, using the equipped routers as customer premise equipment, to deploy digital voice and fax relay. These port adapters receive constant bit-rate telephony information over T1/E1 interfaces and can convert that information to a compressed format for transmission as voice over IP (VoIP). Two types of digital voice port adapters are supported on Cisco 7200 and Cisco 7500 series routers: two-port high-capacity (up to 48 or 120 channels of compressed voice, depending on codec choice), and two-port moderate capacity (up to 24 or 48 channels of compressed voice). These single-width port adapters incorporate two universal ports configurable for either T1 or E1 connection, for use with high-performance digital signal processors (DSPs). Integrated CSU/DSUs, echo cancellation, and DS0 drop-and-insert functionality eliminate the need for external line termination devices and multiplexers.

For more information, refer to the following publications:

- *Cisco 7200 VXR Installation and Configuration Guide*
- *Cisco 7500 Series Installation and Configuration Guide*
- *Two-Port T1/E1 Moderate-Capacity and High-Capacity Digital Voice Port Adapter Installation and Configuration*

Note For current information about supported hardware, see the release notes for the platform and Cisco IOS release being used.

Configuring Basic Parameters on Digital T1/E1 Voice Ports

This section describes commands for basic digital voice port configuration. Make sure you have all the data recommended in the "Preparing Information to Configure Digital Voice Ports" section on page 56 before starting this procedure.

The basic steps for configuring digital voice ports are described in the next three sections. They are grouped by the configuration mode from which they are executed, as follows:

- Configuring Codec Complexity for Digital T1/E1 Voice Ports, page 62

  Codec complexity refers to the amount of processing power assigned to codec processing on a voice port. On most router platforms that support codec complexity, codec complexity is selected in voice card configuration mode, although it is selected in DSP interface mode on the Cisco 7200 and 7500 series. The value configured for codec complexity establishes the choice of codecs that are available on the dial peers. See the *Configuring Dial Plans, Dial Peers, and Digit Manipulation* chapter in this configuration guide for more information about configuring dial peers.

- Configuring Controller Settings for Digital T1/E1 Voice Ports, page 65

  Specific line characteristics must be configured to match those of the PSTN line that is being connected to the voice port. These are typically configured in controller configuration mode.

- Configuring Basic Voice Port Parameters for Digital T1/E1 Voice Ports, page 76

  Voice port configuration mode allows many of the basic voice call attributes to be configured to match those of the PSTN or PBX connection being made on this voice port.

In addition to the basic voice port parameters, there are additional commands that allow for the fine-tuning of the voice port configurations or for configuration of optional features. In most cases, the default values for these commands are sufficient for establishing voice port configurations. If it is necessary to change some of these parameters to improve voice quality or to match parameters in proprietary PBXs to which you are connecting, use the commands in the "Fine-Tuning Analog and Digital Voice Ports" section on page 78.

After voice port configuration, make sure the ports are operational by following the steps described in these sections:

- Verifying Analog and Digital Voice-Port Configurations, page 96
- Troubleshooting Analog and Digital Voice Port Configurations, page 107

For more information on voice port commands, refer to the *Cisco IOS Voice, Video, and Fax Command Reference*.

Configuring Codec Complexity for Digital T1/E1 Voice Ports

On the Cisco 2600, 3600, 7200, and 7500 routers, codec complexity can be configured separately for each T1/E1 digital packet voice trunk network module or port adapter. On a Cisco MC3810 multiservice concentrator, only a single codec complexity setting is used, even when two HCMs are installed. The value specified in this task affects the choice of codecs available when the **codec** dial-peer configuration command is configured.

For details on the number of calls that can be handled simultaneously using each of the codec standards, refer to the entries for **codec** and **codec complexity** in the *Cisco IOS Voice, Video, and Fax Command Reference* and to platform-specific product literature.

For more information on codec complexity, see the "Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules" section on page 45.

Two configuration task tables are shown below: one for the Cisco 2600 and 3600 series routers and the Cisco MC3810 concentrator, which use voice card configuration mode, and the second for the Cisco 7200 and 7500 series routers, which use DSP interface configuration mode.

Cisco 2600 and 3600 Series and Cisco MC3810

This procedure applies to voice ports on digital packet voice trunk network modules on Cisco 2600 series and Cisco 3600 series routers, and to voice ports on HCMs on Cisco MC3810 multiservice concentrators.

Note On Cisco 2600 and 3600 series routers with digital T1/E1 packet voice trunk network modules, codec complexity cannot be configured if DS0 groups are configured. Use the **no ds0-group** command to remove DS0 groups before configuring codec complexity.

Note On the Cisco MC3810 multiservice concentrator with high compression modules, check the DSP voice channel activity with the **show voice dsp** command. If any DSP voice channels are in the busy state, you cannot change the codec complexity. When all of the DSP channels are in the idle state, you can make changes to the codec complexity selection.

To configure codec complexity, use the following commands beginning in privileged EXEC mode:

	Command	Purpose
Step 1	`Router#` **`show voice dsp`**	Checks the DSP voice channel activity. If any DSP voice channels are in the busy state, codec complexity cannot be changed. When all of the DSP channels are in the idle state, continue to Step 2.
Step 2	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 3	`Router(config)#` **`voice-card`** *`slot`*	Enters voice card configuration mode for the card or cards in the slot specified. For the Cisco 2600 and 3600 series routers, the *slot* argument ranges from 0 to 5. For the Cisco MC3810 multiservice concentrator, *slot* must be 0.
Step 4	`Router(config-voicecard)#` **`codec complexity`** `{`**`high`** `|` **`med`**`}`	Specifies codec complexity based on the codec standard being used. This setting restricts the codecs available in dial peer configuration. All voice cards in a router must use the same codec complexity setting. The keywords are as follows: • **high**—(Optional) Specifies up to six voice or fax calls completed per PVDM-12, using the following codecs: G.711, G.726, G.729, G.729 Annex B, G.723.1, G.723.1 Annex A, G.728, and fax relay. • **med**—(Optional) Supports up to 12 voice or fax calls completed per PVDM-12, using the following codecs: G.711, G.726, G.729 Annex A, G.729 Annex B with Annex A, and fax relay. The default is **med**. **Note** On the Cisco MC3810 multiservice concentrator, this command is valid only with one or more HCMs installed, and voice card 0 must be specified. If two HCMs are installed, this command configures both HCMs at once.

Cisco AS5300 Universal Access Server

Codec support on the Cisco AS5300 universal access server is determined by the capability list on the voice feature card, which defines the set of codecs that can be negotiated for a voice call. The capability list is created and populated when VCWare is unbundled and DSPWare is added to VFC Flash memory. The capability list does not indicate codec preference; it simply reports the codecs that are available. The session application decides which codec to use. Codec support is configured on dial peers rather than on voice ports; see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide.

Cisco AS5800 Universal Access Server

Selection of codec support on Cisco AS5800 access servers is made during dial peer configuration. See the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide.

Cisco 7200 Series and Cisco 7500 Series Routers

On Cisco 7200 series and Cisco 7500 series routers, codec complexity is configured on the DSP interface.

Note Check the DSP voice channel activity using the **show interfaces dspfarm** command. If any DSP voice channels are in the busy state, codec complexity cannot be changed. When all of the DSP channels are in the idle state, changes can be made to the codec complexity selection.

To configure the DSP interface, use the following commands beginning in privileged EXEC mode:

	Command	Purpose
Step 1	`Router#` **`show interfaces dspfarm`**	Displays the DSP voice channel activity. If any DSP voice channels are in the busy state, codec complexity cannot be changed. When all of the DSP channels are in the idle state, continue to Step 2.
Step 2	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 3	**Cisco 7200 series** `Router(config)#` **`dspint dspfarm`** *`slot/port`* **Cisco 7500 series** `Router(config)#` **`dspint dspfarm`** *`slot/port-adapter/port`*	Enters DSP interface configuration mode. The arguments are as follows: • *slot/port*—Specifies the slot and port numbers of the interface. • *adapter/port*—Specifies the adapter and port numbers of the interface.

	Command	Purpose
Step 4	`Router(config-dspfarm)#` **`codec {high	med}`**	Specifies the codec complexity based on the codec standard being used. The keyword specified for **codec** affects the choice of codecs available when the **codec** dial-peer configuration command is used. The keywords are as follows: • **high**—Supports two voice channels encoded in any of the following formats: G.711, G.726, G.729, G.729 Annex B, G.723.1, G.723.1 Annex A, G.728, and fax relay. • **med**—(default) Supports up to four calls using the following codecs: G.711, G.726, G.729 Annex A, G.729 Annex B with Annex A, and fax relay.
Step 5	`Router(config-dspfarm)#` **`description`**	Enters a string to include descriptive text about this DSP interface connection. This information is displayed in the output for **show** commands and does not affect the operation of the interface in any way.

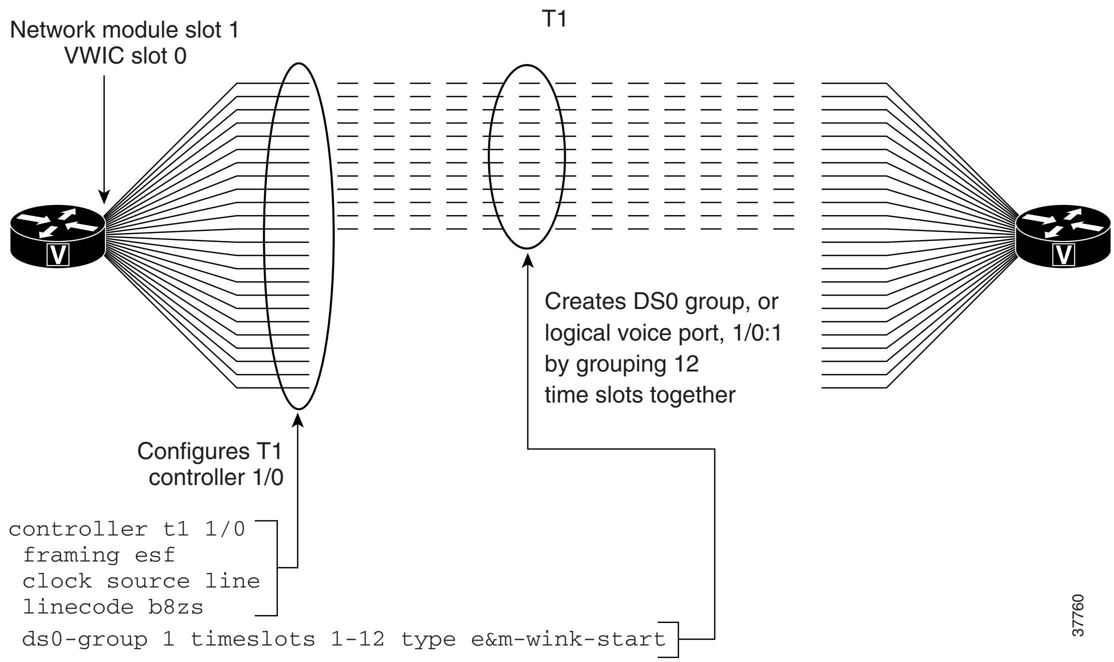
Configuring Controller Settings for Digital T1/E1 Voice Ports

The purpose of configuring controllers for digital T1/E1 voice ports is to match the configuration of the router to the line characteristics of the telephony network connection being made so that voice and signaling can be transferred between them and so that logical voice ports, or DS0 groups, may be established.

Figure 16 shows how a **ds0-group** command gathers some of the DS0 time slots from a T1 line into a group that becomes a single logical voice port, which can later be addressed as a single entity in voice port configurations. Other DS0 groups for voice can be created from the remaining time slots shown in the figure, or the time slots can be used for data or serial pass-through.

Note that all the controller commands in Figure 16 other than **ds0-group** apply to all the time slots in the T1.

Figure 16 T1 Controller Configuration on Cisco 2600 or 3600 Series Routers



Voice port controller configuration includes setting the parameters described in the following sections:

- Framing Formats on Digital T1/E1 Voice Ports
- Clock Sources on Digital T1/E1 Voice Ports
- Line Coding on Digital T1/E1 Voice Ports
- DS0 Groups on Digital T1/E1 Voice Ports

Another controller command that might be needed, **cablelength**, is discussed in the *Cisco IOS Interface Command Reference*, Release 12.2.

Framing Formats on Digital T1/E1 Voice Ports

The framing format parameter describes the way that bits are robbed from specific frames to be used for signaling purposes. The controller must be configured to use the same framing format as the line from the PBX or CO that connects to the voice port you are configuring.

Digital T1 lines use super frame (SF) or extended super frame (ESF) framing formats. SF provides two-state, continuous supervision signaling, in which bit values of 0 are used to represent on-hook and bit values of 1 are used to represent off-hook. ESF robs four bits instead of two, yet has little impact on voice quality. ESF is required for 64-kbps operation on DS0 and is recommended for Primary Rate Interface (PRI) configurations.

E1 lines can be configured for cyclic redundancy check (CRC4) or no cyclic redundancy check, with an optional argument for E1 lines in Australia.

Clock Sources on Digital T1/E1 Voice Ports

Digital T1/E1 interfaces use timers called *clocks* to ensure that voice packets are delivered and assembled properly. All interfaces handling the same packets must be configured to use the same source of timing so that packets are not lost or delivered late. The timing source that is configured can be external (from the line) or internal to the router’s digital interface.

If the timing source is internal, timing derives from the onboard phase-lock loop (PLL) chip in the digital voice interface. If the timing source is line (external), then timing derives from the PBX or PSTN CO to which the voice port is connected. It is generally preferable to derive timing from the PSTN because their clocks are maintained at an extremely accurate level. This is the default setting for the clocks. When two or more controllers are configured, one should be designated as the primary clock source; it will drive the other controllers.

The **line** keyword specifies that the clock source is derived from the active line rather than from the free-running internal clock. The following rules apply to clock sourcing on the controller ports:

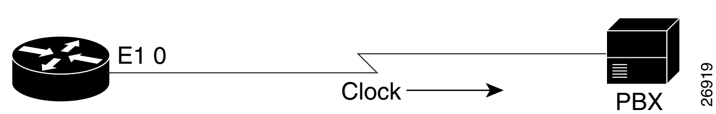
- When both ports are set to line clocking with no primary specification, port 0 is the default primary clock source and port 1 is the default secondary clock source.
- When both ports are set to line and one port is set as the primary clock source, the other port is by default the backup or secondary source and is loop-timed.
- If one port is set to clock source line or clock source line primary and the other is set to clock source internal, the internal port recovers clock from the clock source line port if the clock source line port is up. If it is down, then the internal port generates its own clock.
- If both ports are set to clock source internal, there is only one clock source: internal.

This section describes the five basic timing scenarios that can occur when a digital voice port is connected to a PBX or CO. In all the examples that follow, the PSTN (or CO) and the PBX are interchangeable for purposes of providing or receiving clocking.

- Single Voice Port Providing Clocking—In this scenario, the digital voice hardware is the clock source for the connected device, as shown in Figure 17. The PLL generates the clock internally and drives the clocking on the line. Generally, this method is useful only when connecting to a PBX, key system, or channel bank. A Cisco VoIP gateway rarely provides clocking *to* the CO because CO clocking is much more reliable. The following configuration sets up this clocking method for a digital E1 voice port:

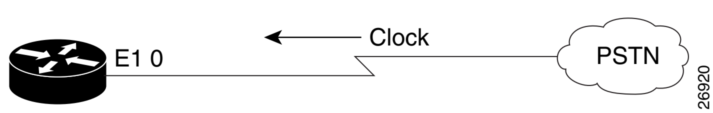
```
controller E1 1/0
 framing crc4
 linecoding hdb3
 clock source internal
 ds0-group timeslots 1-15 type e&m-wink-start
```

Figure 17 Single Voice Port Providing Clocking



- Single Voice Port Receiving Internal Clocking—In this scenario, the digital voice hardware receives clocking from the connected device (CO telephony switch or PBX) (see Figure 18). The PLL clocking is driven by the clock reference on the receive (Rx) side of the digital line connection.

Figure 18 Single E1 Port Receiving Clocking from the Line



The following configuration sets up this clocking method:

```
controller T1 1/0
 framing esf
 linecoding ami
 clock source line
 ds0-group timeslots 1-12 type e&m-wink-start
```

- Dual Voice Ports Receiving Clocking from the Line—In this scenario, the digital voice port has two reference clocks, one from the PBX and another from the CO, as shown in Figure 19. Because the PLL can derive clocking from only one source, this case is more complex than the two preceding examples.

  Before looking at the details, consider the following as they pertain to the clocking method:

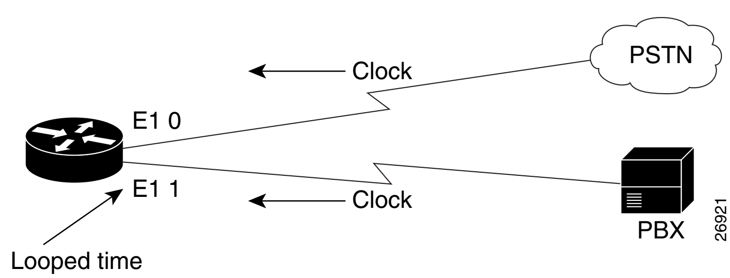
  – Looped-time clocking: The voice port takes the clock received on its Rx (receive) pair and regenerates it on its Tx (transmit) pair. While the port receives clocking, the port is not driving the PLL on the card but is "spoofing" (that is, fooling) the port so that the connected device has a viable clock and does not see slips (that is, loss of data bits). PBXs are not designed to accept slips on a T1 or E1 line, and such slips cause a PBX to drop the link into failure mode. While in looped-time mode, the router often sees slips, but because these are controlled slips, they usually do not force failures of the router's voice port.

  – Slips: These messages indicate that the voice port is receiving clock information that is out of phase (out of synchronization). Because the router has only a single PLL, it can experience controlled slips while it receives clocking from two different time sources. The router can usually handle controlled slips because its single-PLL architecture anticipates them.

Note Physical layer issues, such as bad cabling or faulty clocking references, can cause slips. Eliminate these slips by addressing the physical layer or clock reference problems.

In the dual voice ports receiving clocking from the line scenario, the PLL derives clocking from the CO and puts the voice port connected to the PBX into looped-time mode. This is usually the best method because the CO provides an excellent clock source (and the PLL usually requires that the CO provide that source) and a PBX usually must receive clocking from the other voice port.

Figure 19 Dual E1 Ports Receiving Clocking from the Line



The following configuration sets up this clocking method:

```
controller E1 1/0 << description - connected to the CO
 framing crc4
 linecoding hdb3
 clock source line primary
 ds0-group timeslots 1-15 type e&m-wink-start
!
```

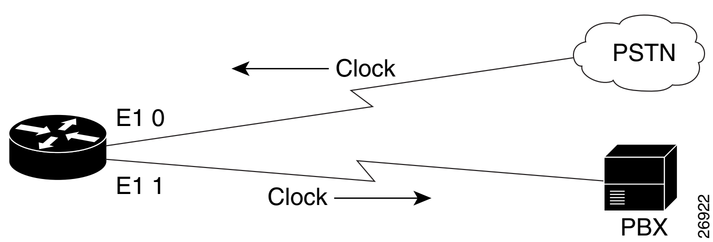
```
controller E1 1/1 << description - connected to the PBX
 framing crc4
 linecoding hdb3
 clock source line
 ds0-group timeslots 1-15 type e&m-wink-start
```

The **clock source line primary** command tells the router to use this voice port to drive the PLL. All other voice ports configured as **clock source line** are then put into an implicit loop-timed mode. If the primary voice port fails or goes down, the other voice port instead receives the clock that drives the PLL. In this configuration, port 1/1 might see controlled slips, but these should not force it down. This method prevents the PBX from seeing slips.

Note When terminating two T1/E1 lines on a two-port interface card, such as the VWIC-2MFT, if both controllers are set for line clocking but the lines are not within clocking tolerance of one another, one of the controllers is likely to experience slips. To prevent slips, ensure that the two T1/E1 lines are within clocking tolerance of one another, even if the lines are from different providers.

- Dual Voice Ports (One Receives Clocking and One Provides Clocking)—In this scenario, the digital voice hardware receives clocking for the PLL from E1 0 and uses this clock as a reference to clock E1 1 (see Figure 20). If controller E1 0 fails, the PLL internally generates the clock reference to drive E1 1.

Figure 20 Dual E1 ports—One Receiving and One Providing Clocking

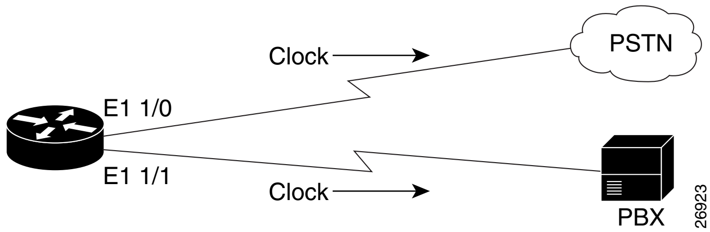


The following configuration sets up this clocking method:

```
controller E1 1/0
 framing crc4
 linecoding hdb3
 clock source line
 ds0-group timeslots 1-15 type e&m-wink-start
!
controller E1 1/1
 framing crc4
 linecoding hdb3
 clock source internal
 ds0-group timeslots 1-15 type e&m-wink-start
```

- Dual Voice Ports (Router Provides Both Clocks)—In this scenario, the router generates the clock for the PLL and, therefore, for both voice ports (see Figure 21).

Figure 21 Dual E1 Ports—both Clocks from the Router



The following configuration sets up this clocking method:

```
controller E1 1/0
 framing crc4
 linecoding hdb3
 clock source internal
 ds0-group timeslots 1-15 type e&m-wink-start
!
controller E1 1/1
 framing esf
 linecoding b8zs
 clock source internal
 ds0-group timeslots 1-15 type e&m-wink-start
```

Line Coding on Digital T1/E1 Voice Ports

Digital T1/E1 interfaces require that line encoding be configured to match that of the PBX or CO that is being connected to the voice port. Line encoding defines the type of framing used on the line.

T1 line encoding methods include alternate mark inversion (AMI) and binary 8 zero substitution (B8ZS). AMI is used on older T1 circuits and references signal transitions with a binary 1, or "mark." B8ZS, a more reliable method, is more popular and is recommended for PRI configurations as well. B8ZS encodes a sequence of eight zeros in a unique binary sequence to detect line-coding violations.

Supported E1 line encoding methods are AMI and high-density bipolar 3 (HDB3), which is a form of zero-suppression line coding.

DS0 Groups on Digital T1/E1 Voice Ports

For digital voice ports, a single command, **ds0-group**, performs the following functions:

- Defines the T1/E1 channels for compressed voice calls.
- Automatically creates a logical voice port.

  The numbering for the logical voice port created as a result of this command is *controller:ds0-group-no*, where *controller* is defined as the platform-specific address for a particular controller. On a Cisco 3640 router, for example, **ds0-group 1 timeslots 1-24 type e&m-wink** automatically creates the voice port 1/0:1 when issued in the configuration mode for controller 1/0. On a Cisco MC3810 universal concentrator, when you are in the configuration mode for controller 0, the command **ds0-group 1 timeslots 1-24 type e&m-wink** creates logical voice port 0:1.

  To map individual DS0s, define additional DS0 groups under the T1/E1 controller, specifying different time slots. Defining additional DS0 groups also creates individual DS0 voice ports.

- Defines the emulated analog signaling method that the router uses to connect to the PBX or PSTN.

Most digital T1/E1 connections used for switch-to-switch (or switch-to-router) trunks are E&M connections, but FXS and FXO connections are also supported. These are normally used to provide emulated-OPX (Off-Premises eXtension) from a PBX to remote stations. FXO ports connect to FXS ports. The FXO or FXS connection between the router and switch (CO or PBX) must use matching signaling, or calls cannot connect properly. Either ground start or loop start signaling is appropriate for these connections. Ground start provides better disconnect supervision to detect when a remote user has hung up the telephone, but ground start is not available on all PBXs.

Digital ground start differs from digital E&M because the A and B bits do not track each other as they do in digital E&M signaling (that is, A is not necessarily equal to B). When the CO delivers a call, it *seizes* a channel (goes off-hook) by setting the A bit to 0. The CO equipment also simulates ringing by toggling the B bit. The terminating equipment goes off-hook when it is ready to answer the call. Digits are usually not delivered for incoming calls.

E&M connections can use one of three different signaling types to acknowledge on-hook and off-hook states: wink start, immediate start, and delay start. E&M wink start is usually preferred, but not all COs and PBXs can handle wink start signaling. The E&M connection between the router and switch (CO or PBX) must match the CO or PBX E&M signaling type, or calls cannot be connected properly.

E&M signaling is normally used for trunks. It is normally the only way that a CO switch can provide two-way dialing with Direct Inward Dialing (DID). In all the E&M protocols, off-hook is indicated by A=B=1 and on-hook is indicated by A=B=0 (robbed-bit signaling). If dial pulse dialing is used, the A and B bits are pulsed to indicate the addressing digits. The are several further important subclasses of E&M robbed-bit signaling:

- E&M Wink Start—Feature Group B

  In the original wink start handshaking protocol, the terminating side responds to an off-hook from the originating side with a short wink (transition from on-hook to off-hook and back again). This wink tells the originating side that the terminating side is ready to receive addressing digits. After receiving addressing digits, the terminating side then goes off-hook for the duration of the call. The originating endpoint maintains off-hook for the duration of the call.

- E&M Wink Start—Feature Group D

  In Feature Group D wink start with wink acknowledge handshaking protocol, the terminating side responds to an off-hook from the originating side with a short wink (transition from on-hook to off-hook and back again) just as in the original wink start. This wink tells the originating side that the terminating side is ready to receive addressing digits. After receiving addressing digits, the terminating side provides another wink (called an *acknowledgment wink*) that tells the originating side that the terminating side has received the dialed digits. The terminating side then goes off-hook to indicate connection. This last indication can be due to the ultimate called endpoint's having answered. The originating endpoint maintains an off-hook condition for the duration of the call.

- E&M Immediate Start

  In the immediate-start protocol, the originating side does not wait for a wink before sending addressing information. After receiving addressing digits, the terminating side then goes off-hook for the duration of the call. The originating endpoint maintains off-hook for the duration of the call.

Note Feature Group D is supported on Cisco AS5300 platforms, and on Cisco 2600, 3600, and 7200 series with digital T1 packet voice trunk network modules. Feature Group D is not supported on E1 or analog voice ports.

To configure controller settings for digital T1/E1 voice ports, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	**Cisco 7200 and 7500 series** `Router(config)# card type {t1 | e1} slot`	Defines the card as T1 or E1 and stipulates the location. The keywords and arguments are as follows: • **t1** | **e1**—Defines the type of card. • *slot*—A value from 0 to 5.

Step 2	**Cisco 2600 and 3600 series, Cisco MC3810, and Cisco 7200 series**
	`Router(config)# controller {t1 | e1} slot/port`
	Cisco AS5300
	`Router(config)# controller {t1 | e1} number`
	Cisco AS5800
	`Router(config)# controller {t1 | e1} shelf/slot/port`
	Cisco 7500 series
	`Router(config)# controller {t1 | e1} slot/port-adapter/slot`
		Enters controller configuration mode.
		The keywords and arguments are as follows:
		• **t1** | **e1**—The type of controller.
		• *slot/port*—The backplane slot number and port number for the interface being configured.
		• *number*—The network processor module number; the range is from 0 to 2.
		• *shelf/slot/port*—Indicates the controller ports; the range for *port* is from 0 to 11.

Step 3	**T1**
	`Router(config-controller)# framing {sf | esf}`
	E1
	`Router(config-controller)# framing {crc4 | no-crc4} [australia]`
		Selects frame type for T1 or E1 line.
		The keywords and arguments are as follows:
		T1 lines
		• **sf**—super frame
		• **esf**—extended super frame
		E1 lines
		• **crc4**—Provides 4 bits of error protection.
		• **no-crc4**—Disables **crc4**.
		• **australia**—(Optional) Specifies the E1 frame type used in Australia.
		The default for T1 is **sf**.
		The default for E1 is **crc4**.

	Command	Purpose
Step 4	`Router(config-controller)# clock source {line [primary	secondary]	internal}`	Configures the clock source. The keywords and arguments are as follows: • **line**—Specifies that the PLL on this port derives clocking from the external source to which the port is connected (generally the CO). • **primary**—(Optional) Specifies that the PLL on this port derives clocking from the external source and puts the other port (generally connected to the PBX) into looped-time mode. Both ports are configured with **line**, but only the port connected to the external source is configured with **primary**. • **secondary**—(Optional) Indicates a backup external source for clocking if the primary clocking shuts down. Configure the **clock source line secondary** command on the controller that has the next-best-known clocking. • **internal**—(Optional) Specifies that the clock is generated from the voice port's internal PLL. For more information about clock sources, see the "Clock Sources on Digital T1/E1 Voice Ports" section on page 66. The default is **line**.
Step 5	**T1 lines** `Router(config-controller)# linecode {ami	b8zs}` **E1 lines** `Router(config-controller)# linecode {ami	hdb3}`	Specifies the line encoding to use. The keywords are as follows: • **ami**—Specifies the alternate mark inversion (AMI) line code type. (T1 and E1) • **b8zs**—Specifies the binary 8 zero substitution (B8ZS) line code type. (T1 only) • **hdb3**—Specifies the high-density bipolar 3 (HDB3) line code type. (E1 only) The default for T1 is **ami**. The default for E1 is **hdb3**.

<table>
<tr><th></th><th>Command</th><th>Purpose</th></tr>
<tr><td>Step 6</td><td>
Cisco 2600 and 3600 Series Routers and Cisco MC3810 Multiservice Concentrators—T1
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {e&m-delay-dial | e&m-fgd | e&m-immediate-start e&m-wink-start | ext-sig | fgd-eana | fxo-ground-start | fxo-loop-start | fxs-ground-start | fxs-loop-start}</code>

Cisco 2600 and 3600 Series Routers and Cisco MC3810 Multservice Concentrators—E1
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {e&m-delay-dial | e&m-immediate-start | e&m-melcas-delay | e&m-melcas-immed | e&m-melcas-wink | e&m-wink-start | ext-sig | fgd-eana | fxo-ground-start | fxo-loop-start | fxo-melcas | fxs-ground-start | fxs-loop-start | fxs-melcas | r2-analog | r2-digital | r2-pulse}</code>

Cisco AS5300 Universal Access Servers—T1
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list [service {data | fax | voice}] [type {e&m-fgb | e&m-fgd | e&m-immediate-start | fxs-ground-start | fxs-loop-start | fgd-eana | fgd-os | r1-itu | sas-ground-start | sas-loop-start | none}]</code>

Cisco AS5300 Universal Access Servers—E1
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {none | p7 | r2-analog | r2-digital | r2-lsv181-digital | r2-pulse}</code>

Cisco AS5800 Universal Access Servers—T1
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {e&m-fgb | e&m-fgd | e&m-immediate-start | fxs-ground-start | fxs-loop-start | fgd-eana | r1-itu | r1-modified | r1-turkey | sas-ground-start | sas-loop-start | none}</code>

Cisco AS5800 Universal Access Servers E1 Voice Ports
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {e&m-fgb | e&m-fgd | e&m-immediate-start | fxs-ground-start | fxs-loop-start | p7 | r2-analog | r2-digital | r2-pulse | sas-ground-start | sas-loop-start | none}</code>

Cisco 7200 and 7500 Series Series Routers T1 and E1 Voice Ports
<code>Router(config-controller)# ds0-group ds0-group-no timeslots timeslot-list type {e&m-delay | e&m-immediate | e&m-wink | fxs-ground-start | fxs-loop-start | fxo-ground-start | fxo-loop-start}</code>
</td><td>
Defines the T1 channels for use by compressed voice calls and the signaling method that the router uses to connect to the PBX or CO.

Note This step shows the basic syntax and signaling types available with the ds0-group command. For the complete syntax, see the Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2.

The keywords and arguments are as follows:
• ds0-group-no—Identifies the DS0 group (number from 0 to 23, for T1, or from 0 to 30, for E1).
• timeslots timeslot-list—Specifies the single time slot number, single range of numbers, or multiple ranges of numbers separated by commas. For T1/E1, allowable values are from 1 to 24. Examples are as follows:
– 2, 3-5
– 1, 7, 9
– 1-12
• service—Indicates the type of calls to be handled by this DS0 group—data, fax, or voice).
• type—Refers to the signaling type of the telephony connection being made. Types include the following:
– e&m-delay-dial—Specifies the originating endpoint that sends an off-hook signal and waits for the off-hook signal followed by an on-hook signal from the destination.
– e&m-fgb—E & M Type II Feature Group B.
– e&m-fgd—E & M Type II Feature Group D.
</td></tr>
</table>

	Command	Purpose
		– **e&m-immediate-start**—E & M Immediate Start. – **e&m-melcas-delay**—E&M Mercury Exchange Limited Channel Associated Signaling (MELCAS) delay start signaling support. – **e&m-melcas-immed**—E&M MELCAS immediate start signaling support. – **e&m-melcas-wink**—E&M MELCAS wink start signaling support. – **e&m-wink-start**—The originating endpoint sends an off-hook signal and waits for a – **ext-sig**—For the specified channel, automatically generates the off-hook signal and stays in the off-hook state. – **fgd-eana**—Feature Group D Exchange Access North American. – **fgd-os**—Feature Group D Operator Services. – **fxo-melcas**—MELCAS Foreign Exchange Office signaling support. – **fxs-melcas**—MELCAS Foreign Exchange Station signaling support. – **fxs-ground-start**—FXS Ground Start. – **fxs-loop-start**—FXS Loop Start. – **none**—Null Signaling for External Call Control. – **p7**—Specifies the p7 switch type. – **r1-itu**—R1 ITU – **sas-ground-start**—SAS Ground Start. – **sas-loop-start**—SAS Loop Start. The **r1** and **r2** keywords refer to line signaling, based on international signaling standards. The **r1 itu** keywords are based on signaling standards in countries besides the United States. An "ITU variant" means that there are multiple R1 standards in a particular country but that Cisco supports the ITU variant.
Step 7	`Router(config-controller)# no shutdown`	Activates the controller.

Configuring Basic Voice Port Parameters for Digital T1/E1 Voice Ports

For FXO and FXS connections the default voice-port parameter values are often adequate. However, for E&M connections, it is important to match the characteristics of your PBX, so voice port parameters may need to be reconfigured from their defaults.

Each voice port that you address in digital voice port configuration is one of the logical voice ports that you created with the **ds0-group** command.

Companding (from *compression* and *expansion*), used in Step 4 of the following table, is the part of the PCM process in which analog signal values are logically rounded to discrete scale-step values on a nonlinear scale. The decimal step number is then coded in its binary equivalent prior to transmission. The process is reversed at the receiving terminal using the same nonlinear scale.

Note The commands, keywords, and arguments that you are able to use may differ slightly from those presented here, based on your platform, Cisco IOS release, and configuration. When in doubt, use Cisco IOS command help (**command ?**) to determine the syntax choices that are available.

To configure basic parameters for digital T1/E1 voice ports, use the following commands beginning in global configuration mode.

	Command	Purpose

Step 1	**Cisco 2600 and 3600 Series Routers**
	`Router(config)# voice-port slot/port:ds0-group-no`
	Cisco MC3810 Multiseries Concentrators
	`Router(config)# voice-port slot:ds0-group-no`
	Cisco AS5300 Universal Access Server
	`Router(config)# voice-port controller:ds0-group-no`
	Cisco AS5800 Universal Access Server
	`Router(config)# voice-port shelf/slot/port:ds0-group-no`
	Cisco 7200 Series Routers
	`Router(config)# voice-port slot/port-adapter:ds0-group-no`
	Cisco 7500 Series Routers
	`Router(config)# voice-port slot/port-adapter/slot:ds0-group-no`
		Enters voice-port configuration mode. The arguments are defined as the following
		• *slot*—Specifies the router location where the network module (Cisco 2600, 3600, and MC3810) or voice port adapter (Cisco AS5300, AS5800, 7200, and 7500) is installed. This is the same number as the controller for the T1/E1 voice port.
		• *port*—Indicates the voice interface card location.
		• *ds0-group-no*—Specifies the logical voice port that was created with the **ds0-group** controller command.
		• *controller*—Indicates the controller for the T1/E1 voice port.
		• *shelf*—Specifies the dial shelf, which is always 0.
		• *port-adapter*—Indicates the port adapter for the voice port.

Step 2	`Router(config-voiceport)# type {1 | 2 | 3 | 5}`	(E&M only) Specifies the type of E&M interface to which this voice port is connected. See Table 5 for an explanation of E&M types. The default is 1.

	Command	Purpose
Step 3	`Router(config-voiceport)# cptone locale`	Selects a two-letter **locale** keyword for the voice call progress tones and other locale-specific parameters to be used on this voice port. Voice call progress tones include dial tone, busy tone, and ringback tone, which vary with geographical region. Other parameters include ring cadence and compand type. Cisco routers comply with the ISO3166 locale name standards; to see valid choices, enter a question mark (?) following the **cptone** command. The default is **us**.
Step 4	`Router(config-voiceport)# compand-type {u-law | a-law}`	(Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators only) Specifies the companding standard used. This command is used in cases when the DSP is not used, such as local cross-connects, and overwrites the **compand-type** value set by the **cptone** command. The keywords are as follows: • **a-law**—Specifies the ITU-T PCM a-law companding standard used primarily in Europe. The default for E1 is **a-law**. • **u-law**—Specifies the ITU-T PCM mu-law companding standard used in North America and Japan. The default for T1 is **u-law**. **Note** If you have a Cisco MC3810 multiservice concentrator or Cisco 3660 router, the **compand-type a-law** command must be configured on the analog ports only. The Cisco 2660, 3620, and 3640 routers do not require the **compand-type a-law** command configured, however, if you request a list of commands, the **compand-type a-law** command will display.

Step 5	**Cisco 2600 series and 3600 series**
	`Router(config-voiceport)# ring frequency {25 | 50}`
	
	Cisco MC3810
	`Router(config-voiceport)# ring frequency {20 | 30}`
		(FXS only) Selects the ring frequency, in hertz, used on the FXS interface. This number must match the connected telephony equipment, and can be country-dependent. If not set properly, the attached telephony device may not ring or it may buzz.
		
		The default is 25 on the Cisco 2600 and 3600 series routers and 20 on the Cisco MC3810 multiservice concentrators.

	Command	Purpose
Step 6	`Router(config-voiceport)#` **ring number** *number*	(FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router. The default is 1.
Step 7	`Router(config-voiceport)#` **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse interval*]}	(FXS only) Specifies an existing pattern for ring, or defines a new one. Each pattern specifies a ring-pulse time and a ring-interval time. The keywords and arguments are as follows: • **pattern01** through **pattern12**—Specifies preset ring cadence patterns. Enter **ring cadence ?** to see ring pattern explanations. • **define** *pulse interval*—Specifies a user-defined pattern as follows: – *pulse* is a number (1 or 2 digits from 1 to 50) specifying ring pulse (on) time in hundreds of milliseconds. – *interval* is a number (1 or 2 digits from 1 to 50) specifying ring interval (off) time in hundreds of milliseconds. The default is the pattern specified by the configured **cptone locale** command.
Step 8	`Router(config-voiceport)#` **description** *string*	Attaches a text string to the configuration that describes the connection for this voice port. This description appears in various displays and is useful for tracking the purpose or use of the voice port. The *string* argument is a character string from 1 to 255 characters in length. The default is that no description is attached to the configuration.
Step 9	`Router(config-voiceport)#` **no shutdown**	Activates the voice port.

Fine-Tuning Analog and Digital Voice Ports

Normally, default parameter values for voice ports are sufficient for most networks. Depending on the specifics of your particular network, however, you may need to adjust certain parameters that are configured on voice ports. Collectively, these commands are referred to as voice port tuning commands.

Note The commands, keywords, and arguments that you are able to use may differ slightly from those presented here, based on your platform, Cisco IOS release, and configuration. When in doubt, use Cisco IOS command help (**command ?**) to determine the syntax choices that are available.

The voice port tuning commands are grouped into these categories and explained in the following sections:

Full descriptions of the commands in this section can be found in the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2.

Auto Cut-Through Command

The **auto-cut-through** command allows you to connect to PBXs that do not provide an M-lead response.

To configure auto-cut-through, use the following command in voice-port configuration mode:

Command	Purpose
`Router(config-voiceport)# auto-cut-through`	(E&M only) Enables call completion on a router when a PBX does not provide an M-lead response.

Bit Modification Commands for Digital Voice Ports

The bit modification commands for digital voice ports modify sent or received bit patterns. Different versions of E&M use different ABCD signaling bits to represent idle and seize. For example, North American CAS E&M represents idle as 0XXX and seize as 1XXX, where X indicates that the state of the BCD bits is ignored. In MELCAS E&M, idle is 1101 and seize is 0101. The commands in this section are provided to modify bit patterns to match particular E&M schemes.

To manipulate bit patterns for digital voice ports, use the following commands as necessary, in voice-port configuration mode:

	Command	Purpose
Step 1	Router(config-voiceport)# **condition** {**tx-a-bit** | **tx-b-bit** | **tx-c-bit** | **tx-d-bit**} {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} {**on** | **off** | **invert**}	Manipulates sent or received bit patterns to match expected patterns on a connected device. Repeat the command for each transmit and/or receive bit to be modified, but be careful not to destroy the information content of the bit pattern. The default is that the signaling format is not manipulated (for all transmit or receive A, B, C, and D bits).
		The keywords are as follows: • **on**—Sets the bit to 1 permanently. • **off**—Sets the bit to 0 permanently. • **invert**—Changes the state to the opposite of the original transmit or receive state. **Note** The **show voice port** command reports at the protocol level, and the **show controller** command reports at the driver level. The driver is not notified of any bit manipulation using the **condition** command. As a result, the **show controller** command output does not account for the bit conditioning.
Step 2	Router(config-voiceport)# **define** {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}	(Digital E1 E&M voice ports on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators only) Defines specific transmit or receive signaling bits to match the bit patterns required by a connected device for North American E&M and E&M MELCAS voice signaling, if patterns different from the preset defaults are required. Also specifies which bits a voice port monitors and which bits it ignores, if patterns that are different from the defaults are required. See the **define** command for the default signaling patterns as defined in American National Standards Institute (ANSI) and code excited linear prediction compression (CEPT) standards. The keywords are as follows: • **tx-bits**—Indicates the pattern applies to transmit signaling bits.

	Command	Purpose
		• **rx-bits**—Indicates the pattern applies to receive signaling bits • **seize**—Indicates that the pattern represents line seizure. • **idle**—Indicates that the pattern represents an idle condition. • **0000**...**1111**—Represents the bit pattern to use.
Step 3	`Router(config-voiceport)# ignore {rx-a-bit | rx-b-bit | rx-c-bit | rx-d-bit}`	(Digital E1 E&M voice ports on Cisco 2600 and 3600 series routers and Cisco MC3810 multiservice concentrators only) Configures the voice port to ignore the specified receive bit for North American E&M or E&M MELCAS, if patterns different from the defaults are required. See the command reference for the default signaling patterns as defined in ANSI and CEPT standards.

Calling Number Outbound Commands

On the Cisco AS5300 universal access server platform, if T1 CAS is configured with the Feature Group-D (FGD)—Exchange Access North American (FGD-EANA) signaling, the automatic number identification (ANI) can be sent for outgoing calls by using the **calling-number outbound** command.

FGD-EANA is a FGD signaling protocol of type EANA, which provides certain call services, such as emergency (USA 911) calls. ANI is an SS7 (Signaling System 7) feature in which a series of digits, analog or digital, are included in the call to identify the telephone number of the calling device. In other words, ANI identifies the number of the calling party. ANI digits are used for billing purposes by Internet service providers (ISPs), among other things. The commands in this section can be issued in voice-port or dial-peer mode, because the syntax is the same.

To configure your digital T1/E1 packet voice trunk network module to generate outbound ANI digits on a Cisco AS5300 universal access server, use the following commands in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)# ` **`calling-number outbound range`** *`string1 string2`*	(Cisco AS5300 universal access server only) Specifies ANI to be sent out when the T1-CAS **fgd-eana** command is configured as signaling type. The *string1* and *string2* arguments are valid E.164 telephone number strings. Both strings must be of the same length and cannot be more than 32 digits long. Only the last four digits are used for specifying the range (*string1* to *string2*) and for generating the sequence of ANI by rotating through the range until *string2* is reached and then starting from *string1* again. If strings are less than four digits in length, then entire strings are used.
Step 2	`Router(config-voiceport)# ` **`calling-number outbound sequence`** `[`*`string1`*`] [`*`string2`*`] [`*`string3`*`] [`*`string4`*`] [`*`string5`*`]`	(Cisco AS5300 universal access server only) Specifies ANI to be sent out when the T1-CAS **fgd-eana** command is configured as signaling type. This option configures a sequence of discrete strings (*string1...string5*) to be passed out as ANI for successive calls using the dial peer or voice port. Limit is five (5) strings. All strings must be valid E.164 numbers, up to 32 digits in length.
Step 3	`Router(config-voiceport)# ` **`calling-number outbound null`**	(Cisco AS5300 universal access server only) Suppresses ANI. No ANI is passed when this voice port is selected.

Disconnect Supervision Commands

PBX and PSTN switches use several different methods to indicate that a call should be disconnected because one or both parties have hung up. The commands in this section are used to configure the router to recognize the type of signaling in use by the PBX or PSTN switch connected to the voice port. These methods include the following:

- Battery reversal disconnect
- Battery denial disconnect
- Supervisory tone disconnect (STD)

Battery reversal occurs when the connected switch changes the polarity of the line in order to indicate changes in call state (such as off-hook or, in this case, call disconnect). This is the signaling looked for when the **battery reversal** command is enabled on the voice port, which is the default configuration.

Battery denial (sometimes called *power denial*) occurs when the connected switch provides a short (approximately 600 ms) interruption of line power to indicate a change in call state. This is the signaling looked for when the **supervisory disconnect** command is enabled on the voice port, which is the default configuration.

Supervisory tone disconnect occurs when the connected switch provides a special tone to indicate a change in call state. Some PBXs and PSTN CO switches provide a 600-millisecond interruption of line power as a supervisory disconnect, and others provide supervisory tone disconnect (STD). This is the signal that the router is looking for when the **no supervisory disconnect** command is configured on the voice port.

Note In some circumstances, you can use the FXO Disconnect Supervision feature to enable analog FXO ports to monitor call progress tones for disconnect supervision that are returned from a PBX or from the PSTN. For more information, see the "FXO Supervisory Disconnect Tone Commands" section on page 84.

To change parameters related to disconnect supervision, use the following commands as appropriate, in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)# no battery-reversal`	(Analog only) Enables battery reversal. The default is that battery reversal is enabled. • For FXO ports—Use the **no battery-reversal** command to configure a loop-start voice port not to disconnect when it detects a second battery reversal. The default is to disconnect when a second battery reversal is detected. This functionality is supported on Cisco MC3810 analog voice ports; on Cisco 1750, Cisco 2600 series, and Cisco 3600 series routers, only analog voice ports on VIC-2FXO cards are able to detect battery reversal. – Also use the **no battery-reversal** command when a connected FXO port does not support battery reversal detection. • For FXS ports—Use the **no battery-reversal** command to configure the voice port not to reverse battery when it connects calls. The default is to reverse battery when a call is connected, then return to normal when the call is over, providing positive disconnect. See also the **disconnect-ack** command (Step 7).
Step 2	`Router(config-voiceport)# no supervisory disconnect`	(FXO only) Enables the PBX or PSTN switch to provide STD. By default the **supervisory disconnect** command is enabled.
Step 3	`Router(config-voiceport)# disconnect-ack`	(FXS only) Configures the voice port to return an acknowledgment upon receipt of a disconnect signal. The FXS port removes line power if the equipment on the FXS loop-start trunk disconnects first. This is the default. The **no disconnect-ack** command prevents the FXS port from responding to the on-hook disconnect with a removal of line power.

FXO Supervisory Disconnect Tone Commands

If the FXO supervisory disconnect tone is configured and a detectable tone from the PSTN or PBX is detected by the digital signal processor (DSP), the analog FXO port goes on-hook. This feature prevents an analog FXO port from remaining in an off-hook state after an incoming call is ended. FXO supervisory disconnect tone enables interoperability with PSTN and PBX systems whether or not they transmit supervisory tones.

Note This feature applies only to analog FXO ports with loop-start signaling on the Cisco 2600 and 3600 series routers and on Cisco MC3810 multiservice concentrators with high-performance compression modules (HCMs).

To configure a voice port to detect incoming tones, you need to know the parameters of the tones expected from the PBX or PSTN. Then create a voice class that defines the tone detection parameters, and, finally, apply the voice class to the applicable analog FXO voice ports. This procedure configures the voice port to go on-hook when it detects the specified tones. The parameters of the tones need to be precisely specified to prevent unwanted disconnects due to detection of nonsupervisory tones or noise.

A supervisory disconnect tone is normally a dual tone consisting of two frequencies; however, tones of only one frequency can also be detected. Use caution if you configure voice ports to detect nondual tones, because unwanted disconnects can result from detection of random tone frequencies. You can configure a voice port to detect a tone with one on/off time cycle, or you can configure it to detect tones in a cadence pattern with up to four on/off time cycles.

Note In the following procedure, the following commands were not supported until Cisco IOS Release 12.2(2)T: **freq-max-deviation**, **freq-max-power**, **freq-min-power**, **freq-power-twist**, and **freq-max-delay**.

To create a voice class that defines the specific tone or tones to be detected and then apply the voice class to the voice port, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`voice class dualtone`** *`tag`*	Creates a voice class for defining one tone detection pattern. The range for the tag number is from 1 to 10000. The tag number must be unique on the router. For more information about configuring voice classes, see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide.
Step 2	`Router(config-voice-class)#` **`freq-pair`** *`tone-id frequency-1 frequency-2`*	Specifies the two frequencies, in Hz, for a tone to be detected (or one frequency if a nondual tone is to be detected). If the tone to be detected contains only one frequency, enter 0 for *frequency-2*. The arguments are as follows: • *tone-id*—Ranges from 1 to 16. There is no default. • *frequency-1* and *frequency-2*—Ranges from 300 to 3600, or you can enter 0 for *frequency-2*. There is no default. **Note** Repeat this command for each additional tone to be specified.
Step 3	`Router(config-voice-class)#` **`freq-max-deviation`** *`frequency`*	Specifies the maximum frequency deviation that will be detected, in Hz. The *frequency* argument ranges from 10 to 125. The default is 10.
Step 4	`Router(config-voice-class)#` **`freq-max-power`** *`dBmO`*	Specifies the maximum tone power that will be detected, in dBmO. The *dBmO* argument ranges from 0 to 20. The default is 10.
Step 5	`Router(config-voice-class)#` **`freq-min-power`** *`dBmO`*	Specifies the minimum tone power that will be detected, in dBmO. The *dBmO* argument ranges from 10 to 35. The default is 30.
Step 6	`Router(config-voice-class)#` **`freq-power-twist`** *`dBmO`*	Specifies the power difference allowed between the two frequencies, in dBmO. The *dBmO* argument ranges from 0 to 15. The default is 6.
Step 7	`Router(config-voice-class)#` **`freq-max-delay`** *`time`*	Specifies the timing difference allowed between the two frequencies, in 10-millisecond increments. The *time* argument ranges from 10 to 100 (100 ms to 1 s). The default is 20 (200 ms).
Step 8	`Router(config-voice-class)#` **`cadence-min-on-time`** *`time`*	Specifies the minimum tone on time that will be detected, in 10-millisecond increments. The *time* argument ranges from 0 to 100 (0 ms to 1 s).
Step 9	`Router(config-voice-class)#` **`cadence-max-off-time`** *`time`*	Specifies the maximum tone off time that will be detected, in 10-millisecond increments. The *time* argument ranges from 0 to 5000 (0 ms to 50 s).

	Command	Purpose
Step 10	`Router(config-voice-class)#` **`cadence-list`** *`cadence-id cycle-1-on-time cycle-1-off-time cycle-2-on-time cycle-2-off-time cycle-3-on-time cycle-3-off-time cycle-4-on-time cycle-4-off-time`*	(Optional) Specifies a tone cadence pattern to be detected. Specify an on time and off time for each cycle of the cadence pattern. The arguments are as follows: • *cadence-id*—Ranges from 1 to 10. There is no default. • *cycle-N-on-time* and *cycle-N-off-time*—Range from 0 to 1000 (0 ms to 10 s). The default is 0.
Step 11	`Router(config-voice-class)#` **`cadence-variation`** *`time`*	(Optional) Specifies the maximum time that the tone onset can vary from the specified onset time and still be detected, in 10-millisecond increments. The *time* argument ranges from 0 to 200 (0 ms to 2 s). The default is 0.
Step 12	`Router(config-voice-class)#` **`exit`**	Exits voice class configuration mode.
Step 13	**Cisco 2600 and 3600 Series Routers** `Router(config)#` **`voice-port`** *`slot/subunit/port`* **Cisco MC3810 Multiservice Concentrators** `Router(config)#` **`voice-port`** *`slot/port`*	Enters voice-port configuration mode. The arguments are as follows: • *slot*—Specifies the slot number where the voice network module is installed (Cisco 2600 and Cisco 3600 series routers) or the router slot number where the analog voice module is installed (Cisco MC3810 multiservice concentrators). • *subunit*—Specifies the voice interface card (VIC) where the voice port is located. • *port*—Identifies the analog voice-port number.
Step 14	`Router(config-voiceport)#` **`supervisory disconnect dualtone`** {**`mid-call`** | **`pre-connect`**} **`voice-class`** *`tag`*	Assigns an FXO supervisory disconnect tone voice class to the voice port. The keywords are as follows: • **mid-call**—Specifies tone detection during the entire call. • **pre-connect**—Specifies tone detection only during call set-up.
Step 15	`Router(config-voiceport)#` **`supervisory disconnect anytone`**	Configures the voice port to disconnect on receipt of any tone.

Timeouts Commands

To change timeouts parameters, use the following commands as appropriate, in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)#` **`timeouts call-disconnect`** *`seconds`*	Configures the call disconnect timeout value in seconds. Valid entries range from 0 to 120. The default is 60.
Step 2	`Router(config-voiceport)#` **`timeouts initial`** *`seconds`*	Sets the number of seconds that the system waits between the caller input of the initial digit and the subsequent digit of the dialed string. If the wait time expires before the destination is identified, a tone sounds and the call ends. The *seconds* argument is the initial timeout duration. A valid entry is an integer from 0 to 120. The default is 10.
Step 3	`Router(config-voiceport)#` **`timeouts interdigit`** *`seconds`*	Configures the number of seconds that the system waits after the caller has input the initial digit or a subsequent digit of the dialed string. If the timeout ends before the destination is identified, a tone sounds and the call ends. This value is important when using variable-length dial peer destination patterns (dial plans). The *seconds* argument is the interdigit timeout wait time in seconds. A valid entry is an integer from 0 to 120. The default is 10.
Step 4	`Router(config-voiceport)#` **`timeouts ringing`** `{`*`seconds`* `|` **`infinity`**`}`	Specifies the duration that the voice port allows ringing to continue if a call is not answered. The keyword and argument are as follows: • **infinity**—Indicates ringing should continue until the caller goes on hook. • *seconds*—Specifies the number of seconds to allow ringing without answer. The range is from 5 to 60000. The default is 180.
Step 5	`Router(config-voiceport)#` **`timeouts wait-release`** `{`*`seconds`* `|` **`infinity`**`}`	Specifies the duration that a voice port stays in the call-failure state while the Cisco device sends a busy tone, reorder tone, or an out-of-service tone to the port. The keyword and argument are as follows: • **infinity**—Indicates the voice port should not be released as long as the call-failure state remains. • *seconds*—Specifies the number of seconds to allow before the call is released. The range is from 3 to 3600. The default is 30.

Timing Commands

To change timing parameters, use the following commands as appropriate, in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)#` **`timing clear-wait`** *`milliseconds`*	(E&M only) Specifies the minimum amount of time between the inactive seizure signal and clearing of the call. Valid entries for the *milliseconds* argument are from 200 to 2000 milliseconds. The default is 400.
Step 2	`Router(config-voiceport)#` **`timing delay-duration`** *`milliseconds`*	(E&M only) Specifies the delay signal duration for delay-dial signaling in milliseconds. Valid entries are from 100 to 5000. The default is 2000.
Step 3	`Router(config-voiceport)#` **`timing delay-start`** *`milliseconds`*	(E&M only) Specifies minimum delay time, in milliseconds, from outgoing seizure to outdial address. Valid entries are from 20 to 2000. The default is 300 for the Cisco 3600 series routers, and 150 for the Cisco MC3810 multiservice concentrators.
Step 4	`Router(config-voiceport)#` **`timing delay-with-integrity`** *`milliseconds`*	(Cisco MC3810 multiservice concentrators E&M ports only) Specifies duration of the wink pulse for the delay dial in milliseconds. Valid entries are from 0 to 5000. The default is 0.
Step 5	`Router(config-voiceport)#` **`timing dial-pulse min-delay`** *`milliseconds`*	Specifies time, in milliseconds, between the generation of wink-like pulses when the type is pulse. Valid entries are from 0 to 5000. The default is 300 for the Cisco 3600 series routers, and 140 for the Cisco MC3810 multiservice concentrators.
Step 6	`Router(config-voiceport)#` **`timing dialout-delay`** *`milliseconds`*	(Cisco MC3810 multiservice concentrators only) Specifies dialout delay, in milliseconds, for the sending digit or cut-through on an FXO trunk or an E&M immediate trunk. Valid entries are from 100 to 5000. The default is 300.
Step 7	`Router(config-voiceport)#` **`timing digit`** *`milliseconds`*	Specifies the DTMF digit signal duration in milliseconds. Valid entries are from 50 to 100. The default is 100.
Step 8	`Router(config-voiceport)#` **`timing guard-out`** *`milliseconds`*	(FXO ports only) Specifies the duration in milliseconds of the guard-out period that prevents this port from seizing a remote FXS port before the remote port detects a disconnect signal. The range is from 300 to 3000. The default is 2000.
Step 9	`Router(config-voiceport)#` **`timing hookflash-out`** *`milliseconds`*	Specifies the duration, in milliseconds, of the hookflash. Valid entries are from 50 to 500. The default is 300.

	Command	Purpose
Step 10	`Router(config-voiceport)#` **`timing interdigit`** *`milliseconds`*	Specifies the DTMF interdigit duration, in milliseconds. Valid entries are from 50 to 500. The default is 100.
Step 11	`Router(config-voiceport)#` **`timing percentbreak`** *`percent`*	(Cisco MC3810 multiservice concentrators FXO and E&M ports only) Specifies the percentage of the break period for the dialing pulses, if different from the default. The range is from 20 to 80. The default is 50**.**
Step 12	`Router(config-voiceport)#` **`timing pulse`** *`pulses-per-second`*	(FXO and E&M only) Specifies the pulse dialing rate in pulses per second. Valid entries are from 10 to 20. The default is 20.
Step 13	`Router(config-voiceport)#` **`timing pulse-digit`** *`milliseconds`*	(FXO only) Configures the pulse digit signal duration. The range of the pulse digit signal duration is from 10 to 20. The default is 20.
Step 14	`Router(config-voiceport)#` **`timing pulse-interdigit`**	(FXO and E&M only) Specifies pulse dialing interdigit timing in milliseconds. Valid entries are from 100 to 1000. The default is 500.
Step 15	`Router(config-voiceport)#` **`timing wink-duration`** *`milliseconds`*	(E&M only) Specifies maximum wink-signal duration, in milliseconds, for a wink-start signal. Valid entries are from 100 to 400. The default is 200.
Step 16	`Router(config-voiceport)#` **`timing wink-wait`** *`milliseconds`*	(E&M only) Specifies maximum wink-wait duration, in milliseconds, for a wink-start signal. Valid entries are from 100 to 5000. The default is 200.

DTMF Timer Inter-Digit Command for Cisco AS5300 Access Servers

To configure the DTMF timer for Cisco AS5300 access servers, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`controller T1`** *`number`*	Configures a T1 controller and enters controller configuration mode.
Step 2	`Router(config)#` **`ds0-group`** *`channel-number`* **`timeslots`** *`range`* **`type`** *`signaling-type`* **`dtmf dnis`**	Configures channelized T1 timeslots, which enables a Cisco AS5300 modem to answer and send an analog call.
Step 3	`Router(config)#` **`cas-custom`** *`channel`*	Customizes E1 R2 signaling parameters for a particular E1 channel group on a channelized E1 line.
Step 4	`Router(conf-ctrl-cas)#` **`dtmf-timer-inter-digit`** *`milliseconds`*	Configures the DTMF inter-digit timer for a DS0 group.

Verifying DTMF Timer Inter-Digit Command

To verify the DTMF timer, use the following command in EXEC mode:

Command	Purpose
`Router#` **`show running-config`**	Displays the configuration information currently running on the terminal.

Voice Activity Detection Commands Related to Voice-Port Configuration Mode

In normal voice conversations, only one person speaks at a time. Today's circuit-switched telephone networks dedicate a bidirectional, 64 kbps channel for the duration of each conversation, regardless of whether anyone is speaking at the moment. This means that, in a normal voice conversation, at least 50 percent of the bandwidth is wasted when one or both parties are silent. This figure can actually be much higher when normal pauses and breaks in conversation are taken into account.

Packet-switched voice networks, on the other hand, can use this "wasted" bandwidth for other purposes when voice activity detection (VAD) is configured. VAD works by detecting the magnitude of speech in decibels and deciding when to cut off the voice from being framed. VAD has some technological problems, however, which include the following:

- General difficulties determining when speech ends
- Clipped speech when VAD is slow to detect that speech is beginning again
- Automatic disabling of VAD when conversations take place in noisy surroundings

VAD is configured on dial peers; by default it is enabled. For more information, see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter in this configuration guide. Two parameters associated with VAD, music threshold and comfort noise, are configured on voice ports.

If VAD is enabled, use the following commands to adjust parameter values associated with VAD, beginning in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)#` **`music-threshold`** *`number`*	Specifies the minimal decibel level of music played when calls are put on hold. The decibel level affects how voice activity detection (VAD) treats the music data. Valid entries range from –70 to –30. When used with VAD, if the level is set too high, the remote end hears no music; if it is set too low, there is unnecessary voice traffic. The default is –38.
Step 2	`Router(config-voiceport)#` **`comfort-noise`**	This parameter creates subtle background noise to fill silent gaps during calls when VAD is enabled on voice dial peers. If comfort noise is not generated, the resulting silence can fool the caller into thinking the call is disconnected instead of being merely idle. The default is that comfort noise is enabled.

Voice Quality Tuning Commands

The commands in this section configure parameters to improve voice quality. Common voice quality issues include the following:

- Delay in Voice Networks
- Jitter Adjustment
- Echo Adjustment
- Voice Level Adjustment

Delay in Voice Networks

Delay is the time it takes for voice packets to travel between two endpoints. Excessive delay can cause quality problems with real-time traffic such as voice. However, because of the speed of network links and the processing power of intermediate devices, some delay is expected.

When listening to speech, the human ear normally accepts up to about 150 ms of delay without noticing delays. The ITU G.114 standard recommends no more than 150 ms of one-way delay for a normal voice conversation. Once the delay exceeds 150 ms, a conversation is more like a “walkie-talkie” conversation in which one person must wait for the other to stop speaking before beginning to talk.

You can measure delay fairly easily by using ping tests at various times of the day with different network traffic loads. If network delay is excessive, it must be reduced for adequate voice quality.

Several different types of delay combine to make up the total end-to-end delay associated with voice calls:

- Propagation delay—Amount of time it takes the data to physically travel over the media.
- Handling delay—Amount of time it takes to process data by adding headers, taking samples, forming packets, etc.
- Queuing delay—Amount of time lost due to congestion.
- Variable delay or jitter—Amount of time that causes the conversation to break and become unintelligible. Jitter is described in detail below.

Propagation, handling, and queuing delay are not addressed by voice-port commands and fall outside the scope of this chapter.

Jitter Adjustment

Delay can cause unnatural starting and stopping of conversations, but variable-length delays (also known as jitter) can cause a conversation to break and become unintelligible. Jitter is not usually a problem with PSTN calls because the bandwidth of calls is fixed and each call has a dedicated circuit for the duration of the call. However, in VoIP networks, data traffic might be bursty, and jitter from the packet network can become an issue. Especially during times of network congestion, packets from the same conversation can arrive at different interpacket intervals, disrupting the steady, even delivery needed for voice calls. Cisco voice gateways have built-in jitter buffering to compensate for a certain amount of jitter; the **playout-delay** command can be used to adjust the jitter buffer.

Normally, the defaults in effect are sufficient for most networks. However, a small playout delay from the jitter buffer can cause lost packets and choppy audio, and a large playout delay can cause unacceptably high overall end-to-end delay.

Note Prior to Cisco IOS Release 12.1(5)T, playout delay was configured in voice-port configuration mode. For Cisco IOS Release 12.1(5)T and later releases, in most cases playout delay should be configured in dial-peer configuration mode on the VoIP dial peer that is on the receiving end of the voice traffic that is to be buffered. This dial peer senses network conditions and relays them to the DSPs, which adjust the jitter buffer as necessary. When multiple applications are configured on the gateway, playout delay should be configured in dial-peer configuration mode. When there are numerous dial peers to configure, it might be simpler to configure playout delay on a voice port. If there are conflicting playout delay configurations on a voice port and also on a dial peer, the dial peer configuration takes precedence.

To configure the playout delay jitter buffer, use the following commands beginning in dial-peer or voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)#` **`playout-delay mode`** `{`**`adaptive`** `|` **`fixed`**`}`	Determines the mode in which the jitter buffer will operate for calls on this voice port. The keywords are as follows: • **adaptive**—Adjusts the jitter buffer size and amount of playout delay during a call based on current network conditions. • **fixed**—Defines the jitter buffer size as fixed so that the playout delay does not adjust during a call. A constant playout delay is added. The default is **adaptive**.

	Command	Purpose
Step 2	`Router(config-voiceport)#` **`playout-delay`** `{`**`nominal`** *`value`* `|` **`maximum`** *`value`* `|` **`minimum`** `{`**`default`** `|` **`low`** `|` **`high`**`}}`	Tunes the playout buffer to accommodate packet jitter caused by switches in the WAN. The keywords and arguments are as follows: • **nominal**—Defines the amount of playout delay applied at the beginning of a call by the jitter buffer in the gateway. In fixed mode, this is also the maximum size of the jitter buffer throughout the call. • *value*—Specifies the range that depends on type of DSP and configured codec complexity. For medium codec complexity, the range is from 0 to 150 ms. For high codec complexity and DSPs that do not support codec complexity, the range is from 0 to 250 ms.
		• **maximum** (adaptive mode only)—Specifies the jitter buffer's upper limit (80ms), or the highest value to which the adaptive delay is set. • **minimum** (adaptive mode only)—Specifies the jitter buffer's lower limit (10 ms), or the lowest value to which the adaptive delay is set. • **default**—Specifies 40 ms.

Echo Adjustment

Echo is the sound of your own voice reverberating in the telephone receiver while you are talking. When timed properly, echo is not a problem in the conversation; however, if the echo interval exceeds approximately 25 milliseconds, it is distracting. Echo is controlled by echo cancellers.

In the traditional telephony network, echo is generally caused by an impedance mismatch when the four-wire network is converted to the two-wire local loop. In voice packet-based networks, echo cancellers are built into the low-bit rate codecs and are operated on each DSP.

By design, echo cancellers are limited by the total amount of time they wait for the reflected speech to be received, which is known as an echo trail. The echo trail is normally 32 milliseconds. In Cisco System’s voice implementations, echo cancellers are enabled using the **echo-cancel enable** command, and echo trails are configured using the **echo-cancel coverage** command.

To configure parameters related to the echo canceller, use the following commands beginning in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)#` **`echo-cancel enable`**	Enables the cancellation of voice that is sent and received on the same interface. Echo cancellation coverage must also be configured. The default is that echo cancellation is enabled. **Note** Not valid for four-wire E&M interfaces. Use **no echo-cancel enable** to disable the feature.
Step 2	`Router(config-voiceport)#` **`echo-cancel coverage`** `{`**`8`** `|` **`16`** `|` **`24`** `|` **`32`**`}`	Adjusts the echo canceller by the specified number of milliseconds. The default is 16.
Step 3	`Router(config-voiceport)#` **`non-linear`**	Enables nonlinear processing (residual echo suppression) in the echo canceler, which shuts off any signal if no near-end speech is detected. Echo cancelling must be enabled for this feature. The default is that nonlinear processing is enabled.

Voice Level Adjustment

As much as possible, it is desirable to achieve a uniform input decibel level to the packet voice network in order to limit or eliminate any voice distortion due to incorrect input and output decibel levels. Adjustments to levels may be required by the type of equipment connected to the network or by local country-specific conditions.

Incorrect input or output levels can cause echo, as can an impedance mismatch. Too much input gain can cause clipped or fuzzy voice quality. If the output level is too high at the remote router's voice port, the local caller will hear echo. If the local router's voice port input decibel level is too high, the remote side will hear clipping. If the local router's voice port input decibel level is too low, or the remote router's output level is too low, the remote side voice can be distorted at a very low volume and DTMF may be missed.

Use the **input gain** and **output attenuation** commands to adjust voice levels, and the **impedance** command to set the impedance value to match that of the voice circuit to which the voice port connects.

To change parameters related to voice levels, use the following commands as appropriate, in voice-port configuration mode:

	Command	Purpose
Step 1	`Router(config-voiceport)# input gain value`	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface, increasing or decreasing the signal. After an input gain setting is changed, the voice call must be disconnected and reestablished before the changes take effect. The *value* argument is any integer from –6 to 14. The default is 0.
Step 2	`Router(config-voiceport)# output attenuation value`	Specifies the amount of attenuation in decibels at the transmit side of the interface, decreasing the signal. A system-wide loss plan can be implemented using the **input gain** and **output attenuation** commands. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that normally there must be –6 dB attenuation between phones. The *value* argument is any integer from –6 to 14. The default is 0.
Step 3	`Router(config-voiceport)# impedance {600c | 600r | 900c | complex1 | complex2}`	Specifies the terminating impedance of a voice port interface, which needs to match the specifications from the specific telephony system to which it is connected. • **600c**—Specifies 600 ohms complex. • **600r**—Specifies 600 ohms real. • **900c**—Specifies 900 ohms complex. • **complex1**—Specifies Complex 1. • **complex2**—Specifies Complex 2. The default is 600r.

	Command	Purpose
Step 4	`Router(config-voiceport)#` **`loss-plan`** `{`**`plan1`** `|` **`plan2`** `|` **`plan5`** `|` **`plan6`** `|` **`plan7`** `|` **`plan8`** `|` **`plan9`**`}`	(Cisco MC3810 multiservice concentrators FXO or FXS analog voice ports only) Specifies the analog-to-digital gain offset loss plan. For definitions of each plan, see the *Cisco IOS Voice, Video, and Fax Command Reference*. The default is the **plan1** keyword.
Step 5	`Router(config-voiceport)#` **`idle-voltage`** `{`**`high`** `|` **`low`**`}`	(Cisco MC3810 multiservice concentrators analog FXS ports only) Specifies the talk-battery (tip-to-ring) voltage condition when the port is idle. The keywords are as follows: • **high**—Specifies that the voltage is high (–48V). • **low**—Specifies that the voltage is low (–24V) and is the default.

Verifying Analog and Digital Voice-Port Configurations

After configuring the voice ports on your router, perform the following steps to verify proper operation:

Step 1 Pick up the handset of an attached telephony device and check for a dial tone.

Step 2 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.

Step 3 To identify port numbers of voice interfaces installed in your router, use the **show voice port summary** command. For examples of the output, see the "show voice port summary Command Examples" section on page 97.

Step 4 To verify voice-port parameter settings, use the **show voice port** command with the appropriate syntax from Table 9. For sample output, see the "show voice port Command Examples" section on page 98.

Table 9 Show Voice Port Command Syntax

Platform	Voice Port Type	Command Syntax
Cisco 1750	Analog	**show voice port** [*slot/port* | **summary**]
Cisco 2600 series Cisco 3600 series	Analog	**show voice port** [*slot/port* | **summary**]
	Digital	**show voice port** [*slot/port:ds0-group-no* | **summary**]
Cisco MC3810	Analog	**show voice port** [*slot/port* | **summary**]
	Digital	**show voice port** [*slot:ds0-group-no* | **summary**]
Cisco AS5300	Digital	**show voice-port** *controller:ds0-group-no*
Cisco AS5800	Digital	**show voice-port** {*shelf/slot/port:ds0-group-no*}
Cisco 7200 series	Digital	**show voice port** {*slot/port-adapter:ds0-group-no*}
Cisco 7500 series	Digital	**show voice port** {*slot/port-adapter/slot:ds0-group-no*}

Step 5 For digital T1/E1 connections, to verify the codec complexity configuration, use the **show running-config** command to display the current voice-card setting. If medium complexity is specified, the codec complexity setting is not displayed. If high complexity is specified, the setting codec complexity high is displayed. The following example shows an excerpt from the command output when high complexity has been specified:

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

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

Step 6 For digital T1/E1 connections, to verify that the controller is up and that no alarms have been reported, and to display information about clock sources and other controller settings, use the **show controller** command. For output examples, see the "show controller Command Examples" section on page 102.

```
Router# show controller {t1 | e1} controller-number
```

Step 7 To display voice-channel configuration information for all DSP channels, use the **show voice dsp** command. For output examples, see the "show voice dsp Command Examples" section on page 103.

```
Router# show voice dsp
```

Step 8 To verify the call status for all voice ports, use the **show voice call summary** command. For output examples, see the "show voice call summary Command Examples" section on page 104.

```
Router# show voice call summary
```

Step 9 To display the contents of the active call table, which shows all of the calls currently connected through the router or concentrator, use the **show call active voice** command. For output examples, see the "show call active voice Command Example" section on page 104.

```
Router# show call active voice
```

Step 10 To display the contents of the call history table, use the **show call history voice** command. To limit the display to the last calls connected through this router, use the keyword **last** and define the number of calls to be displayed with the argument *number*. To limit the display to a shortened version of the call history table, use the **brief** keyword. For output examples, see the "show call history voice Command Example" section on page 105.

```
Router# show call history voice [last | number | brief]
```

show voice port summary Command Examples

In the following sections, output examples of the following types are shown:

- Cisco 3640 Router Analog Voice Port
- Cisco MC3810 Multiservice Concentrator Digital Voice Port

Cisco 3640 Router Analog Voice Port

The following output is from a Cisco 3640 router:

```
Router# show voice port summary
                                     IN       OUT
PORT   CH SIG-TYPE  ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
2/0/0  -- e&m-wnk   up    dorm idle     idle     y
2/0/1  -- e&m-wnk   up    dorm idle     idle     y
2/1/0  -- fxs-ls    up    dorm on-hook  idle     y
2/1/1  -- fxs-ls    up    dorm on-hook  idle     y
```

Cisco MC3810 Multiservice Concentrator Digital Voice Port

The following output is from a Cisco MC3810 multiservice concentrator:

```
Router# show voice port summary
                                     IN       OUT
PORT   CH SIG-TYPE  ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
0:17   18 fxo-ls    down  down idle     on-hook  y
0:18   19 fxo-ls    up    dorm idle     on-hook  y
0:19   20 fxo-ls    up    dorm idle     on-hook  y
0:20   21 fxo-ls    up    dorm idle     on-hook  y
0:21   22 fxo-ls    up    dorm idle     on-hook  y
0:22   23 fxo-ls    up    dorm idle     on-hook  y
0:23   24 e&m-imd   up    dorm idle     idle     y
1/1    -- fxs-ls    up    dorm on-hook  idle     y
1/2    -- fxs-ls    up    dorm on-hook  idle     y
1/3    -- e&m-imd   up    dorm idle     idle     y
1/4    -- e&m-imd   up    dorm idle     idle     y
1/5    -- fxo-ls    up    dorm idle     on-hook  y
1/6    -- fxo-ls    up    dorm idle     on-hook  y
```

show voice port Command Examples

In the following sections, output examples of the following types are shown:

- Cisco 3600 Series Router Analog E&M Voice Port, page 98
- Cisco 3600 Series Router Analog FXS Voice Port, page 99
- Cisco 3600 Series Router Digital E&M Voice Port, page 100
- Cisco AS5300 Universal Access Server T1 CAS Voice Port, page 100
- Cisco 7200 Series Router Digital E&M Voice Port, page 101

Cisco 3600 Series Router Analog E&M Voice Port

The following output is from a Cisco 3600 series router analog E&M voice port:

```
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
```

Cisco 3600 Series Router Analog FXS Voice Port

The following output is from a Cisco 3600 series router analog FXS voice port:

```
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
```

Cisco 3600 Series Router Digital E&M Voice Port

The following output is from a Cisco 3600 series router digital E&M voice port:

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

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

Cisco AS5300 Universal Access Server T1 CAS Voice Port

The following output is from a Cisco AS5300 universal access server T1 CAS voice port:

```
Router# show voice port

DS0 Group 1:0 - 1:0
 Type of VoicePort is CAS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
```

```
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to default
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 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
 Call-Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Companding Type is u-law
 Region Tone is set for US
 Wait Release Time Out is 30 s
 Station name None, Station number None

 Voice card specific Info Follows:

 DS0 channel specific status info:
                               IN       OUT
    PORT   CH SIG-TYPE   OPER STATUS   STATUS    TIP      RING
```

Cisco 7200 Series Router Digital E&M Voice Port

The following output is from a Cisco 7200 series router digital E&M voice port:

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

receEive and transMit Slot is 1, Sub-unit is 0, Port is 1  << voice-port 1/0:1

 Type of VoicePort is E&M

 Operation State is DORMANT

 Administrative State is UP

 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB

 Out Attenuation is Set to 0 dB

 Echo Cancellation is enabled

 Echo Cancel Coverage is set to 8 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s

 Interdigit Time Out is set to 10 s

 Region Tone is set for US
```

show controller Command Examples

In the following sections, output examples of the following types are shown:

- Cisco 3600 Series Router T1 Controller, page 102
- Cisco MC3810 Multiservice Concentrator E1 Controller, page 102
- Cisco AS5800 Universal Access Server T1 Controller, page 102

Cisco 3600 Series Router T1 Controller

The following output is from a Cisco 3600 series router with a T1 controller:

```
Router# show controller T1 1/1/0

T1 1/0/0 is up.
  Applique type is Channelized T1
  Cablelength is long gain36 0db
  No alarms detected.
  alarm-trigger is not set
  Framing is ESF, Line Code is B8ZS, Clock Source is Line.
  Data in current interval (180 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

Cisco MC3810 Multiservice Concentrator E1 Controller

The following output is from a Cisco MC3810 multiservice concentrator with an E1 controller:

```
Router# show controller E1 1/0

E1 1/0 is up.
  Applique type is Channelized E1
  Cablelength is short 133
  Description: E1 WIC card Alpha
  No alarms detected.
  Framing is CRC4, Line Code is HDB3, Clock Source is Line Primary.
  Data in current interval (1 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

Cisco AS5800 Universal Access Server T1 Controller

The following output is from a Cisco AS5800 universal access server with a T1 controller:

```
Router# show controller t1 2

 T1 2 is up.
   No alarms detected.
   Version info of slot 0:  HW: 2, Firmware: 16, PLD Rev: 0

 Manufacture Cookie Info:
  EEPROM Type 0x0001, EEPROM Version 0x01, Board ID 0x42,
  Board Hardware Version 1.0, Item Number 73-2217-4,
  Board Revision A0, Serial Number 06467665,
  PLD/ISP Version 0.0, Manufacture Date 14-Nov-1997.

   Framing is ESF, Line Code is B8ZS, Clock Source is Internal.
```

```
Data in current interval (269 seconds elapsed):
 0 Line Code Violations, 0 Path Code Violations
   0 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
   0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 0 Unavail Secs
```

show voice dsp Command Examples

The following output is from a Cisco 3640 router when a digital voice port is configured:

```
Router# show voice dsp

TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 010 00 g729r8    3.3 busy  idle      0  0 1/015    1     0     67400/85384
         01 g729r8     .8 busy  idle      0  0 1/015    7     0     67566/83623
         02 g729r8        busy  idle      0  0 1/015   13     0     65675/81851
         03 g729r8        busy  idle      0  0 1/015   20     0     65530/83610
C549 011 00 g729r8    3.3 busy  idle      0  0 1/015    2     0     66820/84799
         01 g729r8     .8 busy  idle      0  0 1/015    8     0     59028/66946
         02 g729r8        busy  idle      0  0 1/015   14     0     65591/81084
         03 g729r8        busy  idle      0  0 1/015   21     0     66336/82739
C549 012 00 g729r8    3.3 busy  idle      0  0 1/015    3     0     59036/65245
         01 g729r8     .8 busy  idle      0  0 1/015    9     0     65826/81950
         02 g729r8        busy  idle      0  0 1/015   15     0     65606/80733
         03 g729r8        busy  idle      0  0 1/015   22     0     65577/83532
C549 013 00 g729r8    3.3 busy  idle      0  0 1/015    4     0     67655/82974
         01 g729r8     .8 busy  idle      0  0 1/015   10     0     65647/82088
         02 g729r8        busy  idle      0  0 1/015   17     0     66366/80894
         03 g729r8        busy  idle      0  0 1/015   23     0     66339/82628
C549 014 00 g729r8    3.3 busy  idle      0  0 1/015    5     0     68439/84677
         01 g729r8     .8 busy  idle      0  0 1/015   11     0     65664/81737
         02 g729r8        busy  idle      0  0 1/015   18     0     65607/81820
         03 g729r8        busy  idle      0  0 1/015   24     0     65589/83889
C549 015 00 g729r8    3.3 busy  idle      0  0 1/015    6     0     66889/83331
         01 g729r8     .8 busy  idle      0  0 1/015   12     0     65690/81700
         02 g729r8        busy  idle      0  0 1/015   19     0     66422/82099
         03 g729r8        busy  idle      0  0 1/015   25     0     65566/83852

Router# show voice dsp

TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS ABORT   TX/RX-PAK-CNT
==== === == ======== ==== ===== ======= === == ======= == ===== ===============
C549 007 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    4     0             0/0
                      .13
C549 008 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    5     0             0/0
                      .13
C549 009 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    6     0             0/0
                      .13
C549 010 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    7     0             0/0
                      .13
C549 011 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    8     0             0/0
                      .13
C549 012 00 {medium}  3.3 IDLE  idle      0  0 1/0:1    9     0             0/0
                      .13
C542 001 01 g711ulaw  3.3 IDLE  idle      0  0 2/0/0          0         512/519
                      .13
C542 002 01 g711ulaw  3.3 IDLE  idle      0  0 2/0/1          0         505/502
                      .13
C542 003 01 g711alaw  3.3 IDLE  idle      0  0 2/1/0          0     28756/28966
                      .13
C542 004 01 g711ulaw  3.3 IDLE  idle      0  0 2/1/1          0         834/838
                      .13
```

show voice call summary Command Examples

In the following sections, output examples of the following types are shown:

- Cisco MC3810 Multiservice Concentrator Analog Voice Port
- Cisco 3600 Series Router Digital Voice Port

Cisco MC3810 Multiservice Concentrator Analog Voice Port

The following output is from a Cisco MC3810 multiservice concentrator:

```
Router# show voice call summary

PORT      CODEC    VAD VTSP STATE            VPM STATE
========= ======== === ==================== ========================
1/1       g729r8    y  S_CONNECT            FXSLS_CONNECT
1/2       -         -  -                    FXSLS_ONHOOK
1/3       -         -  -                    EM_ONHOOK
1/4       -         -  -                    EM_ONHOOK
1/5       -         -  -                    FXOLS_ONHOOK
1/6       -         -  -                    FXOLS_ONHOOK
```

Cisco 3600 Series Router Digital Voice Port

The following output is from a Cisco 3600 series router:

```
Router# show voice call summary

PORT      CODEC    VAD VTSP STATE            VPM STATE
========= ======== === ==================== ========================
1/015.1  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.2  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.3  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.4  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.5  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.6  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.7  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.8  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.9  g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.10 g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.11 g729r8    y  S_CONNECT            S_TSP_CONNECT
1/015.12 g729r8    y  S_CONNECT            S_TSP_CONNECT
```

show call active voice Command Example

The following output is from a Cisco 7200 series router:

```
Router# show call active voice

GENERIC:
SetupTime=94523746 ms
Index=448
PeerAddress=##73072

PeerSubAddress=
PeerId=70000

PeerIfIndex=37
```

```
LogicalIfIndex=0
ConnectTime=94524043
DisconectTime=94546241
CallOrigin=1

ChargedUnits=0
InfoType=2
TransmitPackets=6251
TransmitBytes=125020
ReceivePackets=3300
ReceiveBytes=66000
VOIP:
ConnectionId[0x142E62FB 0x5C6705AF 0x0 0x385722B0]
RemoteIPAddress=171.68.235.18

RemoteUDPPort=16580

RoundTripDelay=29 ms

SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPlayout=63690
GapFillWithSilence=0 ms

GapFillWithPrediction=180 ms

GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=40 ms
LostPackets=0 ms

EarlyPackets=1 ms

LatePackets=18 ms

VAD = disabled

CoderTypeRate=g729r8

CodecBytes=20

cvVoIPCallHistoryIcpif=0
SignalingType=cas
```

show call history voice Command Example

The following output is from a Cisco 7200 series router:

```
Router# show call history voice

GENERIC:
SetupTime=94893250 ms
Index=450
PeerAddress=##52258

PeerSubAddress=
PeerId=50000
```

```
PeerIfIndex=35
LogicalIfIndex=0
DisconnectCause=10

DisconnectText=normal call clearing.

ConnectTime=94893780
DisconectTime=95015500
CallOrigin=1

ChargedUnits=0
InfoType=2
TransmitPackets=32258
TransmitBytes=645160
ReceivePackets=20061
ReceiveBytes=401220
VOIP:
ConnectionId[0x142E62FB 0x5C6705B3 0x0 0x388F851C]
RemoteIPAddress=171.68.235.18

RemoteUDPPort=16552

RoundTripDelay=23 ms

SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPlayout=398000
GapFillWithSilence=0 ms

GapFillWithPrediction=1440 ms

GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=97 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=49 ms
LostPackets=1 ms
EarlyPackets=1 ms

LatePackets=132 ms

VAD = disabled

CoderTypeRate=g729r8

CodecBytes=20
cvVoIPCallHistoryIcpif=0
```

Troubleshooting Analog and Digital Voice Port Configurations

The following sections will assist in analyzing and troubleshooting voice port problems:

- Troubleshooting Chart, page 107
- Voice Port Testing Commands, page 109

Troubleshooting Chart

Table 10 lists some problems you might encounter after configuring voice ports and has some suggested remedies.

Table 10 Troubleshooting Voice Port Configurations

Problem	Suggested Action
No connectivity	Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS IP Configuration Guide*.
No connectivity	Enter the **show controller t1** or **show controller e1** command with the controller number for the voice port you are troubleshooting. This will tell you: • If the controller is up. If it is not, use the **no shutdown** command to make it active. • Whether alarms have been reported. • What parameter values have been set for the controller (framing, clock source, line code, cable length). If these values do not match those of the telephony connection you are making, reconfigure the controller. See the "show controller Command Examples" section on page 102 for output.
No connectivity	Enter the **show voice port** command with the voice port number that you are troubleshooting, which will tell you: • If the voice port is up. If it is not, use the **no shutdown** command to make it active. • What parameter values have been set for the voice port, including default values (these do not appear in the output for the **show running-config** command). If these values do not match those of the telephony connection you are making, reconfigure the voice port. See the "show voice port Command Examples" section on page 98 for sample output.
Telephony device buzzes or does not ring	Use the **show voice port** command to confirm that ring frequency is configured correctly. It must match the connected telephony equipment and may be country-dependent.

Table 10 Troubleshooting Voice Port Configurations (continued)

Problem	Suggested Action
Distorted speech	Use the **show voice port** command to confirm the **cptone** keyword setting (also called *region tone*) is US. Setting a wrong cptone could result in faulty voice reproduction during analog-to-digital or digital-to-analog conversions.
Music on hold is not heard	Reduce the music-threshold level.
Background noise is not heard	Enable the **comfort-noise** command.
Long pauses occur in conversation; like speaking on a walkie-talkie	Overall delay is probably excessive; the standard for adequate voice quality is 150 ms one-way transit delay. Measure delay by using ping tests at various times of the day with different network traffic loads. If delay must be reduced, areas to examine include propagation delay of signals between the sending and receiving endpoints, voice encoding delay, and the voice packetization time for various VoIP codecs.
Jerky or choppy speech	Variable delay, or jitter, is being introduced by congestion in the packet network. Two possible remedies are to: • Reduce the amount of congestion in your packet network. Pings between VoIP endpoints will give an idea of the round-trip delay of a link, which should never exceed 300 ms. Network queuing and dropped packets should also be examined. • Increase the size of the jitter buffer with the **playout-delay** command. (See the "Jitter Adjustment" section on page 91.)
Clipped or fuzzy speech	Reduce input gain. (See the "Voice Level Adjustment" section on page 95.)
Clipped speech	Reduce the input level at the listener's router. (See the "Voice Level Adjustment" section on page 95.)
Volume too low or missed DTMF	Increase speaker's output level or listener's input level. (See the "Voice Level Adjustment" section on page 95.)
Echo interval is greater than 25 ms (sounds like a separate voice)	Configure the **echo-cancel enable** command and increase the value for the **echo-cancel coverage** keyword. (See the "Echo Adjustment" section on page 93.)
Too much echo	Reduce the output level at the speaker's voice port. (See the "Voice Level Adjustment" section on page 95.)

Voice Port Testing Commands

These commands allow you to force voice ports into specific states for testing. They require the use of Cisco IOS Release 12.0(7)XK or 12.1(2)T or a later release, and they apply only to Cisco 2600 and 3600 series routers, and to Cisco MC3810 multiservice concentrators. The following types of voice-port tests are covered:

Detector-Related Function Tests

Using the **test voice port detector** command, you are able to force a particular detector into an on or off state, perform tests on the detector, and then return the detector to its original state.

To configure this feature, use the following commands in privileged EXEC mode:

	Command	Purpose
Step 1	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** `Router# test voice port slot/subunit/port detector {m-lead	battery-reversal	loop-current	ring	tip-ground	ring-ground	ring-trip} {on	off}` **Cisco 2600 and 3600 Series Routers Digital Voice Ports** `Router# test voice port slot/port:ds0-group detector {m-lead	battery-reversal	loop-current	ring	tip-ground	ring-ground	ring-trip} {on	off}` **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** `Router# test voice port slot/port detector {m-lead	battery-reversal	loop-current	ring	tip-ground	ring-ground	ring-trip} {on	off}`	Identifies the voice port you want to test. Enter a keyword for the detector under test and specify whether to force it to the on or off state. **Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a detector is in the forced state.

	Command	Purpose
	Cisco MC3810 Multiservice Concentrators Digital Voice Ports `Router# test voice port slot:ds0-group detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} {on | off}`	

Step 2	**Cisco 2600 and 3600 Series Routers Analog Voice Ports**
	`Router# test voice port slot/subunit/port detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} disable`
	
	Cisco 2600 and 3600 Series Routers Digital Voice Ports
	`Router# test voice port slot/port:ds0-group detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} disable`
	
	Cisco MC3810 Multiservice Concentrators Analog Voice Ports
	`Router# test voice port slot/port detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} disable`
	
	Cisco MC3810 Multiservice Concentrators Digital Voice Ports
	`Router# test voice port slot:ds0-group detector {m-lead | battery-reversal | loop-current | ring | tip-ground | ring-ground | ring-trip} disable`
		Identifies the voice port on which you want to end the test. Enter a keyword for the detector under test and the keyword **disable** to end the forced state.
		
		Note For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a detector is in the forced state.

Loopback Function Tests

To establish loopbacks on a voice port, use the following commands in privileged EXEC mode:

	Command	Purpose
Step 1	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** `Router# test voice port slot/subunit/port loopback {local	network}` **Cisco 2600 and 3600 Series Routers Digital Voice Ports** `Router# test voice port slot/port:ds0-group loopback {local	network}` **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** `Router# test voice port slot/port detector loopback {local	network}` **Cisco MC3810 Multiservice Concentrators Digital Voice Ports** `Router# test voice port slot:ds0-group loopback {local	network}`	Identifies the voice port you want to test and enters a keyword for the loopback direction. **Note** A call must be established on the voice port under test.

Step 2	**Cisco 2600 and 3600 Series Routers Analog Voice Ports**
	`Router# test voice port slot/subunit/port loopback disable`
	Cisco 2600 and 3600 Series Routers Digital Voice Ports
	`Router# test voice port slot/port:ds0-group loopback disable`
	Cisco MC3810 Multiservice Concentrators Analog Voice Ports
	`Router# test voice port slot/port detector loopback disable`
	Cisco MC3810 Multiservice Concentrators Digital Voice Ports
	`Router# test voice port slot:ds0-group loopback disable`
		Identifies the voice port on which you want to end the test and enters the keyword disable to end the loopback.

Tone Injection Tests

To inject a test tone into a voice port, use the following commands in privileged EXEC mode:

	Command	Purpose
Step 1	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** `Router#` **`test voice port`** *`slot/subunit/port`* **`inject-tone`** {**`local`** | **`network`**} {**`1000hz`** | **`2000hz`** | **`200hz`** | **`3000hz`** | **`300hz`** | **`3200hz`** | **`3400hz`** | **`500hz`** | **`quiet`**} **Cisco 2600 and 3600 Series Routers Digital Voice Ports** `Router#` **`test voice port`** *`slot/port:ds0-group`* **`inject-tone`** {**`local`** | **`network`**} {**`1000hz`** | **`2000hz`** | **`200hz`** | **`3000hz`** | **`300hz`** | **`3200hz`** | **`3400hz`** | **`500hz`** | **`quiet`**} **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** `Router#` **`test voice port`** *`slot/port detector`* **`inject-tone`** {**`local`** | **`network`**} {**`1000hz`** | **`2000hz`** | **`200hz`** | **`3000hz`** | **`300hz`** | **`3200hz`** | **`3400hz`** | **`500hz`** | **`quiet`**} **Cisco MC3810 Multiservice Concentrators Digital Voice Ports** `Router#` **`test voice port`** *`slot:ds0-group`* **`inject-tone`** {**`local`** | **`network`**} {**`1000hz`** | **`2000hz`** | **`200hz`** | **`3000hz`** | **`300hz`** | **`3200hz`** | **`3400hz`** | **`500hz`** | **`quiet`**}	Identifies the voice port you want to test and enter keywords for the direction to send the test tone and for the frequency of the test tone. **Note** A call must be established on the voice port under test.
Step 2	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** `Router#` **`test voice port`** *`slot/subunit/port`* **`inject-tone disable`** **Cisco 2600 and 3600 Series Routers Digital Voice Ports** `Router#` **`test voice port`** *`slot/port:ds0-group`* **`inject-tone disable`** **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** `Router#` **`test voice port`** *`slot/port detector`* **`inject-tone disable`** **Cisco MC3810 Multiservice Concentrators Digital Voice Ports** `Router#` **`test voice port`** *`slot:ds0-group`* **`inject-tone disable`**	Identifies the voice port on which you want to end the test and enter the keyword **disable** to end the test tone. **Note** The **disable** keyword is available only if a test condition is already activated.

Relay-Related Function Tests

To test relay-related functions on a voice port, use the following commands in privileged EXEC mode:

	Command	Purpose
Step 1	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** Router# **test voice port** *slot/subunit/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off**} **Cisco 2600 and 3600 Series Routers Digital Voice Ports** Router# **test voice port** *slot/port:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off**} **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** Router# **test voice port** *slot/port detector* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off**} **Cisco MC3810 Multiservice Concentrators Digital Voice Ports** Router# **test voice port** *slot:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} {**on** | **off**}	Identifies the voice port you want to test. Enter a keyword for the relay under test and specify whether to force it to the on or off state. **Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a relay is in the forced state.
Step 2	**Cisco 2600 and 3600 Series Routers Analog Voice Ports** Router# **test voice port** *slot/subunit/port* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} **disable** **Cisco 2600 and 3600 Series Routers Digital Voice Ports** Router# **test voice port** *slot/port:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} **disable** **Cisco MC3810 Multiservice Concentrators Analog Voice Ports** Router# **test voice port** *slot/port detector* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} **disable** **Cisco MC3810 Multiservice Concentrators Digital Voice Ports** Router# **test voice port** *slot:ds0-group* **relay** {**e-lead** | **loop** | **ring-ground** | **battery-reversal** | **power-denial** | **ring** | **tip-ground**} **disable**	Identifies the voice port on which you want to end the test. Enter a keyword for the relay under test, and the keyword **disable** to end the forced state. **Note** For each signaling type (E&M, FXO, FXS), only the applicable keywords are displayed. The **disable** keyword is displayed only when a relay is in the forced state.

Fax/Voice Mode Tests

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

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

To force a voice port into fax mode and return it to voice mode, use the following commands in privileged EXEC mode:

	Command	Purpose

Step 1	**Cisco 2600 and 3600 Series Routers Analog Voice Ports**
	`Router# test voice port slot/subunit/port switch fax`
	Cisco 2600 and 3600 Series Routers Digital Voice Ports
	`Router# test voice port slot/port:ds0-group switch fax`
	Cisco MC3810 Multiservice Concentrators Analog Voice Ports
	`Router# test voice port slot/port detector switch fax`
	Cisco MC3810 Multiservice Concentrators Digital Voice Ports
	`Router# test voice port slot:ds0-group switch fax`
		Identifies the voice port you want to test. Enter the keyword **fax** to force the voice port into fax mode.

Step 2	**Cisco 2600 and 3600 Series Routers Analog Voice Ports**
	`Router# test voice port slot/subunit/port switch disable`
	Cisco 2600 and 3600 Series Routers Digital Voice Ports
	`Router# test voice port slot/port:ds0-group switch disable`
	Cisco MC3810 Multiservice Concentrators Analog Voice Ports
	`Router# test voice port slot/port detector switch disable`
	Cisco MC3810 Multiservice Concentrators Digital Voice Ports
	`Router# test voice port slot:ds0-group switch disable`
		Identifies the voice port on which you want to end the test. Enter the keyword **disable** to return the voice port to voice mode.