Voice Port Configuration Guide

Cisco IOS XE Release 3S

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Cisco IOS Voice Port Feature Roadmap

This roadmap provides information about Cisco IOS voice features involving voice port configuration. It contains the following:

- Platforms and Cisco IOS Software Images, page 1
- Cisco IOS Voice Port Feature List, page 1

Note This chapter describes how to access Cisco Feature Navigator. It also lists and describes, by Cisco IOS release, voice port features for that release.

For information about the full set of Cisco IOS voice features, see the entire Cisco IOS Voice Configuration Library—including library preface, glossary, and other documents—at http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.html.

Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Cisco IOS Voice Port Feature List

Table 1 lists voice port features by Cisco IOS release. Features that are introduced in a particular release are available in that and subsequent releases.

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Table 1 Voice Port Features by Cisco IOS Release

Release	Feature Name	Feature Description	Where Documented
12.4(20)T	Software-Configurable 128-ms Echo Cancellation	Beginning in Release 12.4(20)T, software-configurable echo cancellation coverage was expanded to include 80, 96, 112, and 128 milliseconds. The default is changed from 64 ms to 128 ms.	Configuring Echo Cancellation and Configuring Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards and NextPort-Based Voice Tuning and Echo Cancellation
12.4(15)XZ	Enhancements to the Cisco 880 Series Routers	Beginning in Release 12.4(15)XZ, the Cisco 880 broadband series routers integrate access technologies such as the VDSL2, 3G; adds voice for service provider-managed services and for enterprise teleworkers. In addition, the broadband series routers offer UTM features such as URL filtering and optional IEEE 802.11n integrated access points that can be managed using LWAPP.	Configuring Analog Voice Ports and Configuring Digital Voice Ports
12.3(14)T	Hardware Echo Cancellation	Two echo cancellation modules (EC-MFT-32 and EC-MFT-64) provide hardware echo cancellation with a hard-coded tail length of 128 milliseconds for VWIC2s.	Configuring Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards
12.3(11)T	NextPort-Based Voice-Tuning and Dual-Filter G.168 Echo Canceller	NextPort dual-filter G.168 echo canceller (EC) improves voice quality in VoIP connections by providing relatively less residual echo leakage, better nonlinear processing (NLP) timing, less clipping, and better comfort noise generation (CNG) in most environments.	NextPort-Based Voice Tuning and Echo Cancellation

Table 1 Voice Port Features by Cisco IOS Release (continued)

Release	Feature Name	Feature Description	Where Documented
12.3(7)T	IP Communications High-Density Digital Voice/Fax Network Module	This feature supports high-density digital voice and low-density analog voice connectivity along with data and integrated access connectivity. The network modules offer built-in T1/E1 ports, and include a single VIC/voice WAN interface card (VWIC) slot for FXS, FXO, E&M, software-configured CAMA, DID, BRI, or E1 and T1 cards, up to a maximum of four T1/E1 ports. The network modules also support up to 32 HDLC channels with an aggregate capacity of 2.048 Mbps.	Configuring Digital Voice Ports
12.2(13)T	Enhanced ITU-T G.168 Echo Cancellation	This feature offers improved standards for echo cancellation performance. Configurable tail length is increased, up to 64 ms. Minimum ERL is configurable to greater than or equal to 0 dB, 3 dB, or 6 dB. Echo suppression is no longer needed because of faster convergence.	Configuring Echo Cancellation

Voice Port Configuration Overview

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 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. In addition to the commands for basic configuration, there are also commands that provide fine-tuning for voice quality, enable special features, and specify parameters to match those of proprietary PBXs.

This document includes the following chapters:

- Configuring Analog Voice Ports
- Configuring Digital Voice Ports
- Fine-Tuning Analog and Digital Voice Ports
- Configuring Echo Cancellation
- Configuring Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards
- NextPort-Based Voice Tuning and Echo Cancellation
- Verifying Analog and Digital Voice-Port Configurations
- Troubleshooting Analog and Digital Voice Port Configurations

Not all voice-port commands are covered in this document. Some are described in the *Cisco IOS ISDN Voice Configuration Guide*, Release 12.4 or the *Trunk Management Features* document, Cisco IOS Voice Configuration Library, Release 12.4. The voice-port configuration commands included in this document are fully documented in the *Cisco IOS Voice Command Reference*.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Voice Port Configuration Overview

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 following illustrations show examples of how voice ports are used.

- Figure 1 shows one voice port connecting a telephone to the WAN through the router.
- Figure 2 shows one voice port connected to the PSTN and another to a telephone; the router acts like a small PBX.
- Figure 3 shows how two PBXs can be connected over a WAN to provide toll bypass.

Figure 1 Telephone to WAN

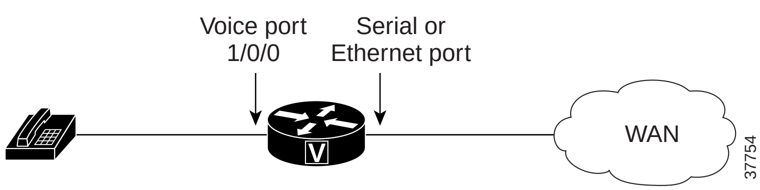


Figure 2 Telephone to PSTN

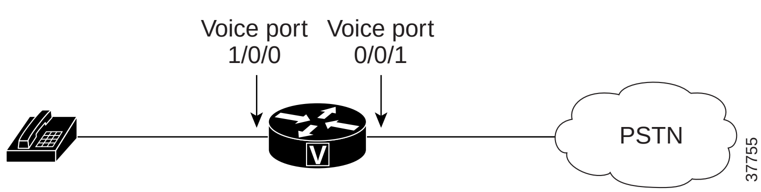


Figure 3 PBX-to-PBX over a WAN

Cisco provides a variety of Cisco IOS commands for flexibility in configuring 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 1 shows the analog and digital voice-port connection support of the router platforms discussed in this document.

Table 1 Analog and Digital Voice-Port Support on Cisco Platforms

Platform	Analog	Digital
Cisco 880 series (includes IAD881B, IAD881F, C881SRST, IAD888B, IAD888F, and C888SRST)	Yes	Yes
Cisco 1750	Yes	No
Cisco 2600 series	Yes	Yes
Cisco 3600 series	Yes	Yes
Cisco 3700 series	Yea	Yes
Cisco 7200 series	No	Yes
Cisco 7500 series	No	Yes
Cisco AS5300	No	Yes
Cisco AS5350	No	Yes
Cisco AS5400	No	Yes
Cisco AS5800	No	Yes
Cisco AS5850	No	Yes
Cisco MC3810	Yes	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 document 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 “FXS and FXO Interfaces” section on page 4 and the “E&M Interfaces” section on page 5. 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 4, FXS signaling is used for end-user telephony equipment, such as a telephone or fax machine. Figure 5 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 6, 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 4 FXS Signaling Interfaces

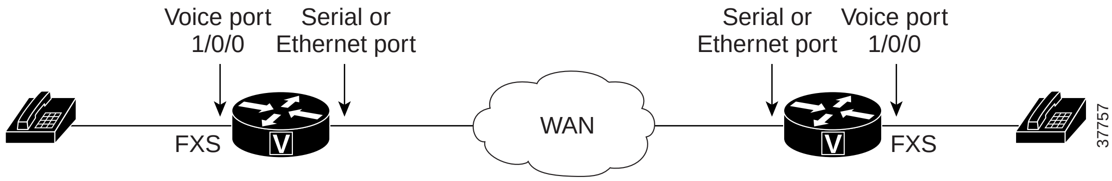


Figure 5 FXS and FXO Signaling Interfaces

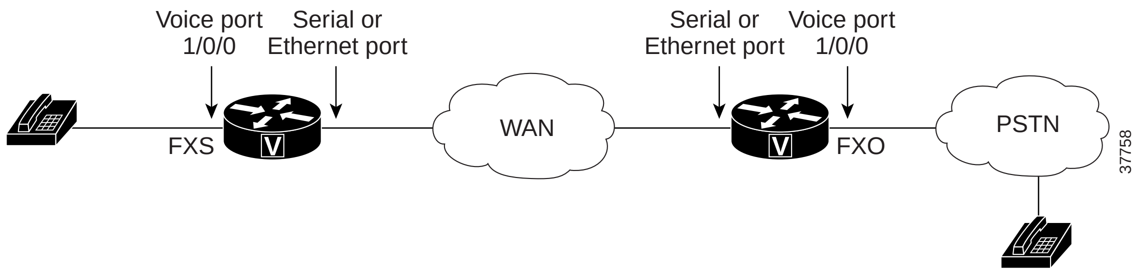
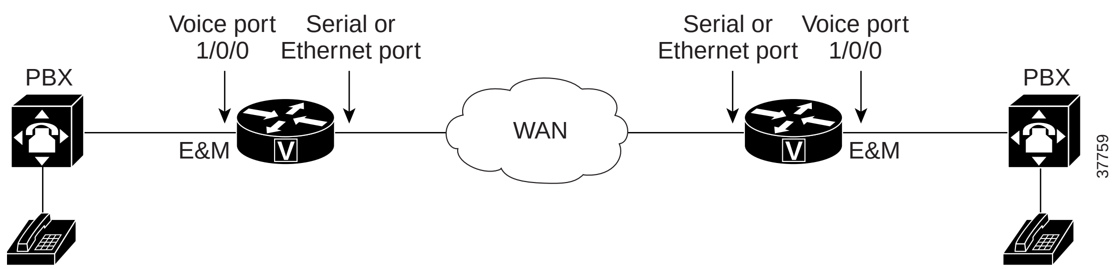


Figure 6 E&M Signaling Interfaces



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 operates only 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 "Configuring Disconnect Supervision" and "Configuring FXO Supervisory Disconnect Tones" sections in the "Fine-Tuning Analog and Digital Voice Ports" chapter 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 telecommunications, 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 2. Cisco voice implementation supports E&M Types I, II, III, and V.

Table 2 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	—	—

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 in the "Configuring Digital Voice Ports" chapter.

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 the following three types of access signaling:

- 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 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.

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. These features include, but are not limited to, the following:

- Disable secondary dial tone on voice ports—By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.
- Cisco router access control lists (ACLs)—Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized Session Initiation Protocol (SIP) or H.323 calls from unknown parties to be processed and connected by the router or gateway.
- Close unused SIP and H.323 ports—If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplex (TDM) trunks or IP, close the unused H.323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.
- Change SIP port 5060—If SIP is actively used, consider changing the port to something other than well-known port 5060.
- SIP registration—If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.
- SIP Digest Authentication—If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.
- Explicit incoming and outgoing dial peers—Use explicit dial peers to control the types and parameters of calls allowed by the router, especially in IP-to-IP connections used on CME, SRST, and Cisco UBE. Incoming dial peers offer additional control on the sources of calls, and outgoing dial peers on the destinations. Incoming dial peers are always used for calls. If a dial peer is not explicitly defined, the implicit dial peer 0 is used to allow all calls.
- Explicit destination patterns—Use dial peers with more granularity than .T for destination patterns to block disallowed off-net call destinations. Use class of restriction (COR) on dial peers with specific destination patterns to allow even more granular control of calls to different destinations on the PSTN.
- Translation rules—Use translation rules to manipulate dialed digits before calls connect to the PSTN to provide better control over who may dial PSTN destinations. Legitimate users dial an access code and an augmented number for PSTN for certain PSTN (for example, international) locations.
- Tcl and VoiceXML scripts—Attach a Tcl/VoiceXML script to dial peers to do database lookups or additional off-router authorization checks to allow or deny call flows based on origination or destination numbers. Tcl/VoiceXML scripts can also be used to add a prefix to inbound DID calls. If the prefix plus DID matches internal extensions, then the call is completed. Otherwise, a prompt can be played to the caller that an invalid number has been dialed.
- Host name validation—Use the "permit hostname" feature to validate initial SIP Invites that contain a fully qualified domain name (FQDN) host name in the Request Uniform Resource Identifier (Request URI) against a configured list of legitimate source hostnames.
- Dynamic Domain Name Service (DNS)—If you are using DNS as the "session target" on dial peers, the actual IP address destination of call connections can vary from one call to the next. Use voice source groups and ACLs to restrict the valid address ranges expected in DNS responses (which are used subsequently for call setup destinations).

For more configuration guidance, see the "Cisco IOS Unified Communications Toll Fraud Prevention" paper.

Configuring Analog Voice Ports

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. Connections to the PSTN central office (CO) are typically made with digital interfaces.

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

Three other sections in this document provide help with fine-tuning and troubleshooting:

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 12.4.
- Install appropriate voice processing and voice interface hardware on the router. See the "Information About Analog Voice Hardware" section on page 2.
- Gather the following information about the telephony connection of the voice port:

- 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

If you are connecting a voice-port interface 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.

Note The slot and port numbering of interface cards differs for each of the voice-enabled routers. For the specific slot and port designations for your hardware platform, refer to the *Cisco Interface Cards Hardware Installation Guide*. More current information may be available in the release notes for the Cisco IOS software you are using.

Information About Analog Voice Hardware

This section describes the general types of analog voice port hardware available for the following router platforms:

- Cisco 880 Series Routers, page 2
- Cisco 1750 Modular Router, page 3
- Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers, page 3
- Cisco MC3810, page 4

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

Cisco 880 Series Routers

Beginning with Cisco IOS Release 12.4(15)XZ, the Cisco 880 series fixed router platforms support the implementation of analog (FXS/DID/FXO) and digital (BRI S/T) voice ports. The IAD881B, IAD881F, IAD888B, and IAD888F models support voice interface FXS or BRI. The IAD881F and IAD888F models have four FXS ports and the IAD881B and IAD888B models support two ports for ISDN BRI digital voice interface.

In the IAD881B and IAD888B models, the voice BRI interface presents an ISDN S/T interface to connect either to an NT1 terminating an ISDN telephone network (TE-side) or to a TE user device such as an ISDN telephone or PBX (NT-side). In the IAD881B and IAD888B models, the BRI interface is available as the primary voice interface and is intended to be connected to a PBX (network side trunk). All the voice interfaces are onboard though they are recognized as a 4-port FXS VIC and a 2-port BRI VIC in order to leverage existing voice drivers.

Note If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.

The C881 and C888 SRST models automatically detect a failure occuring in the network and initiate a process to auto-configure the router. This process provides call-processing backup redundancy for the IP and FXS phones and helps to ensure that telephony capabilities stay operational. All the IP or analog phones hanging off of a telecommuter site are controlled by the headquarters office call control (Cisco Unified CallManager or CallManager Express). In case of a WAN failure, the telecommuter router allows all phones to re-register to it in SRST mode and allow all inbound and outbound dialing to be routed off to the PSTN (using back up FXO or BRI port). Upon restoration of WAN connectivity, the system automatically shifts call processing back to the primary Cisco Unified Call Manager cluster.

Cisco 1750 Modular Router

The Cisco 1750 modular router provides 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 voice interface card (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.

Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers

The Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers are modular, multifunction platforms that combine dial access, routing, LAN-to-LAN services, and multiservice integration of voice, video, and data in the same device.

Voice network modules installed in Cisco 2600 series, Cisco 3600 series, or Cisco 3700 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.

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:

- Hardware installation documents for Cisco 2600 series
- Hardware installation documents for Cisco 3600 series
- *Cisco Network Modules Hardware Installation Guide*
- *Cisco Interface Cards Installation Guide*

Cisco MC3810

To support analog voice circuits, a Cisco MC3810 must be equipped with an AVM, which supports six analog voice ports. When you install 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, 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, refer to the release notes for the platform and Cisco IOS release you are using.

How to Configure Analog Voice Ports

To configure analog voice ports, complete the following tasks:

- Configuring Basic Parameters on Analog FXO, FXS, or E&M Voice Ports
- Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules

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 "Prerequisites for Configuring Analog Voice Ports" section on page 1 should be gathered before you start this procedure.

If you are configuring a Cisco MC3810 that has HCMs, you should also configure the codec complexity by performing the tasks in the "Configuring Codec Complexity for Analog Voice Ports on the Cisco MC3810 with High-Performance Compression Modules" section on page 8.

Note If you have a Cisco MC3810 or Cisco 3660 router, the **compand-type a-law** command must be configured on the analog ports only. The Cisco 2660, Cisco 3620, and Cisco 3640 routers do not require the configuration of the **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 this section and also in the "Fine-Tuning Analog and Digital Voice Ports" chapter.

After the voice port has been configured, make sure that the ports are operational by performing the tasks described in the following chapters:

- Verifying Analog and Digital Voice-Port Configurations
- Troubleshooting Analog and Digital Voice Port Configurations, page 137

For more information on these and other voice port commands, refer to the *Cisco IOS Voice 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 to determine the syntax choices that are available.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port* (for Cisco 880 series)
   or
   voice-port *slot*/*port* (for Cisco 1750 and Cisco MC3810)
   or
   voice-port *slot*/*subunit*/*port* (for Cisco 2600, Cisco 3600, and Cisco 3700 series)
4. **signal** {**loop-start** | **ground-start**} (for FXO and FXS)

   or

   signal {**wink-start** | **immediate-start** | **delay-dial**} (for E&M)
5. **cptone** *locale*
6. **dial-type** {**dtmf** | **pulse**}
7. **operation** {**2-wire** | **4-wire**}
8. **type** {**1** | **2** | **3** | **5**}
9. **ring frequency** {**25** | **50**} (Cisco 1750 router and Cisco 2600, Cisco 3600, and Cisco 3700 series)
   or

   ring frequency {**20** | **30**} (Cisco MC3810)
10. **ring number** *number*
11. **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] | [**define** *pulse interval*]}
12. **description** *string*
13. **no shutdown**

DETAILED STEPS

	Command	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**voice-port** *slot/port*
	
	Example:
	`Router(config)# voice-port 1/0`
	
	or
	
	voice-port *slot/subunit/port*
	
	Example:
	`Router(config)# voice-port 1/0/0`
		Enters voice-port configuration mode.
		Note The slash must be entered between *slot* and *port.*
		• Valid entries vary by router platform; enter the **show voice port summary** command for available values.
		Note For the Cisco 880 series platforms, the command syntax does not include a slot number, only the port is identified.
		If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.

Step 4	**signal** {**loop-start** | **ground-start**}
	
	Example:
	`Router(config-voiceport)# signal ground-start`
	
	or
	
	signal {**wink-start** | **immediate-start** | **delay-dial**}
	
	Example:
	`Router(config-voiceport)# signal wink-start`
		Selects the access signaling type to match that of the telephony connection you are making.
		Note Configuring the **signal** keyword for one voice port on a Cisco 2600 or Cisco 3600 series router VIC changes the signal value for both ports on the VIC.

Step 5	**cptone** *locale* **Example:** `Router(config-voiceport)# cptone us`	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**.

	Command	Purpose
Step 6	**dial-type** {**dtmf** | **pulse**} **Example:** `Router(config-voiceport)# dial-type dtmf`	(FXO only) Specifies the dialing method for outgoing calls.
Step 7	**operation** {**2-wire** | **4-wire**} **Example:** `Router(config-voiceport)# operation 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 8	**type** {**1** | **2** | **3** | **5**} **Example:** `Router(config-voiceport)# type 2`	(E&M only) Specifies the type of E&M interface to which this voice port is connecting. See Table 2 in the *Voice Port Configuration Overview* chapter for an explanation of E&M types. • The default is **1**.

Step 9	**ring frequency** {**25** | **50**}
	
	or
	
	ring frequency {**20** | **30**}
	
	Example:
	`Router(config-voiceport)# ring frequency 50`
	
	`Router(config-voiceport)# ring frequency 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 the ring frequency is not set properly, the attached telephony device may not ring or it may buzz.
		• The keyword default is **25** on the Cisco 1750 router, Cisco 2600 and Cisco 3600 series routers; and **20** on the Cisco MC3810.

Step 10	**ring number** *number* **Example:** `Router(config-voiceport)# ring number 1`	(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 11	**ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] | [**define** *pulse interval*]} **Example:** `Router(config-voiceport)# ring cadence pattern01`	(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 default is the pattern specified by the **cptone** locale that has been configured.
Step 12	**description** *string* **Example:** `Router(config-voiceport)# description 255`	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 13	**no shutdown** **Example:** `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 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 type 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 in dial peers. For more information, refer to the *Dial Peer Configuration on Voice Gateway Routers* document.

Codec complexity refers to the amount of processing power that a codec compression method requires. The greater the codec complexity, the fewer the calls that the DSP interfaces can handle. Codec complexity is either medium or high. The default is medium. All medium-complexity codecs can also run in high-complexity mode, but fewer (usually half as many) channels are available per DSP. The codec complexity value determines the choice of codecs that are available in the dial peers when the **codec** command has been configured. 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 Command Reference*.

To configure codec complexity on the Cisco MC3810 using HCMs, use the following commands:

SUMMARY STEPS

1. **enable**
2. **show voice dsp**
3. **configure terminal**
4. **voice-card 0**
5. **codec complexity** {**high** | **medium**}

DETAILED STEPS

	Command	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`show voice dsp`
	
	Example:
	`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 3.

	Command	Purpose
Step 3	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 4	**voice-card 0** **Example:** `Router(config)# voice-card 0`	Enters voice-card configuration mode and specifies voice card 0.
Step 5	**codec complexity** {**high** | **medium**} **Example:** `Router(config-voicecard)# codec complexity high`	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. **Note** If two HCMs are installed, this command configures both HCMs at once.

Configuring Digital Voice Ports

The digital voice port commands discussed in this section configure channelized T1 or E1 connections; for information on ISDN connections, refer to the *Cisco IOS ISDN Voice 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 endpoints whether a line is idle or busy, whether a device is on-hook or off-hook, and whether a connection is being attempted. An endpoint can be a central office (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: Foreign Exchange Service (FXS), Foreign Exchange Office (FXO), and Ear and Mouth (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, 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 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 document.

This chapter 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*.
- Install appropriate voice processing and voice interface hardware on the router. See the "Information About Digital Voice Hardware" section on page 3.
- (Cisco 2600 and Cisco 3600 series routers) For digital T1 packet voice trunk network modules, install Cisco IOS Release 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 Cisco 3600 series routers) For digital E1 packet voice trunk network modules, install Cisco IOS Release 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.
- Before you can run the IP Communications High-Density Digital Voice/Fax Network Module feature on T1/E1 interfaces, you must install an IP Plus image (minimum) of Cisco IOS Release 12.3(7)T or a later release.
- (Cisco MC3810 concentrators) HCMs require Cisco IOS Release 12.2(1) or a later release.

- (Cisco 7200 and Cisco 7500 series routers) For digital T1/E1 voice port adapters, install Cisco IOS Release 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.
- Gather the following information about the telephony network connection of the voice port:
  - Line interface: T1 or E1
  - Signaling interface: FXO, FXS, or E&M. If the interfaces are PRI or BRI, refer to the *Cisco IOS ISDN Voice 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

Note The slot and port numbering of interface cards differs for each of the voice-enabled routers. For specific slot and port designations, refer to the hardware installation documentation for your router platform. More current information may be available in the release notes that accompany the Cisco IOS software you are using.

After the controllers have been configured, the **show voice port summary** command can 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.

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

```
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
```

Information About Digital Voice Hardware

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

- Cisco 880 Series Routers, page 4
- Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers, page 4
- Cisco 7200 and Cisco 7500 Series Routers, page 5
- Cisco AS5300, page 6
- Cisco AS5350 and Cisco AS5400 Universal Gateways, page 6
- Cisco AS5800, page 7
- Cisco AS5850 Universal Gateway, page 7

- Cisco Catalyst 6500 Series Switches and Cisco 7600 Series Routers with Communication Media Modules, page 8
- Cisco MC3810, page 8

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

Cisco 880 Series Routers

Beginning with Cisco IOS Release 12.4(15)XZ, the Cisco 880 series fixed router platforms support the implementation of analog (FXS/DID/FXO) and digital (BRI S/T) voice ports. The IAD881B, IAD881F, IAD888B, and IAD888F models support voice interface FXS or BRI. The IAD881F and IAD888F models have four FXS ports and the IAD881B and IAD888B models support two ports for ISDN BRI digital voice interface.

In the IAD881B and IAD888B models, the voice BRI interface presents an ISDN S/T interface to connect either to an NT1 terminating an ISDN telephone network (TE-side) or to a TE user device such as an ISDN telephone or PBX (NT-side). In the IAD881B and IAD888B models, the BRI interface is available as the primary voice interface and is intended to be connected to a PBX (network side trunk). All the voice interfaces are onboard though they are recognized as a 4-port FXS VIC and a 2-port BRI VIC in order to leverage existing voice drivers.

Note If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.

The C881and C888 SRST models automatically detect a failure occuring in the network and initiate a process to auto-configure the router. This process provides call-processing backup redundancy for the IP and FXS phones and helps to ensure that telephony capabilities stay operational. All the IP or analog phones hanging off of a telecommuter site are controlled by the headquarters office call control (Cisco Unified CallManager or CallManager Express). In case of a WAN failure, the telecommuter router allows all phones to re-register to it in SRST mode and allow all inbound and outbound dialing to be routed off to the PSTN (using back up FXO or BRI port). Upon restoration of WAN connectivity, the system automatically shifts call processing back to the primary Cisco Unified Call Manager cluster.

Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers

Digital voice hardware on Cisco 2600 series, Cisco 3600 series, and Cisco 3700 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 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 allow enterprises or service providers, using the voice-equipped routers as customer premises 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.

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 time-division multiplex (TDM) cross-connect, uses circuit switching rather than the digital signal processors (DSPs) that VoIP technology employs. (Drop-and-insert is described in the *Trunk Management Features* document.

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 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:

- Hardware installation documents for Cisco 2600 series
- Hardware installation documents for Cisco 3600 series
- *Cisco Network Modules Hardware Installation Guide*
- *Cisco Interface Cards 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 reduced instruction set computing (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 CPE, 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 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 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, refer to the release notes for the platform and Cisco IOS release you are using.

Cisco AS5300

The Cisco AS5300 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 can support a total of 96 or 120 simultaneous voice calls.

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:

- *Hardware installation documents for Cisco AS5300*
- *Configuration documents for Cisco AS5300*

Cisco AS5350 and Cisco AS5400 Universal Gateways

The Cisco AS5350 and Cisco AS5400 universal gateways are versatile data and voice communications platforms that provide the functions of a gateway, router, and digital modems in a single modular chassis.

The gateways are intended for Internet service providers (ISPs), telecommunications carriers, and other service providers that offer managed Internet connections, and also medium to large sites that provide both digital and analog access to users on an enterprise network.

The cards that reside in the Cisco AS5350 and AS5400 chassis, sometimes referred to as dial feature cards (DFCs), are of two types: trunk cards, which provide an E1, T1, or T3 interface, and universal port cards, which host the universal DSPs that dynamically handle voice, dial, and fax calls.

For more information, refer to the following publications:

- *Cisco AS5350 and AS5400 Universal Gateway Card Installation Guide*
- *Cisco AS5350 and AS5400 Universal Gateway Software Configuration Guide*

Cisco AS5800

The Cisco AS5800 has 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 front panel of the DSC. 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 21).

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 1344 voice calls in split-dial-shelf configuration with two 7206VXR router shelves.

For more information, refer to the following publications:

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

Cisco AS5850 Universal Gateway

The Cisco AS5850 is a high-density ISDN and port WAN aggregation system that provides both digital and analog call termination. It is intended to be used in service-provider dial point-of-presence (POP) or centralized-enterprise dial environments. The feature cards and the route switch controller (RSC) communicate over a nonblocking interconnect that supports Fast Ethernet and full-duplex service.

The Cisco AS5850 contains ingress interfaces (CT3 and CE1/PRI) that terminate ISDN and modem calls and break out individual calls (DS0s) from the appropriate telco services. Digital or ISDN calls are terminated on the trunk-card HDLC controllers, and analog calls are sent to port resources on the same card or on separate port cards. As a result, any DS0 can be mapped to any HDLC controller or port module. Unlike the Cisco AS5800, trunk-termination and port-handling services can be performed on the same card in the same slot.

For more information, refer to the following publications:

- *Cisco AS5850 Hardware Installation Guide*
- *Cisco AS5850 Universal Gateway Operations, Administration, Maintenance, and Provisioning Guide*

Cisco Catalyst 6500 Series Switches and Cisco 7600 Series Routers with Communication Media Modules

The Communication Media Module (CMM) acts as the VoIP gateway and media services module by using Media Gateway Control Protocol (MGCP), H.323, and SIP protocols with Cisco CallManager and other call agents. The CMM can support single or multiple Cisco CallManagers in an IP communication network.

These VoIP gateway and media services features are provided through the four different types of CMM port adapters as shown in Table 1.

Table 1 CMM Port Adapters

CMM Port Adapters	Description
• WS-SVC-CMM-6T1 • WS-SVC-CMM-6E1	The 6-port T1 and E1 port adapters have onboard digital signal processor (DSP) resources that allow you to connect the interfaces to the public switched telephone network (PSTN) or private branch exchanges (PBXs) through T1/E1R2 Channel Associated Signaling (CAS) or T1/E1 ISDN Primary Rate Interface (PRI). The DSP resources on the port adapters provide packetization, echo cancellation, fax relay, tone detection and generation, concealment, and jitter buffers.
WS-SVC-CMM-24FXS	The 24-port FXS port adapter has onboard DSP resources that allow the FXS interfaces to emulate the central office (CO) or PBX analog trunk lines by providing service to analog phones and fax machines, which behave as if connected to a standard CO or PBX line.
WS-SVC-CMM-ACT	The ACT port adapter has DSP resources for conferencing, transcoding, and media termination point (MTP) services. A CMM with an ACT port adapter supports a single conference with up to 64 participants. A single ACT port adapter supports up to 128 audio conference ports, which can be distributed among different conferences of two or more parties.

For specific configuration information for the Catalyst 6500 series and Cisco 7600 series, see the following documents:

- *Cisco 6500 and 7600 series Manager Installation Guide, Release 2.1*
- *Cisco 6500 and 7600 series Manager User Guide, Release 2.1*
- *Cisco 6500 and 7600 series Manager Release Notes, Release 2.1*

For specific installation and configuration information for the CMM, see the following document:

- *Catalyst 6500 Series and Cisco 7600 Series CMM Installation and Verification Note*
- *Cisco Communication Media Module Voice Features for Catalyst 6500 Series and Cisco 7600 Series*

Cisco MC3810

To support a T1 or E1 digital voice interface, the Cisco MC3810 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 BNC connectors.

Optional HCMs can replace standard VCMs to operate according to the voice compression coding algorithm (codec) specified when the Cisco MC3810 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 cannot be combined with a VCM in the same chassis.

For more information, refer to the following publications:

- *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*
- *Cisco MC3810 Multiservice Concentrator Configuration Guide*

How to Configure Digital T1/E1 Voice Ports

This section describes commands for the basic configuration of digital voice ports. Make sure you have all the data recommended in the "Prerequisites for Configuring Digital Voice Ports" section on page 2 before starting these procedures.

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 on Digital T1/E1 Voice Ports, page 10

  Codec complexity refers to the amount of processing power assigned to a codec method 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 Cisco 7500 series.

- Configuring Controller Settings for Digital T1/E1 Voice Ports, page 19

  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 29

  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 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 51.

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

- Verifying Analog and Digital Voice-Port Configurations, page 125
- Troubleshooting Analog and Digital Voice Port Configurations, page 137

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

Configuring Codec Complexity on Digital T1/E1 Voice Ports

This section provides two configuration task tables: one for the Cisco 2600, Cisco 3600, and Cisco 3700 series routers and the Cisco MC3810 concentrator, which use voice-card configuration mode, and the second for the Cisco 7200 and Cisco 7500 series routers, which use DSP interface configuration mode. The task tables can be found in the following sections:

- Configuring Codec Complexity on Cisco 880 Series, Cisco 2600, Cisco 3600, Cisco 3700 Series and Cisco MC3810, page 10
- Configuring Codec Complexity on Cisco 7200 Series and Cisco 7500 Series Routers, page 17

Configuring Codec Complexity on Cisco 880 Series, Cisco 2600, Cisco 3600, Cisco 3700 Series and Cisco MC3810

On the Cisco 880 series, Cisco 2600, Cisco 3600, Cisco 3700, Cisco 7200, and Cisco 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, the codec complexity setting applies to both HCMs if two HCMs are installed.

Note On Cisco 2600, Cisco 3600, and Cisco 3700 series routers with digital T1/E1 packet voice trunk network modules, codec complexity cannot be configured if DS0 or PRI groups are configured. If DS0 or PRI groups are configured, see the "Changing Codec Complexity on Cisco 880 Series, Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco MC3810" section on page 11.

To configure codec complexity for digital voice ports on the Cisco 880 series, Cisco 2600 series, Cisco 3600 series, and Cisco 3700 series routers, and for voice ports on HCMs on the Cisco MC3810, use the following commands:

SUMMARY STEPS

1. **enable**
2. **show voice dsp**
3. **configure terminal**
4. **voice-card** *slot*
5. **codec complexity** {**high** | **medium**}

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**show voice dsp** **Example:** `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 DSP channels are in the idle state, continue to Step 2.
Step 3	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 4	**voice-card** *slot* **Example:** `Router(config)# voice-card 0`	Enters voice card-configuration mode for the card or cards in the slot specified. Range is 0 to 5.
Step 5	**codec complexity** {**high** | **medium**} **Example:** `Router(config-voicecard)# codec complexity high`	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. Default is **medium**. **Note** On the Cisco MC3810, 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.

Changing Codec Complexity on Cisco 880 Series, Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco MC3810

To change codec complexity after the controller and voice ports have already been configured, use the following commands:

Note Use the **show voice dsp** command to check the DSP voice channel activity. If any DSP voice channels are in the busy state, the codec complexity cannot be changed. You must clear all calls before performing the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port:ds0-group-number*
4. **shutdown**
5. **exit**

6. **controller** {**t1** | **e1**} *slot/port*
7. **no ds0-group** *ds0-group-number*

   or

   no pri-group timeslots *timeslot-list*
8. **exit**
9. **voice-card** *slot*
10. **codec complexity** {**high** | **medium**} [**ecan-extended**]
11. **exit**
12. Repeat Step 6, then continue with Step 13.
13. **ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}

    or

    pri-group timeslots *timeslot-list*
14. **exit**
15. Repeat Step 3, then continue with Step 16.
16. **no shutdown**
17. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** *slot/port*:*ds0-group-number* **Example:** `Router(config)# voice-port 1/0:23`	Enters voice-port configuration mode on the selected slot, port, and DS0 group. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice CommandReference*. **Note** For the Cisco 880 series platforms, the command syntax does not include a slot number, only the port is identified. If the primary voice interface is FXS and the backup is BRI, then ports 0, 1, 2, and 3 are analog voice ports, and ports 4 and 5 are digital. If the primary voice interface is BRI, then ports 1, 2, 3, and 4 are digital.
Step 4	**shutdown** **Example:** `Router(config-voiceport)# shutdown`	Shuts down all voice ports assigned to the T1 interface on the voice card.
Step 5	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.
Step 6	**controller** {**t1** | **e1**} *slot/port* **Example:** `Router(config)# controller t1 1/0`	Enters controller configuration mode on the T1 controller on the selected slot and port.

Step 7	**no ds0-group** *ds0-group-number*
	
	or
	
	no pri-group timeslots *timeslot-list*
	
	Example:
	`Router(config-controller)# no ds0-group 1`
	
	Example:
	`Router(config-controller)# no pri-group timeslots 1,7,9`
		Removes the related DS0 groups.
		
		or
		
		Removes the related PRI group.

Step 8	**exit** **Example:** `Router(config-controller) exit`	Exits controller configuration mode and returns to global configuration mode.
Step 9	**voice-card** *slot* **Example:** `Router(config)# voice-card 1`	Enters voice-card configuration mode on the specified slot. • *slot*—Slot number of the voice card. Range is 0 to 6, depending on platform.

	Command or Action	Purpose
Step 10	**codec complexity** {**high** | **medium**} [**ecan-extended**] **Example:** `Router(voice-card)# codec complexity high ecan-extended`	Changes codec complexity or changes the echo canceller (EC) from the proprietary Cisco G.165 EC to the G.168 extended EC. • **high**—Supports up to six voice or fax calls per DSP module (PVDM-12), using the codecs: G.723, G.728, G.729, G.729 Annex B, GSMEFR, GSMFR, fax relay, or any of the medium complexity codecs. • **medium**—Supports up to 12 voice or fax calls per DSP module (PVDM-12), using the codecs: G.711, G.726, G.729 Annex A, G.729 Annex A with Annex B, and fax relay. Default value. • **ecan-extended**—(Optional) Selects the G.168 extended echo canceller. For more information, see the "How to Configure the Extended G.168 Echo Canceller" section. Specifying the codec complexity restricts the codecs available in dial-peer configuration mode. All voice cards in a gateway must use the same codec complexity.
Step 11	**exit** **Example:** `Router(voice-card) exit`	Exits voice-card configuration mode and returns to global configuration mode.
Step 12	Repeat Step 6, then continue with Step 13.	—

Step 13	**ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-immediate** | **e&m-delay** | **e&m-wink-start** | **fxs-ground-start** | **fxs-loop-start** | **fxo-ground-start** | **fxo-loop-start**}
	
	or
	
	pri-group timeslots *timeslot-list*
	
	Example:
	`Router(config-controller)# ds0-group 0 timeslots 1-24 type e&m-wink-start`
	
	Example:
	`Router(config-controller)# pri-group timeslots 1,7,9`
		Defines the T1 or E1 channels for use by compressed voice calls and the signaling method that the router uses to connect to the PBX or CO.
		Note If you are configuring PRI groups instead of DS0 groups, omit this step and proceed to Step 15.
		
		or
		
		Specifies an ISDN PRI on a channelized T1 or E1 controller.
		Note When configuring PRI groups, you must also configure the **isdn switch-type** command. Also, only one PRI group can be configured on a controller.

Step 14	**exit** **Example:** `Router(config-controller)# exit`	Exits controller configuration mode and completes the process for adding back the PRI groups or DS0 groups.
Step 15	Repeat Step 3, then continue with Step 16.	—

	Command or Action	Purpose
Step 16	**no shutdown** **Example:** `Router(config-controller)# no shutdown`	Saves the controller configurations on the slot and port specified.
Step 17	**end** **Example:** `Router(config-controller)# end`	Exits controller configuration mode and completes the process for bringing the T1 controller back up.

Configuring the Flex Option on Codec Complexity for the Cisco 2600 XM, Cisco 2691, and Cisco 3700 Series

The IP Communications High-Density Digital Voice/Fax Network Module feature enables the flex option for configuring codec complexity.

On the Cisco 2600 XM, Cisco 2691, Cisco 3700 series routers, codec complexity can be configured using the *flex* option for configuring codec complexity. This option allows the DSP to process up to 16 channels. In addition to continuing support for configuring a fixed number of channels per DSP, the flex option enables the DSP to handle a flexible number of channels. The total number of supported channels varies from 6 to 16, depending on which codec is used for a call. Therefore, the channel density varies from 6 per DSP (high-complexity codec) to 16 per DSP (g.711 codec).

Requirements

The following requirements apply to the IP Communications High-Density Digital Voice/Fax Network Module feature.

- When the IP Communications High-Density Digital Voice/Fax Network Module feature is used in a Cisco CallManager network, the CCM 4.0(1) SR1 or CCM 3.3(4) release must be installed.
- Software echo cancellation is the default configuration—G.168-compliant echo cancellation is enabled by default with a coverage of 64 milliseconds.
- Only Packet Fax/Voice DSP modules (PVDM2s) are supported on the IP Communications High-Density Digital Voice/Fax Network Module.
- Only voice interface cards that start with VIC2 are supported in the IP Communications High-Density Digital Voice/Fax Network Module feature except for VIC-1J1, VIC-2DID, and VIC-4FXS/DID.
- The direct inward dial (DID) feature in VIC-4FXS/DID is not supported.
- The CAMA card (VIC-2CAMA) is not supported. Any port on the VIC2-2FXO and the VIC2-4FXO can be software configured to support analog CAMA for dedicated E-911 services (North America only).

Codec Combinations for DSP Sharing

When network modules or PVDM2s on the motherboard are configured for DSP sharing, the codec complexity has to match. A local resource sharing or importing from a remote network module must match its characteristics, that is, a high-complexity network module can only share from another high-complexity network module, whereas a flex-complexity network module can share DSPs from both high-complexity and flex-complexity network modules. Table 2 summarizes the codec combinations for DSP-sharing.

Table 2 Codec Complexity Settings for DSP Resource Sharing Between Local and Remote Sources

Local DSP Resource (Import)	Remote DSP Resource (Export)		
	High complexity	Medium complexity	Flexible complexity
High complexity	Yes	No	No
Medium complexity	Yes	Yes	No
Flexible complexity	Yes	No	Yes

Caution If you are configuring a Cisco 2600 XM router, you should not use the **network-clock-participate** command for slot 1 of the router. This may cause a disruption in service to the router.

Using Flex Mode

In flex mode, you can connect (or configure in the case of DS0 groups and PRI groups) more voice channels to the module than the DSPs can accommodate. This is referred to as oversubscription. If all voice channels should go active simultaneously, the DSPs will be oversubscribed and calls that are unable to allocate a DSP resource will fail to connect.

To enable the IP Communications Voice/Fax Network Module feature, perform this task to configure the voice card for the flex option in codec complexity.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-card** *slot*
4. **codec complexity flex** [**reservation-fixed** {**high** | **medium**}]
5. **voice local-bypass**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-card** *slot* **Example:** `Router(config)# voice-card 1`	Enters voice-card configuration mode and specifies the slot location. • For the *slot* argument, specify a value from 1 to 4, depending on your router.
Step 4	**codec complexity flex** [**reservation-fixed** {**high** | **medium**}] **Example:** `Router(config-voicecard)# codec complexity flex`	Specifies the flex option for codec complexity. • **flex**—Up to 16 calls can be completed per DSP. The number of supported calls varies from 6 to 16, depending on the codec used for a call. In this mode, reservation for analog VICs may be needed for certain appplications such as CAMA E-911 calls because oversubscription of DSPs is possible. If this is true, then the **reservation-fixed** option may be enabled. There is no reservation by default. – **reservation-fixed**—Appears as an option only when there is an analog VIC present. Ensures that sufficient DSP resources are available to handle a call. If you enter this keyword, then specify if the complexity should be high or medium. **Note** You cannot change codec complexity while DS0 groups are defined. If they are already set up, perform the steps in the "Changing Codec Complexity on Cisco 880 Series, Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco MC3810" section on page 11.

Step 5	`voice local-bypass`
	
	Example:
	`Router(config-voicecard)# voice local-bypass`
		Configures local calls to bypass the DSP. This is the default.
		• Using this command enables intranetwork-module hairpinning (no DSPs).
		
		Note For POTS-to-POTS calls between two network modules, hairpinning is not supported. If the connection manager in Cisco IOS software does not automatically handle this, it might be necessary to disable local-bypass so that DSPs are used for these calls.

Step 6	`exit`
	
	Example:
	`Router(config-voicecard)# exit`
		Exits voice-card configuration mode and returns the router to global configuration mode.

Configuring Codec Complexity on Cisco 7200 Series and Cisco 7500 Series Routers

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

Note Use the **show interfaces dspfarm** command to check the DSP voice channel activity. If any DSP voice channels are in the busy state, the codec complexity cannot be changed. You must clear all calls before performing the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dspint dspfarm** *slot*/**0**

   or

   dspint dspfarm *slot*/*port-adapter*/*port*
4. **codec** {**high** | **medium**} [**ecan-extended**]
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**dspint dspfarm** *slot*/**0** or **dspint dspfarm** *slot*/*port-adapter*/*port* **Example:** `Router(config)# dspint dspfarm 2/0`	Enters DSP interface configuration mode for the Cisco 7200 series. or Enters DSP interface configuration mode for the Cisco 7500 series.

	Command or Action	Purpose
Step 4	**codec** {**high** | **medium**} [**ecan-extended**] **Example:** `Router(config-dspfarm)# codec medium ecan-extended`	Sets the codec complexity. • The optional **ecan-extended** keyword selects the G.168 extended echo canceller. This keyword is supported only in Cisco IOS Release 12.2(13)T. For more information, see the "How to Configure the Extended G.168 Echo Canceller" section. • This command affects the choice of codecs available when the **codec** command is used in dial-peer configuration mode.
Step 5	**exit** **Example:** `Router(config-dspfarm)# exit`	Exits to global configuration mode.

Cisco 7200 Series

On the Cisco 7200 series, the PA-MCX-2TE1 port adapter (PA) card can be used for making voice calls. This PA does not have any DSPs but uses the DSP resources of the PA-VXC-2TE1+ card present in another slot. If the PA-MCX card is used, codec complexity is configured for PA-VXC, while all other echo cancellation configurations are done for PA-MCX.

The PA-MCX card borrows the DSP resources from the PA-VXC, PA-VXB, or PA-VXA card. If one of the PA-VXC, PA-VXB, or PA-VXA cards has extended echo cancellation configured on the DSP interface, extended echo cancellation is enabled for the PA-MCX card. It is recommended that you have the same codec complexity and echo cancellation configuration on all the PA-VXC, PA-VXB, or PA-VXA cards in the router.

Cisco AS5300

Codec support on the Cisco AS5300 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; refer to the *Dial Peer Configuration on Voice Gateway Routers* document.

Cisco AS5800

Codec support is selected on Cisco AS5800 access servers during dial peer configuration. Refer to the *Dial Peer Configuration on Voice Gateway Routers* document.

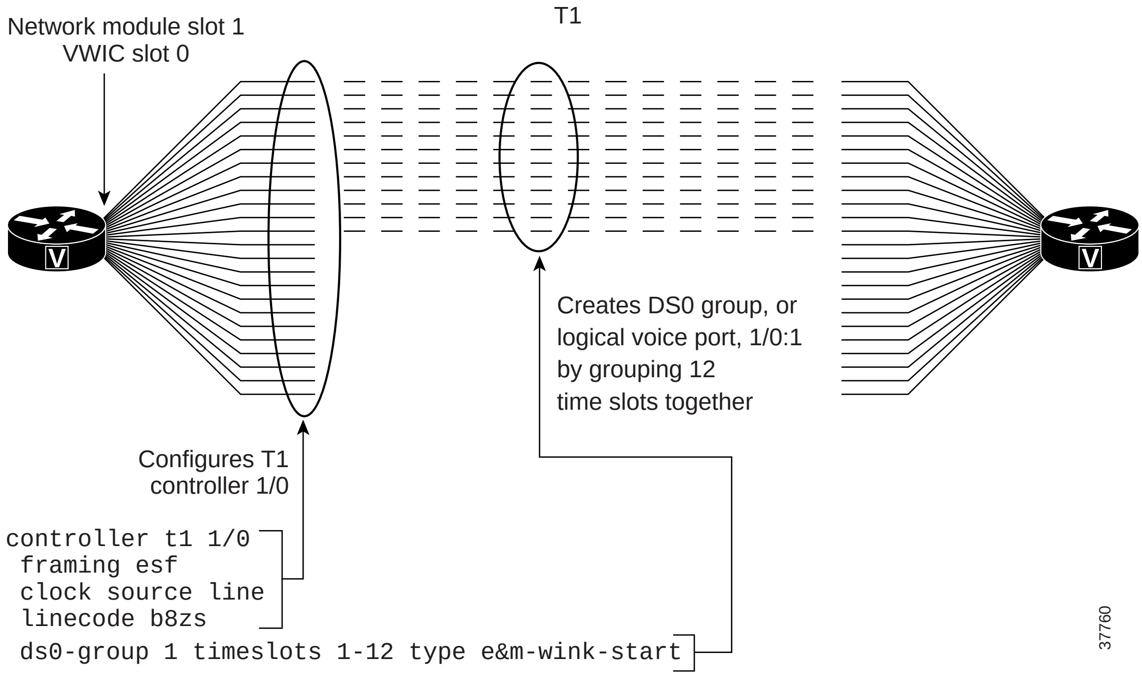
Configuring Controller Settings for Digital T1/E1 Voice Ports

The controller configuration for digital T1/E1 voice ports must match the line characteristics of the telephony network connection so that voice and signaling can be transferred between them and so that logical voice ports, or DS0 groups, may be established.

Figure 1 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 that 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 All controller commands shown in Figure 1, other than **ds0-group**, apply to all time slots in the T1 line.

Figure 1 T1 Controller Configuration on Cisco 2600 or Cisco 3600 Series Routers



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

- Framing Formats on Digital T1/E1 Voice Ports, page 20
- Clock Sources on Digital T1/E1 Voice Ports, page 21
- Network Clock Timing, page 24
- Line Coding on Digital T1/E1 Voice Ports, page 26
- DS0 Groups on Digital T1/E1 Voice Ports, page 26

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

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 SF or 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 PRI configurations.

E1 lines can be configured for 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 becauseits 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:

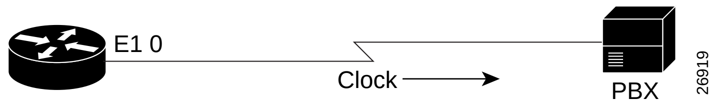
- When both ports are set to line clocking with no primary specification, port 0 is the default primary clock source and port 1 is the default secondary clock source.
- When both ports are set to line and one port is set as the primary clock source, the other port is by default the backup or secondary 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 2. 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:

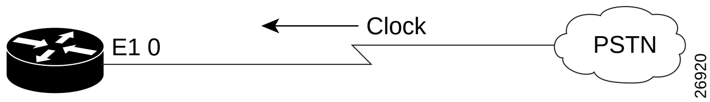
```
controller E1 1/0
 framing crc4
 linecoding hdb3
 clock source internal
 ds0-group timeslots 1-15 type e&m-wink-start
```

Figure 2 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 3). The PLL clocking is driven by the clock reference on the receive (Rx) side of the digital line connection.

Figure 3 Single E1 Port Receiving Clocking from the Line

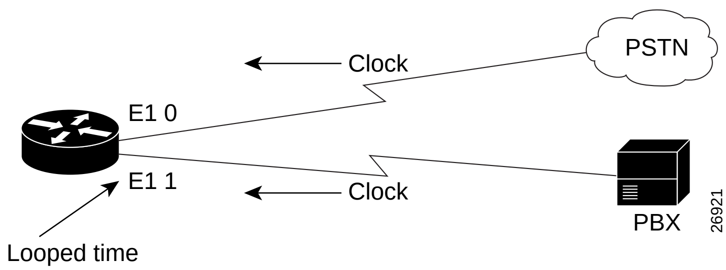


The following configuration sets up this clocking method:

```
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 4.

Figure 4 Dual E1 Ports Receiving Clocking from the Line



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.

The following configuration sets up this clocking method (controller E1 1/0 is connected to the CO; controller E1 1/1 is connected to the PBX:

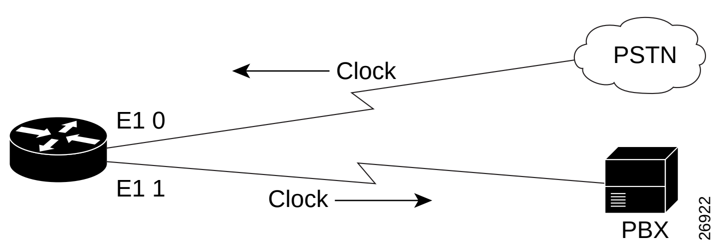
```
controller E1 1/0
 framing crc4
 linecoding hdb3
 clock source line primary
 ds0-group timeslots 1-15 type e&m-wink-start
!
controller E1 1/1
 framing crc4
 linecoding hdb3
 clock source line
 ds0-group timeslots 1-15 type e&m-wink-start
```

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 two T1/E1 lines terminate on a two-port interface card, such as the VWIC-2MFT, and 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 or 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 5). If controller E1 0 fails, the PLL internally generates the clock reference to drive E1 1.

Figure 5 Dual E1 Ports—One Receiving and One Providing Clocking



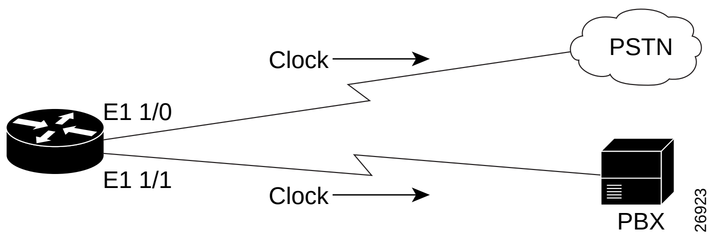
The following configuration sets up this clocking method:

```
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 6).

Figure 6 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
```

Network Clock Timing

Voice systems that pass digitized (pulse code modulation or PCM) speech have always relied on the clocking signal being embedded in the received bit stream. This reliance allows connected devices to recover the clock signal from the bit stream, and then use this recovered clock signal to ensure that data on different channels keep the same timing relationship with other channels.

If a common clock source is not used between devices, the binary values in the bit streams may be misinterpreted because the device samples the signal at the wrong moment. As an example, if the local timing of a receiving device is using a slightly shorter time period than the timing of the sending device, a string of eight continuous binary 1s may be interpreted as nine continuous 1s. If this data is then re-sent to further downstream devices that used varying timing references, the error could be compounded. By ensuring that each device in the network uses the same clocking signal, you can ensure the integrity of the traffic.

If timing between devices is not maintained, a condition known as clock slip can occur. Clock slip is the repetition or deletion of a block of bits in a synchronous bit stream due to a discrepancy in the read and write rates at a buffer.

Slips are caused by the inability of an equipment buffer store (or other mechanisms) to accommodate differences between the phases or frequencies of the incoming and outgoing signals in cases where the timing of the outgoing signal is not derived from that of the incoming signal.

A T1 or E1 interface sends traffic inside repeating bit patterns called frames. Each frame is a fixed number of bits, allowing the device to see the start and end of a frame. The receiving device also knows exactly when to expect the end of a frame simply by counting the appropriate number of bits that have come in. Therefore, if the timing between the sending and receiving device is not the same, the receiving device may sample the bit stream at the wrong moment, resulting in an incorrect value being returned.

Even though Cisco IOS software can be used to control the clocking on these platforms, the default clocking mode is effectively free running, meaning that the received clock signal from an interface is not connected to the backplane of the router and used for internal synchronization between the rest of the router and its interfaces. The router will use its internal clock source to pass traffic across the backplane and other interfaces.

For data applications, this clocking generally does not present a problem as a packet is buffered in internal memory and is then copied to the transmit buffer of the destination interface. The reading and writing of packets to memory effectively removes the need for any clock synchronization between ports.

Digital voice ports have a different issue. It would appear that unless otherwise configured, Cisco IOS software uses the backplane (or internal) clocking to control the reading and writing of data to the DSPs. If a PCM stream comes in on a digital voice port, it will be using the external clocking for the received bit stream. However, this bit stream will not necessarily be using the same reference as the router backplane, meaning the DSPs may misinterpret the data coming in from the controller.

This clocking mismatch is seen on the router's E1 or T1 controller as a clock slip—the router is using its internal clock source to send the traffic out the interface but the traffic coming in to the interface is using a completely different clock reference. Eventually, the difference in the timing relationship between the transmit and receive signal becomes so great that the controller registers a slip in the received frame.

To eliminate the problem, change the default clocking behavior through Cisco IOS configuration commands. It is *absolutely critical* to set up the clocking commands properly.

Even though these commands are optional, we strongly recommend you enter them as part of your configuration to ensure proper network clock synchronization:

network-clock-participate [**slot** *slot-number* | **wic** *wic-slot* | **aim** *aim-slot-number*]
network-clock-select *priority* {**bri** | **t1** | **e1**} *slot/port*

The **network-clock-participate** command allows the router to use the clock from the line via the specified slot/WIC/AIM and synchronize the onboard clock to the same reference.

If multiple VWICS are installed, the commands must be repeated for each installed card. The system clocking can be confirmed using the **show network clocks** command.

Caution If you are configuring a Cisco 2600 XM voice gateway with an NM-HDV2 or NM-HD-2VE installed in slot 1, do not use the **network-clock-participate slot 1** command in the configuration. In this particular hardware scenario, the **network-clock-participate slot 1** command is not necessary.

If the **network-clock-participate slot 1** command is configured, voice and data connectivity on interfaces terminating on the NM-HDV2 or NM-HD-2VE network module may fail to operate properly. Data connectivity to peer devices may not be possible, and even loopback plug tests to the serial interface spawned via a channel group configured on the local T1/E1 controller will fail. Voice groups such as CAS DS0 groups and ISDN PRI groups may fail to signal properly. The T1/E1 controller may accumulate large amounts of timing slips and Path Code Violations (PCVs) and Line Code Violations (LCVs).

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 AMI and 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 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-number*, 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 **ds0-group 1 timeslots 1-24 type e&m-wink** command 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 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, Cisco 3600, and Cisco 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:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **card type** {**t1** | **e1**} *slot*
   or
   card type {**t1** | **e1**} *slot*/*port*
4. **controller** {**t1** | **e1**} *slot*/*port*
   or
   controller {**t1** | **e1**} *number*
   or
   controller {**t1** | **e1**} *shelf*/*slot*/*port*
5. **framing** {**sf** | **esf**}
   or
   framing {**crc4** | **no-crc4**} [**australia**]
6. **clock source** {**line** [**primary** | **secondary**] | **internal**}
7. **linecode** {**ami** | **b8zs**}
   or
   linecode {**ami** | **hdb3**}

8. **ds0-group** *ds0-group-number* **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**}
9. **no shutdown**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`card type`** {**`t1`** | **`e1`**} *`slot`* **Example:** `Router(config)# card type t1 0`	Defines the card as T1 or E1 and identifies the location.

Step 4	**`controller`** {**`t1`** | **`e1`**} *`slot/port`*
	or
	`controller` {**`t1`** | **`e1`**} *`number`*
	or
	`controller` {**`t1`** | **`e1`**} *`shelf/slot/port`*
	
	Example:
	`Router(config)# controller t1 1/0`
	or
	
	Example:
	`Router(config)# controller t1 1`
	or
	
	Example:
	`Router(config)# controller t1 1/0/0`
		Enters controller configuration mode and specifies either T1 or E1 for the line.
		• For the Cisco 2600, Cisco 3600 series, Cisco MC3810, and Cisco 7200 series, identifies the slot and port.
		• For the Cisco AS5300, identifies the port number.
		• For the Cisco AS5800 and Cisco 7500 series, identifies the shelf, slot, and port number.

	Command	Purpose

Step 5	**framing** {**sf** | **esf**}
	
	or
	
	framing {**crc4** | **no-crc4**} [**australia**]
	
	Example:
	`Router(config-controller)# framing esf`
	
	or
	
	Example:
	`Router(config-controller)# framing crc4`
		Selects frame type for T1 or E1 line.
		• For T1, the frame type can be **sf** or **esf**. Default for T1 is **sf**.
		• For E!, the frame type can be **crc4** or **no crc4** or **australia**. Default for E1 is **crc4**.

Step 6	**clock source** {**line** [**primary** | **secondary**] | **internal**} **Example:** `Router(config-controller)# clock source line primary`	Configures the clock source. • Default is **line**. • For more information about clock sources, see the "Clock Sources on Digital T1/E1 Voice Ports" section on page 21.

Step 7	**linecode** {**ami** | **b8zs**}
	or
	linecode {**ami** | **hdb3**}
	
	Example:
	`Router(config-controller)# linecode b8zs`
	or
	
	Example:
	`Router(config-controller)# linecode hdb3`
		Specifies the line encoding to use for T1 or E1 line.
		• For T1, the line encoding can be ami or b8zs. Default for T1 is **ami**.
		• For E1, the line encoding can be ami or hdb3. Default for E1 is **hdb3**.

Step 8	**ds0-group** *ds0-group-number* **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**} **Example:** `Router(config-controller)# ds0-group 30 timeslots 0 type e&m-immediate-start`	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, refer to the *Cisco IOS Voice Command Reference*.
Step 9	**no shutdown** **Example:** `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 6 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 to determine the syntax choices that are available.

To configure basic parameters for digital T1/E1 voice ports, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port*
   or
   voice-port *slot*/*port*:*ds0-group-number*
   or
   voice-port *slot*/*port-adapter*:*ds0-group-number*
   or
   voice-port *slot*/*port-adapter*/*slot*:*ds0-group-number*
   or
   voice-port *controller*:{*ds0-group-number* | **D**}
   or
   voice-port *slot*/*controller*:{*ds0-group-number* | **D**}
   or
   voice-port *shelf*/*slot*/*port*:*ds0-group-number*
4. **type** {**1** | **2** | **3** | **5**}
5. **cptone** *locale*
6. **compand-type** {**u-law** | **a-law**}
7. **ring frequency** {**25** | **50**}
8. **ring number** *number*
9. **ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse interval*]}
10. **description** *string*
11. **no shutdown**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**`voice-port`** *`port`*
	or
	`voice-port` *`slot/port`*`:`*`ds0-group-number`*
	or
	`voice-port` *`slot/port-adapter`*`:`*`ds0-group-number`*
	or
	`voice-port` *`slot/port-adapter/slot`*`:`*`ds0-group-number`*
	or
	`voice-port` *`controller`*`:{`*`ds0-group-number`* `|` **`D`**`}`
	or
	`voice-port` `slot/`*`controller`*`:{`*`ds0-group-number`* `|` **`D`**`}`
	or
	`voice-port` *`shelf/slot/port`*`:`*`ds0-group-number`*
	
	Example:
	`Router(config)# voice-port 1:0`
	or
	
	Example:
	`Router(config)# voice-port 1/1:0`
	or
	
	Example:
	`Router(config)# voice-port 1/1/1:1`
	or
	
	Example:
	`Router(config)# voice-port 1:1`
	or
	
	Example:
	`Router(config)# voice-port 1/0 D`
	or
	
	Example:
	`Router(config)# voice-port 1/2/0:1`
		Enters voice-port configuration mode and identifies the port to be configured.
		• For the Cisco 880 series, specify the port number.
		• For the Cisco 2600, Cisco 3600, and Cisco 3700 series, specify the slot, port, and DS0 group number.
		• For the Cisco 7200 series, specify the slot, port adapter,and DS0 group number.
		• For the Cisco 7500 series, specify the slot, port adapter, slot, and DS0 group number.
		• For the Cisco AS5300, specify the controller and DS0 group number or the keyword **D**.
		• For the Cisco AS5350, Cisco AS5400, and Cisco AS5850 universal gateways, specify the slot, controller, and DS0 group number or the keyword **D**.
		• For the Cisco AS5800, specify the shelf, slot, port, and DS0 group number.

	Command	Purpose
Step 4	**type** {**1** | **2** | **3** | **5**} **Example:** `Router(config-voiceport)# type 1`	(E&M only) Specifies the type of E&M interface to which this voice port is connected. See Table 3 in the Vo ice Port Configuration Overview chapter for an explanation of E&M types. • Default is 1.
Step 5	**cptone** *locale* **Example:** `Router(config-voiceport)# cptone us`	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. • Default is **us**.
Step 6	**compand-type** {**u-law** | **a-law**} **Example:** `Router(config-voiceport)# compand-type u-law`	(Cisco 2600 and Cisco 3600 series routers.) 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 default for E1 is **a-law**. • The default for T1 is **u-law**. **Note** If you have a 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 7	**ring frequency** {**25** | **50**} **Example:** `Router(config-voiceport)# ring frequency 50`	(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 the ring frequency is not set properly, the attached telephony device may not ring or it may buzz. • Default is **25**.
Step 8	**ring number** *number* **Example:** `Router(config-voiceport)# ring number 1`	(FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router. • Default is **1**.

	Command	Purpose
Step 9	**ring cadence** {[**pattern01** | **pattern02** | **pattern03** | **pattern04** | **pattern05** | **pattern06** | **pattern07** | **pattern08** | **pattern09** | **pattern10** | **pattern11** | **pattern12**] [**define** *pulse interval*]} **Example:** `Router(config-voiceport)# ring cadence pattern01 define 12 15`	(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 10	**description** *string* **Example:** `Router(config-voiceport)# description 1`	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 11	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Activates the voice port.

Fine-Tuning Analog and Digital Voice Ports

The default parameter values for voice ports are usually 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 to determine the syntax choices that are available.

Information About Fine-Tuning Analog and Digital Voice Ports

This section provides basic information about some the features that can be fine-tuned to improve the performance of your voice network:

- Channel Bank Support for T1/E1 Voice Ports—Provides support for the time-division multiplexing (TDM) cross-connect functionality between analog voice ports and digital DS0s on the same NM-HD-2VE using channel associated signaling (CAS).
- Auto Cut-Through—Allows you to connect to PBXs that do not provide an M-lead response.
- Modification of Bit Patterns for Digital Voice Ports—Enables commands for digital voice ports to modify sent or received bit patterns. Different versions of E&M use different ABCD signaling bits to represent idle and seize.
- ANI for Outbound Calling—Allows the automatic number identification (ANI) to be sent for outgoing calls on the Cisco AS5300 (if T1 CAS is configured with the Feature Group-D (FGD)—Exchange Access North American (FGD-EANA) signaling).
- Disconnect Supervision—Configures 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)

- FXO Supervisory Disconnect Tones—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.
- Timeouts Parameters—Modifies values for timeouts. For example, you can adjust the wait time for 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.
- Timing Parameters—Changes a wide range of timing values. For example, you can specify the minimum delay time, in milliseconds, from outgoing seizure to outdial address.
- DTMF Timer—Modifies the value for the DTMF interdigit timer.
- Comfort Noise and Music Threshold for VAD—Specifies the minimal decibel level of music played when calls are put on hold and 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.

How to Configure Fine-Tuning Features for Analog and Digital Voice Ports

To configure the voice port tuning features for analog and digital voice ports, complete these tasks:

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 to determine the syntax choices that are available. Full descriptions of the commands in this section can be found in the *Cisco IOS Voice Command Reference*.

Configuring Channel Bank Support for T1/E1 Voice Ports

The channel bank feature provides support for the time-division multiplexing (TDM) cross-connect functionality between analog voice ports and digital DS0s on the same NM-HD-2VE using channel associated signaling (CAS).

To establish a channel bank connection between an analog voice port and a T1 DS0, configure the **connect** (voice-port) command in global configuration mode. To verify the channel bank connection, use the **show connection all** command.

Restrictions for Channel Bank Support

- The configuration for cross-connect must be on the same network module.
- A maximum of four Foreign Exchange Service (FXS) or Foreign Exchange Office (FXO) ports can be cross-connected to a T1 interface.
- A BRI-to-PRI cross-connect cannot be configured.
- Analog-to-BRI/PRI cross-connect cannot be configured; the only connection for analog is analog-to-T1/E1 CAS (ds0-group).
- The **local-bypass** command has no effect when cross-connect is configured. It is applicable only to calls that are hairpinned via POTS-to-POTS dial peers.
- The DS0 group must contain only one time slot. The signaling type of the DS0 group must match that of the analog voice port.
- If the channel bank feature is used for the T1 controller, the rest of the unused DS0 group cannot be used for fractional PRI signaling.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **controller** {**t1** | **e1**} *slot/port*
4. **ds0-group** *ds0-group-number* **timeslots** *timeslot-list* **type** {**e&m-delay-dial** | **e&m-fgd** | **e&m-immediate-start** | **e&m-wink-start** | **fxo-ground-start** | **fxo-loop-start** | **fxs-ground-start** | **fxs-loop-start**}
5. **exit**
6. **voice-port** *slot/port*
7. **operation {2-wire | 4-wire}**
8. **type** {**1** | **2** | **3** | **5**}
9. **signal {loop-start | ground-start}**

   or

   signal {**wink-start** | **immediate** | **delay-dial**}
10. **exit**
11. **connect** *connection-name* **voice-port** *voice-port-number* {**t1** | **e1**} *controller-number ds0-group-number*
12. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**`controller {t1 | e1}`** *`slot/port`* **Example:** `Router(config)#` **`controller t1 1/0`**	Enters controller configuration mode and identifies the controller type (T1 or E1) and a slot and port for configuration commands that specifically apply to the T1 or E1 interface. • Valid values for the *slot* and *port* arguments are 0 and 1.
Step 4	**`ds0-group`** *`ds0-group-number`* **`timeslots`** *`timeslot-list`* **`type {e&m-delay-dial | e&m-fgd | e&m-immediate-start | e&m-wink-start | fxs-ground-start | fxs-loop-start | fxo-ground-start | fxo-loop-start}`** **Example:** `Router(config-controller)#` **`ds0-group 1 timeslots 1 type e&m-wink-start`**	Defines the T1 or E1 channels for use by compressed voice calls and the signaling method the router uses to connect to the PBX or central office (CO). • The **ds0-group** command automatically creates a logical voice port. • *ds0-group-number*—Value from 0 to 23 that identifies the DS0 group. • *timeslot-list*—Single number, numbers separated by commas, or a pair of numbers separated by a hyphen to indicate a range of time slots. For T1, allowable values are 1 to 24; for E1, allowable values are 1 to 31. The signaling method selection for **type** depends on the connection that you are making: • Ear and Mouth (E&M) connects PBX trunk lines (tie lines) and telephone equipment. The wink and delay settings both specify confirming signals between the sending and receiving ends, or the immediate setting stipulates no special off-hook/on-hook signal. • **FXO** connects a CO to a standard PBX interface where permitted by local regulations. • **FXS** connects basic telephone equipment and PBXs.
Step 5	`exit` **Example:** `Router(config-controller)#` **`exit`**	Exits controller configuration mode and returns to global configuration mode.
Step 6	**`voice-port`** *`slot/port`* **Example:** `Router(config)#` **`voice-port 2/1`**	Enters voice-port configuration mode and identifies a slot and port for configuration parameters.

	Command or Action	Purpose
Step 7	**operation {2-wire | 4-wire}** **Example:** `Router(config-voiceport)# operation 4-wire`	Selects a specific cabling scheme for E&M ports: • This command is not applicable to FXS or FXO interfaces because they are, by definition, 2-wire interfaces. • Using this command on a voice port changes the operation of both voice ports on a VPM card. The voice port must be shut down and then opened again for the new value to take effect.
Step 8	**type {1 | 2 | 3 | 5}** **Example:** `Router(config-voiceport)# type 2`	Specifies the E&M interface type.

Step 9	**signal {loop-start | ground-start}**
	or
	signal {wink-start | immediate | delay-dial}
	
	Example:
	`Router(config-voiceport)# signal loop-start`
	or
	`Router(config-voiceport)# signal wink-start`
		Defines the signal type to be used.

Step 10	**exit** **Example:** `Router(config-voiceport)#` **`exit`**	Exits voice-port configuration mode and returns to global configuration mode.
Step 11	**connect** *connection-name* **voice-port** *voice-port-number* {**t1** | **e1**} *controller-number ds0-group-number* **Example:** `Router(config)#` **`connect connect1 voice-port 1/1/0 t1 1/0 0`**	Creates a named connection between two voice ports associated with T1 or E1 interfaces where you have already defined the groups by using the **ds0-group** command.
Step 12	**exit** **Example:** `Router(config)#` **`exit`**	Exits the current configuration session and returns to privileged EXEC mode.

Configuring Auto Cut-Through

The **auto-cut-through** command allows you to connect to PBXs that do not provide an M-lead response. To configure auto-cut-through, complete the following task:

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice-port** *slot/port*
4. **auto-cut-through**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot/port`* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**`auto-cut-through`** **Example:** `Router(config-voiceport)# auto-cut-through`	(E&M only) Enables call completion on a router if a PBX does not provide an M-lead response.
Step 5	**`exit`** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Modifying Bit Patterns 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.

To manipulate bit patterns to match particular E&M schemes, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port*
4. **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**}

5. **define** {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**}
6. **ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**}
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot/port`* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**`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`**} **Example:** `Router(config-voiceport)# condition tx-a-bit on`	Manipulates sent or received bit patterns to match expected patterns on a connected device. Repeat the command for each transmit 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). **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.

	Command	Purpose
Step 5	**define** {**tx-bits** | **rx-bits**} {**seize** | **idle**} {**0000** | **0001** | **0010** | **0011** | **0100** | **0101** | **0110** | **0111** | **1000** | **1001** | **1010** | **1011** | **1100** | **1101** | **1110** | **1111**} **Example:** `Router(config-voiceport)# define tx-bits seize 0000`	(Digital E1 E&M voice ports on Cisco 2600 and Cisco 3600 series routers 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 European Conference of Posts and Telecommunication Administration (CEPT) standards.
Step 6	**ignore** {**rx-a-bit** | **rx-b-bit** | **rx-c-bit** | **rx-d-bit**} **Example:** `Router(config-voiceport)# ignore rx-a-bit`	(Digital E1 E&M voice ports on Cisco 2600 and Cisco 3600 series routers 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 *Cisco IOS Voice Command Reference* for the default signaling patterns as defined in ANSI and CEPT standards.
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Configuring ANI for Outbound Calling

On the Cisco AS5300 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 a Signaling System 7 (SS7) 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 configuration 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, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice-port** *slot/port*
4. **calling-number outbound range** *string1 string2*
5. **calling-number outbound sequence** [*string1*] [*string2*] [*string3*] [*string4*] [*string5*]
6. **calling-number outbound null**
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot/port`* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**`calling-number outbound range`** *`string1 string2`* **Example:** `Router(config-voiceport)# calling-number outbound range 3000 4000`	(Cisco AS5300 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 fewer than four digits in length, then entire strings are used.
Step 5	**`calling-number outbound sequence`** `[`*`string1`*`] [`*`string2`*`] [`*`string3`*`] [`*`string4`*`] [`*`string5`*`]` **Example:** `Router(config-voiceport)# calling-number outbound sequence 2000 3000 4000`	(Cisco AS5300 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 strings. All strings must be valid E.164 numbers, up to 32 digits in length.

	Command	Purpose
Step 6	**calling-number outbound null** **Example:** `Router(config-voiceport)# calling-number outbound null`	(Cisco AS5300 only) Suppresses ANI. No ANI is passed when this voice port is selected.
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Configuring Disconnect Supervision

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 milliseconds) 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 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 "Configuring FXO Supervisory Disconnect Tones" section on page 12.

To change parameters related to disconnect supervision, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port*
4. **no battery-reversal**
5. **no supervisory disconnect**

6. **disconnect-ack**
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot/port`* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**`no battery-reversal`** **Example:** `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. **Note** This functionality is supported 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 6).
Step 5	**`no supervisory disconnect`** **Example:** `Router(config-voiceport)# no supervisory disconnect`	(FXO only) Enables the PBX or PSTN switch to provide STD. The **supervisory disconnect** command is enabled by default.

	Command	Purpose

Step 6	`disconnect-ack`
	
	Example:
	`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.

Step 7	`exit`
	
	Example:
	`Router(config-voiceport)# exit`
		Exits voice-port configuration mode and completes the configuration.

Configuring FXO Supervisory Disconnect Tones

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.

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 because of nonsupervisory tones or noise detection.

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:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class dualtone** *tag*
4. **freq-pair** *tone-id frequency-1 frequency-2*
5. **freq-max-deviation** *hertz*

6. **freq-max-power** *dBmO*
7. **freq-min-power** *dBmO*
8. **freq-power-twist** *dBmO*
9. **freq-max-delay** *milliseconds*
10. **cadence-min-on-time** *milliseconds*
11. **cadence-max-off-time** *milliseconds*
12. **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*]
13. **cadence-variation** *milliseconds*
14. **exit**
15. **voice-port** *slot/subunit/port*
16. **supervisory disconnect dualtone** {**mid-call** | **pre-connect**} **voice-class** *tag*
17. **supervisory disconnect anytone**
18. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice class dualtone`** *`tag`* **Example:** `Router(config)# voice class dualtone 1`	Enters voice-class configuration mode and creates a voice class for defining one tone detection pattern. Range is 1 to 10000. The tag number must be unique on the router. • For more information about configuring voice classes, refer to the *Dial Peer Configuration onVoice Gateway Routers*.
Step 4	**`freq-pair`** *`tone-id frequency-1 frequency-2`* **Example:** `Router(config-voice-class)# freq-pair 16 300 0`	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*. **Note** Repeat this command for each additional tone to be specified.
Step 5	**`freq-max-deviation`** *`hertz`* **Example:** `Router(config-voice-class)# freq-max-deviation 10`	Specifies the maximum frequency deviation that will be detected, in Hz. Range is 10 to 125. Default is 10.

	Command	Purpose
Step 6	**freq-max-power** *dBmO* **Example:** `Router(config-voice-class)# freq-max-power 20`	Specifies the maximum tone power that will be detected, in dBmO. Range is 0 to 20. Default is 10.
Step 7	**freq-min-power** *dBmO* **Example:** `Router(config-voice-class)# freq-min-power 35`	Specifies the minimum tone power that will be detected, in dBmO. Range is 10 to 35. Default is 30.
Step 8	**freq-power-twist** *dBmO* **Example:** `Router(config-voice-class)# freq-power-twist 15`	Specifies the power difference allowed between the two frequencies, in dBmO. Range is 0 to 15. Default is 6.
Step 9	**freq-max-delay** *time* **Example:** `Router(config-voice-class)# freq-max-delay 10`	Specifies the timing difference allowed between the two frequencies, in 10-millisecond increments. Range is 10 to 100 (100 ms to 1 second). Default is 20 (200 ms).
Step 10	**cadence-min-on-time** *time* **Example:** `Router(config-voice-class)# cadence-min-on-time 10`	Specifies the minimum tone on time that will be detected, in 10-millisecond increments. Range is 0 to 100 (0 ms to 1 second).
Step 11	**cadence-max-off-time** *time* **Example:** `Router(config-voice-class)# cadence-max-off-time 2000`	Specifies the maximum tone off time that will be detected, in 10-millisecond increments. Range is 0 to 5000 (0 ms to 50 seconds).
Step 12	**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*] **Example:** `Router(config-voice-class)# cadence-list 1 0 1000`	(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*—Range is 1 to 10. There is no default. • *cycle-N-on-time*—Range is 0 to 1000 (0 ms to 10 seconds). Default is 0. • *cycle-N-off-time*—Range is 0 to 1000 (0 ms to 10 seconds). Default is 0.
Step 13	**cadence-variation** *time* **Example:** `Router(config-voice-class)# cadence-variation 200`	(Optional) Specifies the maximum time that the tone onset can vary from the specified onset time and still be detected, in 10-millisecond increments. Range is 0 to 200 (0 ms to 2 seconds). Default is 0.
Step 14	**exit** **Example:** `Router(config-voice-class)# exit`	Exits voice class configuration mode.

	Command	Purpose
Step 15	**voice-port** *slot/subunit/port* **Example:** `Router(config)# voice-port 0/1/0`	Enters voice-port configuration mode.
Step 16	**supervisory disconnect dualtone** {**mid-call** | **pre-connect**} **voice-class** *tag* **Example:** `Router(config-voiceport)# supervisory disconnect dualtone mid-call voice-class 1`	Assigns an FXO supervisory disconnect tone voice class to the voice port.
Step 17	**supervisory disconnect anytone** **Example:** `Router(config-voiceport)# supervisory disconnect anytone`	Configures the voice port to disconnect on receipt of any tone.
Step 18	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Configuring Timeouts Parameters

To change timeouts parameters, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port*
4. **timeouts call-disconnect** *seconds*
5. **timeouts initial** *seconds*
6. **timeouts interdigit** *seconds*
7. **timeouts ringing** {*seconds* | **infinity**}
8. **timeouts wait-release** {*seconds* | **infinity**}
9. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot/port* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**timeouts call-disconnect** *seconds* **Example:** `Router(config-voiceport)# timeouts call-disconnect 60`	Configures the call disconnect timeout value in seconds. Range is 0 to 120. Default is 60.
Step 5	**timeouts initial** *seconds* **Example:** `Router(config-voiceport)# timeouts initial 10`	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. Range is 0 to 120. Default is 10.
Step 6	**timeouts interdigit** *seconds* **Example:** `Router(config-voiceport)# timeouts interdigit 10`	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 you are using variable-length dial peer destination patterns (dial plans). • The *seconds* argument is the interdigit timeout wait time in seconds. Range is 0 to 120. Default is 10.
Step 7	**timeouts ringing** {*seconds* | **infinity**} **Example:** `Router(config-voiceport)# timeouts ringing infinity`	Specifies the duration that the voice port allows ringing to continue if a call is not answered. • Default for *seconds* is 180.

	Command	Purpose
Step 8	**timeouts wait-release** {*seconds* | **infinity**} **Example:** `Router(config-voiceport)# timeouts wait-release 30`	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. • Default for *seconds* is 30.
Step 9	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Changing Timing Parameters

To change timing parameters, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port*
4. **timing clear-wait** *milliseconds*
5. **timing delay-duration** *milliseconds*
6. **timing delay-start** *milliseconds*
7. **timing delay-with-integrity** *milliseconds*
8. **timing dial-pulse min-delay** *milliseconds*
9. **timing dialout-delay** *millisecond*
10. **timing digit** *milliseconds*
11. **timing guard-out** *milliseconds*
12. **timing hookflash-out** *milliseconds*
13. **timing interdigit** *milliseconds*
14. **timing percentbreak** *percent*
15. **timing pulse** *pulses-per-second*
16. **timing pulse-digit** *milliseconds*
17. **timing pulse-interdigit** *milliseconds*
18. **timing wink-duration** *milliseconds*
19. **timing wink-wait** *milliseconds*
20. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot/port* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**timing clear-wait** *milliseconds* **Example:** `Router(config-voiceport)# timing clear-wait 200`	(E&M only) Specifies the minimum amount of time, in milliseconds, between the inactive seizure signal and clearing of the call. • Range is 200 to 2000. Default is 400.
Step 5	**timing delay-duration** *milliseconds* **Example:** `Router(config-voiceport)# timing delay-duration 100`	(E&M only) Specifies the delay signal duration for delay-dial signaling, in milliseconds. • Range is 100 to 5000. Default is 2000.
Step 6	**timing delay-start** *milliseconds* **Example:** `Router(config-voiceport)# timing delay-start milliseconds`	(E&M only) Specifies minimum delay time, in milliseconds, from outgoing seizure to outdial address. • Range is 20 to 2000. Default is 300.
Step 7	**timing delay-with-integrity** *milliseconds* **Example:** `Router(config-voiceport)# timing delay-with-integrity 0`	(Cisco MC3810 E&M ports only) Specifies duration of the wink pulse for the delay dial, in milliseconds. • Range is 0 to 5000. Default is 0.
Step 8	**timing dial-pulse min-delay** *milliseconds* **Example:** `Router(config-voiceport)# timing dial-pulse min-delay 300`	Specifies time, in milliseconds, between the generation of wink-like pulses when the type is pulse. • Range is 0 to 5000. Default is 300 for Cisco 3600 series and 140 for Cisco MC3810.
Step 9	**timing dialout-delay** *milliseconds* **Example:** `Router(config-voiceport)# timing dialout-delay 100`	(Cisco MC3810 only) Specifies dial-out delay, in milliseconds, for the sending digit or cut-through on an FXO trunk or an E&M immediate trunk. • Range is 100 to 5000. Default is 300.

	Command	Purpose
Step 10	**timing digit** *milliseconds* **Example:** `Router(config-voiceport)# timing digit 50`	Specifies the DTMF digit signal duration in milliseconds. • Range is 50 to 100. Default is 100.
Step 11	**timing guard-out** *milliseconds* **Example:** `Router(config-voiceport)# timing guard-out 300`	(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. • Range is 300 to 3000. Default is 2000.
Step 12	**timing hookflash-out** *milliseconds* **Example:** `Router(config-voiceport)# timing hookflash-out 500`	Specifies the duration, in milliseconds, of the hookflash. • Range is 50 to 500. Default is 300.
Step 13	**timing interdigit** *milliseconds* **Example:** `Router(config-voiceport)# timing interdigit 100`	Specifies the dual-tone multifrequency (DTMF) interdigit duration, in milliseconds. • Range is 50 to 500. Default is 100.
Step 14	**timing percentbreak** *percent* **Example:** `Router(config-voiceport)# timing percentbreak 20`	(Cisco MC3810 FXO and E&M ports only) Specifies the percentage of the break period for the dialing pulses, if different from the default. • Range is 20 to 80. Default is 50.
Step 15	**timing pulse** *pulses-per-second* **Example:** `Router(config-voiceport)# timing pulse 20`	(FXO and E&M only) Specifies the pulse dialing rate in pulses per second. • Range is 10 to 20. Default is 20.
Step 16	**timing pulse-digit** *milliseconds* **Example:** `Router(config-voiceport)# timing pulse-digit 10`	(FXO only) Configures the pulse digit signal duration. • Range is 10 to 20. Default is 20.
Step 17	**timing pulse-interdigit** *milliseconds* **Example:** `Router(config-voiceport)# timing pulse-interdigit 500`	(FXO and E&M only) Specifies pulse dialing interdigit timing in milliseconds. • Range is 100 to 1000. Default is 500.
Step 18	**timing wink-duration** *milliseconds* **Example:** `Router(config-voiceport)# timing wink-duration 200`	(E&M only) Specifies maximum wink-signal duration, in milliseconds, for a wink-start signal. • Range is 100 to 400. Default is 200.

	Command	Purpose
Step 19	**timing wink-wait** *milliseconds* **Example:** `Router(config-voiceport)# timing wink-wait 200`	(E&M only) Specifies maximum wink-wait duration, in milliseconds, for a wink-start signal. • Range is 100 to 5000. Default is 200.
Step 20	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Configuring the DTMF Timer

To configure the DTMF timer, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **controller T1** *number*
4. **ds0-group** *channel-number* **timeslots** *range* **type** *signaling-type* **dtmf dnis**
5. **cas-custom** *channel*
6. **dtmf timer-inter-digit** *milliseconds*
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**controller T1** *number* **Example:** `Router(config)# controller T1 1`	Configures a T1 controller and enters controller configuration mode.

	Command	Purpose
Step 4	**ds0-group** *channel-number* **timeslots** *range* **type** *signaling-type* **dtmf dnis** **Example:** `Router(config-controller)# ds0-group 0 timeslots 1-4 type e&m-immediate-start dtmf dnis`	Configures channelized T1 time slots, which enables a Cisco AS5300 modem to answer and send an analog call.
Step 5	**cas-custom** *channel* **Example:** `Router(config-controller)# cas-custom 2`	Enters cas-controller configuration mode and customizes signaling parameters for a particular E1 or T1 channel group on a channelized line.
Step 6	**dtmf timer-inter-digit** *milliseconds* **Example:** `Router(conf-ctrl-cas)# dtmf timer-inter-digit 100`	Configures the DTMF interdigit timer for a DS0 group.
Step 7	**exit** **Example:** `Router(conf-ctrl-cas)# exit`	Exits cas-controller configuration mode and completes the configuration.

Configuring Comfort Noise and Music Threshold for VAD

In normal voice conversations, only one person speaks at a time. 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 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 stop segmenting voice packets into frames. 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 in dial peers; by default it is enabled. 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 music threshold and comfort noise:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **vad** [**aggressive**]
5. **exit**
6. **voice vad-time** *milliseconds*
7. **voice-port** *slot/port*
8. **music-threshold** *number*
9. **comfort-noise**
10. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice`** *`tag`* **`voip`** **Example:** `Router(config)# dial-peer voice 555 voip`	Enters dial-peer configuration mode.
Step 4	**`vad`** `[`**`aggressive`**`]` **Example:** `Router(config-dial-peer)# vad`	Enables VAD for calls using this dial peer. **Note** VAD is enabled by default. Use the **vad** command only if you have previously disabled the feature by using the **no vad** command.
Step 5	**`exit`** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 6	**`voice vad-time`** *`milliseconds`* **Example:** `Router(config)# voice vad-time 500`	Modifies the minimum silence detection time for VAD.

	Command	Purpose
Step 7	**voice-port** *slot/port* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode. **Note** The syntax of this command is platform-specific. For information, refer to the *Cisco IOS Voice Command Reference*.
Step 8	**music-threshold** *number* **Example:** `Router(config-voiceport)# music-threshold -70`	Specifies the minimal decibel level of music played when calls are put on hold. The decibel level affects how VAD treats the music data. • Valid values range from –70 to –30. If the music threshold is set too high and VAD is configured, the remote end hears no music; if the level is set too low, there is unnecessary voice traffic. Default is –38.
Step 9	**comfort-noise** **Example:** `Router(config-voiceport)# comfort-noise`	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. • Comfort noise is enabled by default.
Step 10	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.

PSTN Fallback

The PSTN Fallback feature monitors congestion in the IP network and redirects calls to the Public Switched Telephone Network (PSTN) or rejects calls on the basis of network congestion. This feature can also use the ICMP ping mechanism to detect loss of network connectivity and then reroute calls. The fallback subsystem has a network traffic cache that maintains the Calculated Planning Impairment Factor (ICPIF) or delay/loss values for various destinations. Performance is improved because each new call to a well-known destination does not have to wait on a probe to be admitted and the value is usually cached from a previous call.

ICPIF calculates an impairment factor for every piece of equipment along the voice path and then adds them up to get the total impairment value. Refer to International Telecommunication Union (ITU) standard G.113 for more information. The ITU assigns a value to the types of impairment, such as noise, delay, and echo.

Feature Information

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Contents

This document contains the following sections:

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Information About PSTN Fallback

This section provides the following information about the PSTN Fallback feature:

- Service Assurance Agent
- Application of PSTN Fallback

Service Assurance Agent

Service Assurance Agent (SAA) is a network congestion analysis mechanism that provides delay, jitter, and packet loss information for the configured IP addresses. SAA is based on a client/server protocol defined on the User Datagram Protocol (UDP). UDP is a connectionless transport layer protocol in the IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. The SAA probe packets go out on randomly selected ports from the top end of the audio UDP port range.

The information that the SAA probes gather is used to calculate the ICPIF or delay/loss values that are stored in a fallback cache, where they remain until the cache ages out or overflows. Until an entry ages out, probes are sent periodically for that particular destination. This time interval is user configurable.

With this feature enhancement, you can also configure codes that indicate the cause of the network rejection; for example, packets that are lost or that take too long to be transmitted. A default cause code of 49 displays the message **qos-unavail**, which means Quality of Service is unavailable.

Note The Cisco SAA functionality in Cisco IOS software was formerly known as Response Time Reporter (RTR). In the "How to Configure PSTN Fallback" section, note that the command-line interface still uses the keyword **rtr** for configuring RTR probes, which are now actually the SAA probes.

Application of PSTN Fallback

The PSTN Fallback feature and enhancement provide the following benefits:

- Automatically re-routes calls when the data network is congested at the time of the call setup.
- Enables the service provider to give a reasonable guarantee about the quality of the conversation to its Voice over IP (VoIP) users at the time of call admission.
- Provides delay, jitter, and packet loss information for the configured IP addresses.
- Caches call values from previous calls. New calls do not have to wait for probe results before they are admitted.
- Enables a user-configurable cause code display that indicates the type of call rejection.

Restrictions for PSTN Fallback

The PSTN Fallback feature has the following restrictions:

- When detecting network congestion, the PSTN fallback feature does nothing to the existing call. It affects only subsequent calls.
- Only a single ICPIF/delay-loss value is allowed per system.

- A small additional call setup delay can be expected for the first call to a new IP destination.

Caution Configuring **call fallback active** in a gateway creates an SAA jitter probe against other (target) gateways to which the calls are sent. In order for the call fallback active to work properly, the target gateways must have the **rtr responder** command (in Cisco IOS releases prior to 12.3(14)T) or the **ip sla monitor responder** command (in Cisco IOS Release 12.3(14)T or later) in their configurations. If one of these commands is not included in the configuration of each target gateway, calls to the target gateway will fail.

How to Configure PSTN Fallback

This section contains the following procedures (each identified as either optional or required):

- Configuring Call Fallback to Use MD5 Authentication for SAA Probes (required)
- Configuring Destination Monitoring without Fallback to Alternate Dial Peers (optional)
- Configuring Call Fallback Cache Parameters (optional)
- Configuring Call Fallback Jitter-Probe Parameters (optional)
- Configuring Call Fallback Probe-Timeout and Weight Parameters (optional)
- Configuring Call Fallback Threshold Parameters (optional)
- Configuring Call Fallback Wait-Timeout (optional)
- Configuring VoIP Alternate Path Fallback SNMP Trap (optional)
- Configuring ICMP Pings to Monitor IP Destinations (optional)

Configuring Call Fallback to Use MD5 Authentication for SAA Probes

To configure call fallback to use MD5 authentication for SAA probes, use the following commands.

SUMMARY STEPS

1. **enable**
2. **config terminal**
3. **call fallback active**
4. **call fallback key-chain** *name-of-chain*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback active** **Example:** `Router(config)# call fallback active`	Enables the PSTN fallback feature to alternate dial peers in case of network congestion.
Step 4	**call fallback key-chain** *name-of-chain* **Example:** `Router(config)# call fallback key-chain sample`	Specifies the use of message digest algorithm 5 (MD5) authentication for sending and receiving Service Assurance Agents (SAA) probes.

Configuring Destination Monitoring without Fallback to Alternate Dial Peers

To configure destination monitoring without fallback to alternate dial peers, use the following commands.

SUMMARY STEPS

1. **enable**
2. **config terminal**
3. **call fallback monitor**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback monitor** **Example:** `Router(config)# call fallback monitor`	Enables the monitoring of destinations without fallback to alternate dial peers.

Configuring Call Fallback Cache Parameters

To configure the call fallback cache parameters, use the following commands.

SUMMARY STEPS

1. **enable**
2. **config terminal**
3. **call fallback cache-size** *number*
4. **call fallback cache-timeout** *seconds*
5. **clear call fallback cache** [*ip-address*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback cache-size** *number* **Example:** `Router(config)# call fallback cache-size 5`	Specifies the call fallback cache size.
Step 4	**call fallback cache-timeout** *seconds* **Example:** `Router(config)# call fallback cache-timeout 300`	Specifies the time after which the cache entry is purged, in seconds. Default: 600.
Step 5	**clear call fallback cache** [*ip-address*] **Example:** `Router(config)# clear call fallback cache 10.1.1.1`	Clears the current ICPIF estimates for all IP addresses or a specific IP address in the cache.

Configuring Call Fallback Jitter-Probe Parameters

To configure call fallback jitter-probe parameters, use the following commands.

SUMMARY STEPS

1. **enable**
2. **config terminal**
3. **call fallback jitter-probe num-packets** *number-of-packets*
4. **call fallback jitter-probe precedence** *precedence*
   or
   call fallback jitter-probe dscp *dscp-number*
5. **call fallback jitter-probe priority-queue**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback jitter-probe num-packets** *number-of-packets* **Example:** `Router(config)# call fallback jitter-probe num-packets 10`	Specifies the number of packets for jitter. Default: 15.

Step 4	**call fallback jitter-probe precedence** *precedence*
	or
	call fallback jitter-probe dscp *dscp-number*
	
	Example:
	`Router(config)# call fallback jitter-probe precedence 2`
	or
	`Router(config)# call fallback jitter-probe dscp 2`
		Specifies the treatment of the jitter-probe transmission. Default: 2.
		Specifies the differentiated services code point (dscp) packet of the jitter-probe transmission.
		Note The **call fallback jitter-probe precedence** command is mutually exclusive with the **call fallback jitter-probe dscp** command. Only one of these command can be enabled on the router. Usually, the **call fallback jitter-probe precedence** command is enabled.
		When the **call fallback jitter-probe dscp** command is configured, the precedence value is replaced by the DSCP value. To disable DSCP and restore the default jitter probe precedence value, use the **no call fallback jitter-probe dscp** command.

Step 5	**call fallback jitter-probe priority-queue** **Example:** `Router(config)# call fallback jitter-probe priority-queue`	Assigns a priority to the queue for jitter probes.

Configuring Call Fallback Probe-Timeout and Weight Parameters

To configure call fallback probe-timeout and weight parameters, use the following commands.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback probe-timeout** *seconds*

4. **call fallback instantaneous-value-weight** *weight*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`call fallback probe-timeout`** *`seconds`* **Example:** `Router(config)# call fallback probe-timeout 20`	Sets the timeout for an SAA probe, in seconds. Default: 30.
Step 4	**`call fallback instantaneous-value-weight`** *`percent`* **Example:** `Router(config)# call fallback instantaneous-value-weight 50`	Configures the call fallback subsystem to take an average from the last two probes registered in the cache for call requests: • *percent*—Instantaneous value weight, expressed as a percentage. Range: 0 to 100. Default: 66.

Configuring Call Fallback Threshold Parameters

To configure call fallback threshold parameters, use the following commands.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback threshold delay** *delay-value* **loss** *loss-value*
   or
   call fallback threshold icpif *threshold-value*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**call fallback threshold delay** *delay-value* **loss** *loss-value*
	or
	call fallback threshold icpif *threshold-value*
	
	Example:
	`Router(config)# call fallback threshold delay 100 loss 150`
	or
	`Router(config)# call fallback threshold icpif 100`
		Specifies fallback threshold to use packet delay and loss values. No defaults.
		Note The amount of delay set by the **call fallback threshold delay loss** command should not be more than half the amount of the time-to-wait value set by the **call fallback wait-timeout** command; otherwise the threshold delay will not work correctly. Because the default value of the **call fallback wait-timeout** command is set to 300 milliseconds, you can configure a delay of up to 150 milliseconds for the **call fallback threshold delay loss** command. If you want to configure a higher threshold, the time-to-wait delay has to be increased from its default (300 milliseconds) using the **call fallback wait-timeout** command.
		
		Specifies fallback threshold to use the Calculated Planning Impairment Factor (ICPIF) threshold for network traffic.

Configuring Call Fallback Wait-Timeout

To configure the call fallback wait-timeout parameters, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback wait-timeout** *milliseconds*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`call fallback wait-timeout`** *`milliseconds`* **Example:** `Router(config)# call fallback wait-timeout 200`	Configures the waiting timeout interval for a response to a probe in milliseconds. Default: 300 milliseconds. **Note** The time-to-wait period set by the **call fallback wait-timeout** command should always be greater than or equal to twice the amount of the threshold delay time set by the **call fallback threshold delay loss** command; otherwise the probe will fail. The delay configured by the **call fallback threshold delay loss** command corresponds to a one-way delay, whereas the time-to-wait period configured by the **call fallback wait-timeout** command corresponds to a round-trip delay. The threshold delay time should be set at half the value of the time-to-wait value.

Configuring VoIP Alternate Path Fallback SNMP Trap

The VoIP Alternate Path Fallback SNMP Trap feature adds a Simple Network Management Protocol (SNMP) trap generation capability. This feature is built on top of the fallback subsystem to provide an SNMP notification trap when the fallback subsystem redirects or rejects a call because a network condition has failed to meet the configured threshold. The SNMP trap provides VoIP management status MIB information without flooding management systems with unnecessary messages about call status by triggering only when a call has been redirected to the public switched telephone network (PSTN) or the alternative IP port. A call can be rejected because of a network problem such as loss of WAN connection, delay, packet loss, or jitter. This feature supports only VoIP signaling protocol with H.323 in this release.

This feature has to be configured on the originating gateway and the terminating gateway. To configure the SNMP trap parameters, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback active**
4. **snmp-server enable traps voice fallback**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`call fallback active`** **Example:** `Router(config)# call fallback active`	Enables the PSTN fallback feature to alternate dial peers in case of network congestion.
Step 4	**`snmp-server enable traps voice fallback`** **Example:** `Router(config)# snmp-server enable traps voice fallback`	Configures the SNMP trap parameters.

What to Do Next

Configure the **rtr responder** command on the terminating voice gateway. If the **rtr responder** is enabled on the terminating gateway, the terminating gateway responds to the probe request when the originating gateway sends an Response Time Report (RTR) probe to the terminating gateway to check the network conditions.

Configuring Call Fallback Map Parameters

The **call fallback map** command option provides a target network summary/consolidation mode. For example, if there are four individual voice gateway routers connected together on a remote LAN via a separate LAN-to-WAN access router, the map option allows a single probe to be sent to the single remote WAN access router (instead of having to maintain separate probes for each of the four voice gateway routers' IP addresses). Because the remote access and voice gateway routers are connected together on the same remote LAN, the probes to the access router returns similar results to probes to the individual voice gateway routers.

To configure call fallback map parameters, use the following commands.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback map** *map* **target** *ip-address* **address-list** *ip-address1 ip-address2 … ip-address7*
   or
   call fallback map *map* **target** *ip-address subnet ip-network netmask*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`call fallback map`** *`map`* **`target`** *`ip-address`* **`address-list`** *`ip-address1 ip-address2 ... ip-address7`* or **`call fallback map`** *`map`* **`target`** *`ip-address subnet ip-network netmask`*	Specifies the call fallback router to keep a cache table (by IP addresses) of distances for several destination peers sitting behind the router. • *map*—Fallback map. Range is from 1 to 16. There is no default. • **target** *ip-address*—Target IP address. • *ip-address1 ip-address2* ... *ip-address7*—Lists the IP addresses that are kept in the cache table. The maximum number of IP addresses is seven. Specifies the call fallback router to keep a cache table (by subnet addresses) of distances for several destination peers sitting behind the router.

Configuring ICMP Pings to Monitor IP Destinations

This capability is enabled to monitor the IP destinations in a VoIP network, which may not support RTR. This monitoring is referred to as ICMP pinging. Based on the RTR or ICMP pinging, results change the operational state of the dial-peer. The configurations described in this section also provide support for monitoring the following session targets configured under a VoIP dial-peer:

- DNS
- IP version 4
- SIP-server
- enum

To configure call-fallback monitor probes to ping IP destinations, complete one of the following tasks:

- Dial Peer Configuration of the call fallback icmp-ping and monitor probe Commands
- Global Configuration of the call fallback icmp-ping Command
- Voice Port Configuration of the busyout monitor probe icmp-ping Command
- Voice Class Configuration of the busyout monitor probe icmp-ping Command

Dial Peer Configuration of the call fallback icmp-ping and monitor probe Commands

To configure dial-peer parameters to use ICMP pings to monitor IP destinations, complete this task. This configuration applies only to VoIP dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **call fallback** [**icmp-ping** | **rtr**]
5. **monitor probe** {**icmp-ping** | **rtr**} [*ip address*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`dial-peer voice tag voip`** **Example:** `Router(config)# dial-peer voice 10 voip`	Enters dial peer configuration mode, specifies the method of voice encapsulation, and defines a particular dial peer: *tag*—Digits that define a particular dial peer. Range is from 1 to 2147483647.

	Command or Action	Purpose
Step 4	**call fallback** [**icmp-ping** | **rtr**] **Example:** `Router(config-dial-peer)# call fallback icmp-ping`	Configures dial-peer parameters for pings to IP destinations: • **icmp-ping**—Uses ICMP pings to monitor the IP destinations. • **rtr**—Uses RTR probes to monitor the session target and update the status of the dial peer. RTR probes are the default. **Note** If this **call fallback icmp-ping** command is not entered, the **call fallback active** command in global configuration is used for measurements. If this **call fallback icmp-ping** command is entered, these values override the global configuration. One of these two commands must be in effect before the **monitor probe icmp-ping** command can be used. If neither of **call fallback** commands is in effect, the **monitor probe icmp-ping** command will not work properly.
Step 5	**monitor probe** {**icmp-ping** | **rtr**} [*ip address*] **Example:** `Router(config-dial-peer)# monitor probe icmp-ping`	Enables dial-peer status changes based on the result of the probe: • **icmp-ping**—Uses ICMP ping as the method for the probe. • **rtr**—Uses RTR as the method for the probe. *ip address*—IP address of the destination to be probed. If no IP address is specified, the IP address is read from the session target.

Global Configuration of the call fallback icmp-ping Command

To configure global parameters to use ICMP pings to monitor IP destinations, complete this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback active** [**icmp-ping** | **rtr**]
4. **call fallback icmp-ping** [**count** *number*] [**codec** *type*] | **size** *number*] **interval** *number* [**loss** *number*] [**timeout** *value*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`call fallback active`** `[`**`icmp-ping`** `|` **`rtr`**`]` **Example:** `Router(config)# call fallback active icmp-ping`	Configures global parameters for pings to IP destinations: • **icmp-ping**—Uses ICMP pings to monitor the IP destinations. • **rtr**—Uses RTR probes to monitor the IP destinations. RTR probes are the default. **Note** The **call fallback active icmp-ping** command must be entered before the **call fallback icmp-ping** command can be used. If you do not enter this command first, the **call fallback icmp ping** command will not work properly.
Step 4	**`call fallback icmp-ping`** `[`**`count`** *`number`*`] [`**`codec`** *`type`*`] |` **`size`** *`bytes`*`]` **`interval`** *`seconds`* `[`**`loss`** *`number`*`] [`**`timeout`** *`milliseconds`*`]` **Example:** `Router(config)# call fallback icmp ping codec g729 interval 10 loss 10`	Configures the parameters for ICMP pings: • **count**—Number of ping packets to be sent to the destination IP address. Default is 5. • **codec**—Codec type for deciding the ping packet size. • *type*—Acceptable codec types are **g711a**, **g711u**, **g729**, and **g729b**. • **size—**Size (in bytes) of the ping packet. Default is 32. • **interval**—Time (in seconds) between ping packet sets. Default is 5. This value should be more than the **timeout** value. • **loss—**Threshold packet loss, expressed as a percentage. Default is 20. • **timeout—**Timeout (in milliseconds) for the echo packets. Default is 500.

Voice Port Configuration of the busyout monitor probe icmp-ping Command

To configure voice-port parameters to use ICMP pings to monitor IP destinations, complete this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot/port*

4. **busyout monitor probe icmp-ping** *ip address* [**codec** *type* | **size** *bytes*] [**loss** *percent*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice-port`** *`slot/port`* **Example:** `Router(config)# voice-port 1/0`	Enters voice-port configuration mode and identifies the slot and port where the configuration parameters take effect. **Note** The syntax for this command varies by platform. For more information, see the *Cisco IOS Voice Command Reference*.
Step 4	**`busyout monitor probe icmp-ping`** *`ip address`* `[`**`codec`** *`type`* `|` **`size`** *`bytes`*`] [`**`loss`** *`percent`*`]` **Example:** `Router(config-voiceport)# busyout monitor probe 10.1.1.1 g711u loss 10 delay 2000`	Specifies the parameters for ICMP pings for monitoring under voice-port configuration: • *ip address*—IP address of the destination to which the ping is sent. • **codec**—(Optional) Codec type for deciding the ping packet size. • *type*—Acceptable codec types are **g711a**, **g711u**, **g729**, and **g729b**. • **size**—(Optional) Size (in bytes) of the ping packet. Default is 32. **loss**—(Optional) Threshold packet loss, expressed as a percentage. Default is 20.

Voice Class Configuration of the busyout monitor probe icmp-ping Command

To configure voice-class parameters to use ICMP pings to monitor IP destinations, complete this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class busyout** *tag*
4. **busyout monitor probe icmp-ping** *ip address* [**codec** *type* | **size** *bytes*] [**loss** *percent*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice class busyout** *tag* **Example:** `Router(config)# voice class busyout 10`	Creates a voice class for local voice busyout functions: *tag*—Unique identification number assigned to one voice class. Range is 1 to 10000.
Step 4	**busyout monitor probe icmp-ping** *ip address* [**codec** *type* | **size** *bytes*] [**loss** *percent*] **Example:** `Router(config-class)# busyout monitor probe icmp-ping 10.1.1.1 codec g729b size 32`	Configures the parameters for ICMP pings for monitoring under voice-port: • *ip address*—IP address of the destination to which the ping is sent. • **codec**—(Optional) Codec type for deciding the ping packet size. • *type*—Acceptable codec types are **g711a**, **g711u**, **g729**, and **g729b**. • **size**—(Optional) Size (in bytes) of the ping packet. Default is 32. • **loss**—(Optional) Threshold packet loss, expressed as a percentage. Default is 20.

How to Verify and Monitor the PSTN Fallback Feature

This section provides the following information:

- Verifying PSTN Fallback Configuration
- Monitoring and Maintaining PSTN Fallback

Verifying PSTN Fallback Configuration

The **show** commands in this section can be used to display statistics and configuration parameters to verify the operation of the PSTN Callback feature:

- **show running-config**—Displays the contents of the currently running configuration file to see if the new feature is configured.
- **show call history voice**—Displays the call history table for voice calls and verify call fallback, call delay, and call loss parameters.

- **show call fallback cache**—Displays the current Calculated Planning Impairment Factor (ICPIF) estimates for all IP addresses in the call fallback cache.
- **show call fallback config—**Displays the current configuration.
- **show call fallback stats**—Displays the call fallback statistics.

Monitoring and Maintaining PSTN Fallback

Use the following commands to monitor and maintain the PSTN Fallback feature:

- **clear call fallback cache**—Clears the current ICPIF estimates for all IP addresses in the cache.
- **clear call fallback stats**—Clears the call fallback statistics.
- **debug call fallback detail**—Displays details of VoIP call fallback.
- **debug call fallback probes—**Displays details of voice fallback probes.
- **test call fallback probe** *ip-address*—Tests a probe to a particular IP address and displays the ICPIF SAA values.
- **debug snmp packets**—Displays information about every Simple Network Management Protocol (SNMP) packet sent or received by the router.

What To Do Next

The "Configuring ICMP Pings to Monitor IP Destinations" describes the mechanism whereby a dial-peer becomes temporarily disabled because of poor SAA/RTR probe results (for example, ICPIF, jitter, or loss), or because of failure of the ICMP ping test. When this occurs, the normal alternate dial-peer selection process (hunting) is triggered to search for an alternate dial-peer that represents an alternate route.

The global configuration **voice hunt** command controls whether hunting (continue to look or "hunt" for an alternate dial-peer match) occurs, based on the specific cause code that describes why the initial dial-peer path failed. Hunting is usually appropriate if the cause code indicates network congestion, but usually inappropriate if the failure cause code indicates that the called user is actually busy. Even if an alternate path is taken to reach the called user, and if the user is actually busy, the user will be busy regardless of which path is used.

For more information about the **voice hunt** command, see the *Cisco IOS Voice Command Reference*.

Configuring Echo Cancellation

Echo cancellation is a key function in packet voice. Much of the perceived quality of the connection depends on the performance of the echo canceller. The G.168 extended echo cancellation (EC) provides an alternative to the proprietary Cisco G.165 EC with improved performance for trunking gateway applications.

The following sections provide configuration information for echo cancellation:

Information About Echo Cancellation

To use the   feature, you should understand the following concepts:

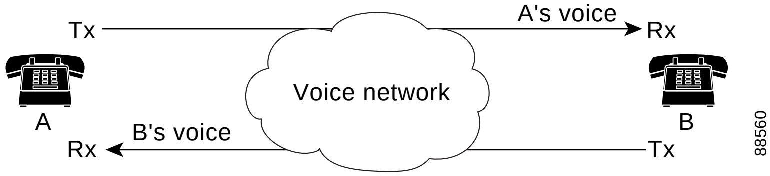
Voice Call Transmit and Receive Paths

Every voice conversation has at least two participants. From the perspective of each participant, there are two voice paths in every call:

- Transmit path (also called the send or Tx path)—The transmit path is created when a person speaks. The sound is transmitted from the mouth of the speaker to the ear of the listener.
- Receive path (also called the return or Rx path)—The receive path is created when a person hears the conversation. The sound is received by the ear of the listener from the mouth of the speaker.

Figure 1 shows a simple voice call between caller A and caller B. The top line represents the Tx path for caller A, which becomes the Rx path for caller B. The bottom line represents the Tx path for caller B, which becomes the Rx path for caller A.

Figure 1 Transmit and Receive Paths in a Voice Network



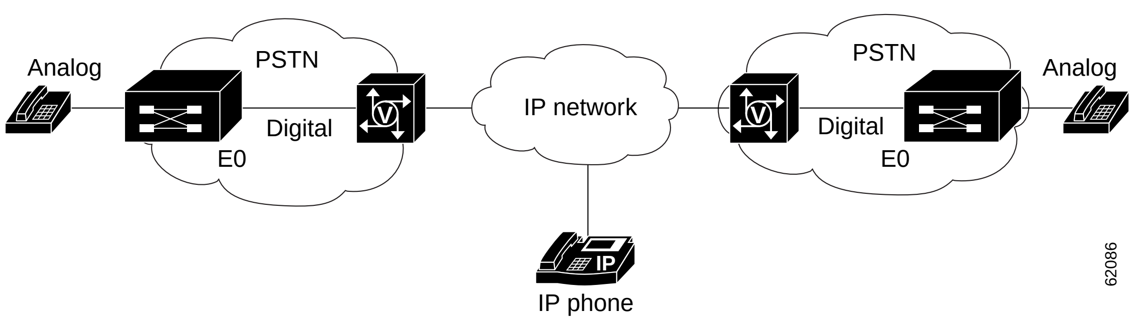
Echo Cancellation

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 (ms), it can be distracting to the speaker. 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. Echo is controlled by echo cancellers (ECs).

A packet voice gateway, which operates between a digital packet network and the PSTN, can include both digital (time division multiplexing [TDM]) and analog links. The analog circuit is known as the tail circuit. It forms the tail or termination of the call from the perspective of the person experiencing the echo. The tail circuit is everything connected to the PSTN side of a packet voice gateway—all the switches, multiplexers, cabling, and PBXs between the voice gateway and the telephone.

Figure 2 shows a common voice network where echo cancellation might be used.

Figure 2 Echo Cancellation Network



An echo canceller reduces the level of echoes that leak from the Rx path (from the gateway out into the tail circuit) into the Tx path (from the tail circuit into the gateway). From the perspective of the echo canceller in a voice gateway, the Rx signal is a voice coming across the network from another location. The Tx signal is a mixture of the voice call in the other location and the echo of the original voice, which comes from the tail circuit on the initiating end and is sent to the receiving end.

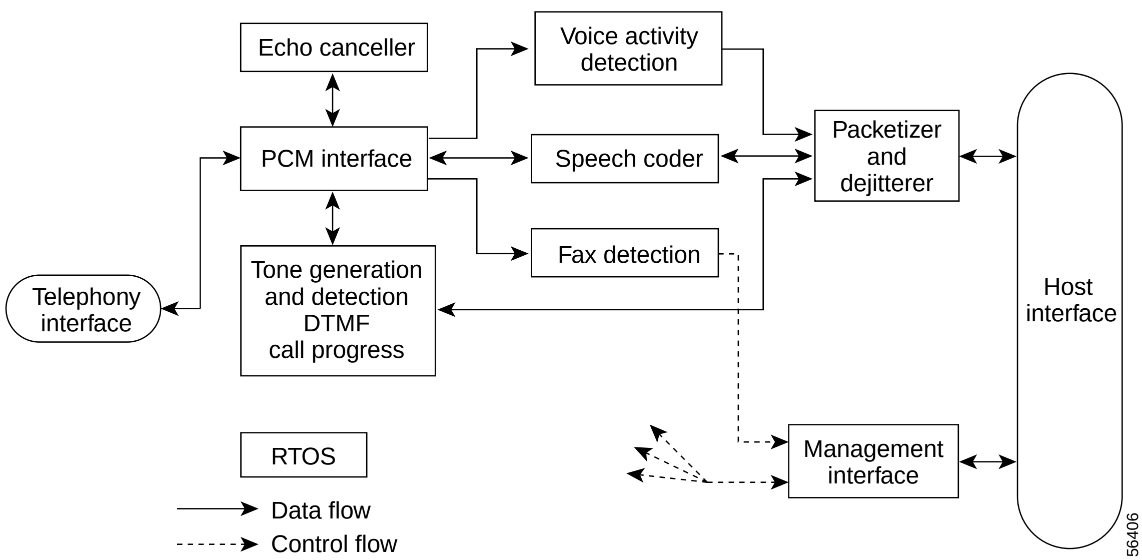
Echo cancellers face into the PSTN tail circuit. They eliminate echoes in the tail circuit on its side of the network.The echo canceller in the originating gateway looks out into the tail circuit and is responsible for eliminating the echo signal from the initiation Tx signal and allowing a voice call to go through unimpeded. By design, ECs are limited by the total amount of time they wait for the reflected speech to be received, which is known as an echo tail. The echo tail is normally 32 ms.

Note Delay and jitter in the WAN do not affect the operation of the echo canceller because the tail circuit, where the echo canceller operates, is static.

Echo cancellation is implemented in digital signal processor (DSP) firmware (DSPWare) on Cisco voice gateways and is independent of other functions implemented in the DSP (the DSP protocol and compression algorithm). In voice packet-based networks, ECs are built into the low-bit-rate codecs and are operated on each DSP.

Figure 3 shows a typical DSP channel configured for voice processing.

Figure 3 DSP Channel Configured for Voice Processing



Echo Canceller Operation

An echo canceller removes the echo portion of the signal coming out of the tail circuit and headed into the WAN. It does so by learning the electrical characteristics of the tail circuit and forming its own model of the tail circuit in its memory, and creating an estimated echo signal based on the current and past Rx signal. It subtracts the estimated echo from the actual Tx signal coming out of the tail circuit. The quality of the estimation is continuously improved by monitoring the estimation error.

Following are descriptions of the primary measurements of relative signal levels used by echo cancellers. They are all expressed in decibels (dB).

- Echo return loss (ERL)—Reduction in the echo level produced by the tail circuit without the use of an echo canceller. If an Rx speech signal enters the tail circuit from the network at a level of X dB, the echo coming back from the tail circuit into the echo canceller is X less ERL.

- Echo return loss enhancement (ERLE)—Additional reduction in echo level accomplished by the echo canceller. An echo canceller is not a perfect device; the best it can do is attenuate the level of the returning echo. ERLE is a measure of this echo attenuation. It is the difference between the echo level arriving from the tail circuit at the echo canceller and the level of the signal leaving the echo canceller.
- Acombined (ACOM)—Total ERL seen across the terminals of the echo canceller. ACOM is the sum of ERL + ERLE, or the total ERL seen by the network.

For more information about the echo canceller, refer to the *Echo Analysis for Voice over IP* document.

Echo Canceller Components

A typical echo canceller includes two components: convolution processor (CP) and a nonlinear processor (NLP).

Convolution Processor

The CP first stage captures and stores the outgoing signal toward the far-end hybrid. The CP then switches to monitoring mode and, when the echo signal returns, estimates the level of the incoming echo signal and subtracts the attenuated original voice signal from the echo signal.

The time required to adjust the level of attenuation needed in the original signal is called the convergence time. Because the convergence process requires that the voice signal be stored in memory, the EC has limited coverage of tail circuit delay, normally 64, 96, and up to 128 ms. After convergence, the CP provides about 18 dB of ERLE. Because a typical analog phone circuit provides at least 12 dB of ERL (that is, the echo path loss between the echo canceller and the far-end hybrid), the expected permanent ERL of the converged echo canceller is about 30 dB or greater.

Nonlinear Processor

In single-talk mode, that is, when one person is talking and the other is silent, the NLP replaces the residual echo at the output of the echo canceller with comfort noise based on the actual background noise of the voice path. The background noise normally changes over the course of a phone conversation, so the NLP must adapt over time. The NLP provides an additional loss of at least 25 dB when activated. In double-talk mode, the NLP must be deactivated because it would create a one-way voice effect by adding 25 to 30 dB of loss in only one direction.

To completely eliminate the perception of echo, the talker echo loudness rating (TELR) should be greater than 65 dB in all situations. To reflect this reality, ITU-T standard G.168 requires an ERL equal to or greater than 55 dB. Segmentation local reference (SLR), receive loudness rating (RLR), and cell loss ratio (CLR) along the echo path should allow another 10 dB to meet the expected TELR. CP, NLP and loudness ratings (LRs) must be optimized to make sure that echo is canceled effectively.

Echo Canceller Coverage

Echo canceller coverage (also known as tail coverage or tail length) is the length of time that the echo canceller stores its approximation of an echo in memory. It is the maximum echo delay that an echo canceller is able to eliminate.

The echo canceller faces into a static tail circuit with input and an output. If a word enters a tail circuit, the echo is a series of delayed and attenuated versions of that word, depending on the number of echo sources and the delays associated with them. After a certain period of time, no signal comes out. This time period is known as the ringing time of the tail circuit—the time required for all of the ripples to disperse. To fully eliminate all echoes, the coverage of the echo canceller must be as long as the ringing time of the tail circuit.

ITU-T Echo Cancellation History

ITU-T standard G.164 defines the performance of echo suppressors, which are the predecessors of echo cancellation technology. G.164 also defines the disabling of echo suppressors in the presence of 2100 Hz tones (which precede low bit rate modems).

ITU-T standard G.165 defines echo cancellation and provides a number of objective tests that ensure a minimum level of performance. These tests check convergence speed of the echo canceller, stability of the echo canceller filter, performance of the nonlinear processor, and a limited amount of double talk testing. The signal used to perform these tests is white noise. Additionally, G.165 defines the disabling of echo cancellers in the presence of 2100 Hz signals with periodic phase reversals in order to support echo cancelling modem technology (V.34, for example), which does not work if line echo cancellation is performed in the connection.

ITU-T standard G.168 allows more rigorous testing and satisfies more testing requirements. White noise is replaced with a pseudospeech signal for the convergence tests. Most echo cancellation algorithms use a least mean square (LMS) algorithm to adapt the echo cancellation filter. LMS works best with random signals, and slows down with more correlated signals such as speech. Using the pseudospeech signal in testing provides a more realistic portrayal of the echo cancellers performance in real use.

In Cisco IOS Release 12.3(4)XD and later releases, the G.168 EC is the default and you can no longer select the Cisco G.165 EC on any supported platform except the Cisco AS5300. The Cisco AS5300 still supports the Cisco G.165 EC and the extended G.168 EC. provides a summary of the Extended ITU-T G.168 Echo Cancellation feature availability in Cisco IOS releases.

Table 1 Feature History for Extended ITU-T G.168 Echo Cancellation

Release	Modification
12.2(13)T	This feature was introduced.
12.2(13)ZH	The extended G.168 EC became the default on the Cisco 1700 series and the Cisco ICS 7750.
12.2(15)ZJ	The extended G.168 EC became the default on the Cisco 2600 series, Cisco 3600 series, Cisco 3700 series. **Note** The extended G.168 EC is not supported on the High-Density Analog Network Modules (NM-HDA) and Asynchronous Interface Module (AIM)-Voice modules on the Cisco 2600 series in this release.
12.3(1)	The extended G.168 EC became the default on the Cisco IAD2420, Cisco MC3810, and Cisco VG200.
12.3(4)T	The extended G.168 EC became the default on the Cisco 7200 series and Cisco Catalyst 4000 AGM.

Table 1 Feature History for Extended ITU-T G.168 Echo Cancellation (continued)

12.3(4)XD	The G.168 extended EC became the only EC on all voice packet platforms that support the extended G.168 EC; the Cisco G.165 EC is no longer a selectable option. **Note** The Cisco AS5300 still supported choosing between the Cisco G.165 EC and the extended G.168 EC.
12.3(3)	The G.148 extended EC was configurable with no codec restriction on the Cisco AS5300.
12.3(8)XY	The G.168 extended EC was supported for the WS-SVC-CMM-6T1, WS-SVC-CMM-6E1, and WS-SVC-CMM-24FXS port adapters on the Cisco Communication Media Module (WS-SVC-CMM).
12.3(9)	The extended G.168 EC was supported for the NM-HDA and AIM-Voice modules on the Cisco 2600 series.
12.3(11)T	The dual-filter G.168 echo canceller capability was added to NextPort SPE firmware (SPEware) version 10.2.2 and later versions with Cisco IOS Release 12.3(11)T and later releases. See the chapter *NextPort-Based Voice Tuning and Echo Cancellation*.
12.4(20)T	Software-configurable echo cancel coverage was extended to 80, 96, 112, and 128 ms. The default parameter is 128 ms.

Finding Support Information for Platforms and Cisco IOS and Catalyst OS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Extended G.168 Echo Canceller Features

- Configuration and reporting of extended echo path capacity and worst-case ERL
- Test mode support for manually freezing, thawing, and clearing the EC h-register
- Reporting of statistics for location of the largest reflector and the internal state of the EC
- No changes to platform—Improves platform functionality by updating the EC module through a DSPWare upgrade and a Cisco IOS software upgrade
- Enabling and disabling of nonlinear processor—Enables and disables NLP spectrally matched comfort noise
- ERL configuration—Can be set to three values: 0 dB, 3 dB, and 6 dB
- Expansion of EC capacity—EC capacity is expanded to 64 ms (128 ms in Release 12.4(20)T or later)

Extended EC Comparison

Table 2 contains comparison information for G.165 and G.168 echo cancellation.

Table 2 Echo Canceller Comparison

Feature	G.165 EC	G.168 EC
Tail Coverage	Up to 32 ms	Up to 64 ms (128 ms in Release 12.4(20)T or later
Minimum ERL	Greater than or equal to 6 dB	Configurable to greater than or equal to 0 dB, 3 dB, or 6 dB
Echo Suppression	Up to 10 seconds	Not required because of faster convergence

Extended Echo Canceller Support by Platform

Table 3 lists the support for the extended G.168 EC by platform, network module, high-complexity and medium-complexity codecs, and minimum Cisco IOS release.

Table 3 Extended Echo Canceller Algorithm Coverage by Platform

Platform	Network Module	High Complexity Codec		Medium Complexity Codec		Comments
		Analog	Digital	Analog	Digital	
Cisco 1700 series	—	12.2(8)YN 12.2(13)T	12.2(8)YN 12.2(13)T	12.2(8)YN 12.3(2)T	12.2(8)YN 12.3(2)T	Flexi6 support in Cisco IOS Release 12.2(8)YN.
Cisco 2600 series Cisco 2600XM Cisco 3600 series Cisco 3700 series Cisco VG200	NM-HDV (C549)	—	12.2(13)T	—	12.2(13)T	Full support.
Cisco 2600 series Cisco 2691 Cisco 3600 series Cisco 3700 series Cisco VG200	NM-1V, NM-2V (C542)	—	—	—	—	Not supported.
Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series	NM-HD*xx*	12.3(4)XD	12.3(4)XD	12.3(4)XD	12.3(4)XD	—
Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series	AIM-Voice (C5421), AIM-Voice-30 (C542)	—	12.2(15)ZJ 12.3(4)T	—	12.2(15)ZJ 12.3(4)T	AIM.

Table 3 Extended Echo Canceller Algorithm Coverage by Platform (continued)

Platform	Network Module	High Complexity Codec		Medium Complexity Codec		Comments
		Analog	Digital	Analog	Digital	
Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series	NM-HDA (C5421)	12.2(15)ZJ 12.3(4)T	—	12.2(15)ZJ, 12.3(4)T	12.2(15)ZJ 12.3(4)T	NM-HDA. **Note** G.728 high complexity is not supported.
Cisco 2600 series	NM-HDA (C5421)	12.3(9)	—	12.3(9)	—	—
Cisco 2600 series	AIM-Voice (C5421)	—	12.3(9)	—	12.3(9)	—
Cisco 7200 series	PA-VXx-2TE1+ and PA-MCX-nTE1	—	12.2(13)T	—	12.2(13)T	PA-MCX-nTE1 port adapters do not have their own DSPs, so they use the DSPs of PA-VXx-2TE1+ port adapters.
Cisco 7500 series	—	—	12.2(13)T	—	—	No medium complexity.
Cisco 7600 series	Communiction Media Module (WS-SVC-CMM) with one of the following port adapters: WS-SVC-CMM-6T1 WS-SVC-CMM-6E1	—	12.3(8)XY 12.3(14)T	—	—	—
	WS-SVC-CMM-24FXS	12.3(8)XY 12.3(14)T	—	—	—	—
Cisco AS5300	—	—	12.2(13)T (restricted) 12.3(3) (unrestricted)	—	—	1-channel DSP on C549 with extended EC, any codec (unrestricted).
Cisco AS5350 Cisco AS5400 Cisco AS5850	NextPort DFC modules: DFC60 DFC108 1 CT3_UPC 216 UPC324	—	Digital - 12.3(11)T	—	12.3(11)T	See the NextPort-Based Voice Tuning and Echo Cancellation chapter in this guide.
Cisco Catalyst 4000	AGM	12.3(4)T	—	—	12.3(4)T	High-complexity analog and medium-complexity digital is planned.
Cisco Catalyst 6000	Cisco 6624	A002040-00002	—	A002040-00002	—	—
	Cisco 6608	—	A004040-00002	—	A004040-00002	—

Table 3 Extended Echo Canceller Algorithm Coverage by Platform (continued)

Platform	Network Module	High Complexity Codec		Medium Complexity Codec		Comments
		Analog	Digital	Analog	Digital	
Cisco Catalyst 6500 series	Communiction Media Module (WS-SVC-CMM) with one of the following port adapters: WS-SVC-CMM-6T1 WS-SVC-CMM-6E1	—	12.3(8)XY 12.3(14)T	—	—	—
	WS-SVC-CMM-24FXS	12.3(8)XY 12.3(14)T	—	—	—	—
Cisco IAD2420	—	12.2(13)T	12.2(13)T	12.3(1) mainline	12.3(1) mainline	—
Cisco IAD243*x*	VIC2-4FXO onboard T1	12.3(4)XD	12.3(4)XD	12.3(4)X D	12.3(4)XD	—
Cisco ICS 7750	—	12.2(13)T	12.2(13)T	12.2(13)T	12.2(13)T	Flexi6 support.
Cisco MC3810	HCM 549	12.2(13)T	12.2(13)T	12.3(1) mainline	12.3(1) mainline	—

How to Configure the Extended G.168 Echo Canceller

The Extended ITU-T standard G.168 Echo Cancellation feature provides an alternative to the default proprietary Cisco G.165 EC. Beginning in Release 12.4(20)T, the extended EC provides improved performance for trunking gateway applications and provides a configurable tail length that supports up to 128 ms of echo cancellation. The G.165 EC is not configurable in Cisco IOS Release 12.3(4)XD and later releases, except on the Cisco AS5300.

Note Extended echo cancellation is configured differently depending on the version of Cisco IOS software that you are using. If you are using Cisco IOS Release 12.3(4)XD or a later release, you do not have to use any Cisco IOS commands to enable the Extended ITU-T standard G.168 Echo Cancellation feature because the extended G.168 EC is the only available echo canceller. You have the option of disabling the extended EC, but it is highly recommended that you leave it enabled.

To configure the NextPort dual-filter G.168 echo canceller, see the NextPort-Based Voice Tuning and Echo Cancellation chapter in this guide.

This section contains the following procedures:

- Changing Echo Cancellers on Digital Voice Ports in Releases Prior to Cisco IOS Release 12.3(4)XD, page 11
- Configuring the Extended G.168 EC on the Cisco AS5300, page 13
- Modifying Echo Cancellation Default Settings, page 16

Table 4 lists the Cisco IOS commands that are used for selecting the extended G.168 EC based on your platform and Cisco IOS release.

Table 4 Cisco IOS Commands for Selecting Extended E.168 EC by Platform and Cisco IOS Release

Cisco IOS Release	Cisco IOS Command
Cisco 1700 Series and Cisco ICS 7750	
12.2(13)T	`Router(config)#` **`voice echo-canceller extended`**
12.2(13)ZH 12.2(15)ZJ 12.3(1)	`Router(voice-card)#` **`codec complexity medium`**
12.3(4)T and later	No configuration necessary. G.168 EC enabled by default.
Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, Cisco MC3810, Cisco VG200	
12.2(13)T 12.2(13)ZH 12.3(1)	`Router(voice-card)#` **`codec complexity medium ecan-extended`** or `Router(voice-card)#` **`codec complexity high ecan-extended`**
12.2(15)ZJ 12.3(4)T	`Router(voice-card)#` **`codec complexity medium`**
12.3(4)XD and later	No configuration necessary. G.168 EC enabled by default.
Cisco 7200 Series and Cisco 7500 Series	
12.2(13)T	`Router(config-dspfarm)#` **`codec complexity medium ecan-extended`**
12.2(13)ZH and later	No configuration necessary. G.168 EC enabled by default.
Cisco 7600 Series with Communication Media Modules	
12.3(8)XY and later	No configuration necessary. G.168 EC enabled by default.
Cisco AS5300	
12.2(13)T	`Router(config)#` **`voice echo-canceller extended codec small`** *`codec`* **`large`** *`codec`*
12.3(3)	`Router(config)#` **`voice echo-canceller extended`** `[`**`codec small`** *`codec`* **`large`** *`codec`*`]`
Cisco Catalyst 6500 Series with Communication Media Modules	
12.3(8)XY and later	No configuration necessary. G.168 EC enabled by default.
Cisco Catalyst 4000 AGM	
12.3(4)T and later	No configuration necessary. G.168 EC enabled by default.

Restrictions for G.168 Extended Echo Canceller

- Not all Cisco platforms that use C542 or C549 DSPs support the extended EC.
- The G.168 extended EC is not supported on the Cisco AS5300 in Cisco IOS Release 12.2(13)ZH.
- The Cisco 1700 series does not support the T1/E1 card in Cisco IOS Release 12.2(13)T.

Changing Echo Cancellers on Digital Voice Ports in Releases Prior to Cisco IOS Release 12.3(4)XD

To use the G.168 extended EC in a release prior to Cisco IOS Release 12.3(4)XD, perform one of the following tasks depending on your hardware platform:

Cisco 1700 or Cisco ICS 7750

- Enabling the Extended G.168 EC on the Cisco 1700 Series and Cisco ICS 7750 in Cisco IOS Release 12.2(13)T, page 11
- Enabling the Extended G.168 EC on the Cisco 1700 Series and Cisco ICS 7750 in Cisco IOS Release 12.2(13)ZH, page 12

Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, Cisco MC3810, and Cisco VG200

- Changing Codec Complexity on Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco MC3810 in the Configuring Digital Voice Ports chapter.

Cisco 7200 Series and Cisco 7500 Series

- Configuring Codec Complexity on Cisco 7200 Series and Cisco 7500 Series Routers in the Configuring Digital Voice Ports chapter

Note See Table 3 for extended EC algorithm coverage by platform.

Enabling the Extended G.168 EC on the Cisco 1700 Series and Cisco ICS 7750 in Cisco IOS Release 12.2(13)T

To change codec complexity on the Cisco 1700 series and Cisco ICS 7750 and switch between the proprietary Cisco EC and the extended G.168 EC, use the following commands.

Note You must clear all calls on the system before using the following commands. If there are active calls on the system, the commands are ignored and a warning message is issued.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice echo-canceller extended**
4. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	`voice echo-canceller extended`
	
	Example:
	`Router(config)# voice echo-canceller extended`
		Enables the G.168 extended echo canceller on the Cisco 1700 series or Cisco ICS 7750.
		• You do not have to shut down all the voice ports on the Cisco 1700 or Cisco ICS 7750 to switch the echo canceller, but you should make sure that when you switch the echo canceller, there are no active calls on the router.
		• To return to the proprietary Cisco G.165 default EC, use the **no** form of the command.

Step 4	`exit`
	
	Example:
	`Router(config)# exit`
		Exits global configuration mode.

Enabling the Extended G.168 EC on the Cisco 1700 Series and Cisco ICS 7750 in Cisco IOS Release 12.2(13)ZH

The **codec complexity medium** command enables the extended echo canceller by default on the Cisco 1700 series and the Cisco ICS 7750 in Cisco IOS Release 12.2(13)ZH.

Note You must clear all calls on the system before using the following commands. If there are active calls on the system, the commands are ignored and a warning message is issued.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-card** *slot*
4. **codec complexity** {**high** | **medium**}
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-card** *slot* **Example:** `Router(config)# voice-card 1`	Enters voice card configuration mode on the specified slot.
Step 4	**codec complexity medium** **Example:** `Router(voice-card)# codec complexity medium`	Enables the extended EC (default).
Step 5	**end** **Example:** `Router(voice-card)# end`	Exits voice-card configuration mode and completes the steps for configuring the extended EC on the Cisco 1700 series and Cisco ICS 7750.

Configuring the Extended G.168 EC on the Cisco AS5300

Perform this task to enable the Extended ITU-T standard G.168 Echo Cancellation feature on the Cisco AS5300. You must designate which EC to use on the Cisco AS5300 C542 and C549 DSPM high-complexity platform. You can use the extended EC with codec restrictions on the C542 DSP firmware or with or without codec restrictions on the C549 DSP firmware.

Note A firmware upgrade can be made by upgrading Cisco VCWare. For upgrade information, refer to the *Combined Version Release Notes and Compatibility Matrix for Cisco VCWare on Cisco AS5300 Universal Access Servers/Voice Gateways*.

Prerequisites for Extended G.168 EC on Cisco AS5300

Extended EC with Restricted Codecs on C542 or C549 DSP

1. Create a backup of your router configuration.
2. Determine which codecs are required when enabling the extended echo canceller. For this information, see the codec restriction options for the **voice echo-canceller extended** command.
   a. Specify a single small codec (g711 or g726) and a single large codec (g723, g728, GSM FR, GSM EFR, g729, or fax-relay). Any call setup involving other codecs is rejected.

b. If fax-relay is not selected as the large codec, the VoIP dial peer requires that you use the **fax rate disabled** command in dial-peer configuration mode to reset the dial peer for voice calls.

3. Review your existing configuration and look for all dial peers that select codecs or fax-relay specification that are different from the codecs that you decide on. After choosing the codecs to be supported by the extended echo canceller, either remove all dial peers with different codecs not supported by your new configuration or modify the dial-peer codec selection by selecting a voice codec or fax-relay that is supported by the new configuration.
4. Ensure that modem relay is not configured in any of the dial-peer configurations. If modem relay is configured, it should be disabled using the **no modem relay** command.

Extended EC with Unrestricted Codecs on C549 DSP

1. Create a backup of your router configuration.
2. Ensure that modem relay is not configured in any of the dial-peer configurations. If modem relay is configured, it should be disabled using the **no modem relay** command.

Restrictions for Extended G.168 EC on the Cisco AS5300

- The extended G.168 EC can be used only in one of the following ways on the Cisco AS5300:
  - With a restricted set of codecs with C542 or C549 DSP firmware. Two channels of voice are supported per DSP, and full call handling capacity is supported.
  - With no restrictions on codecs with C549 DSP firmware. One channel of voice is supported per DSP, and call handling capacity is reduced by half.
- Not all Cisco platforms that use C542 or C549 DSPs support the extended EC. Other platforms continue to use the proprietary Cisco G.165 EC if they do not support the extended EC.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **no dial-peer voice** *tag* **voip**
4. **dial-peer voice** *tag* **voip**
5. **codec** {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** |**g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*]
6. **exit**
7. **voice echo-canceller extended**

   or

   voice echo-canceller extended [**codec small** *codec* **large** *codec*]
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**no dial-peer voice** *tag* **voip** **Example:** `Router(config)# no dial-peer voice 1 voip`	(Optional) Removes VoIP dial peers, one dial peer at a time. • When configuring the extended EC in global configuration mode, you must remove or modify all existing VoIP dial peers before the **voice echo-canceller extended** command is accepted.
Step 4	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	Enters dial-peer configuration mode so that you can modify a codec type.
Step 5	**codec** {**g711alaw** | **g711ulaw** | **g723ar53** | **g723ar63** | **g723r53** | **g723r63** | **g726r16** | **g726r24** | **g726r32** | **g726r53** | **g726r63** | **g728** | **g729abr8** | **g729ar8** | **g729br8** | **g729r8** | **gsmefr** | **gsmfr**} [**bytes** *payload-size*] **Example:** `Router(config-dialpeer)# codec g711alaw`	Specifies the voice codec rate for the dial peer.
Step 6	**exit** **Example:** `Router(config-dialpeer)# exit`	Exits dial-peer configuration mode and returns to global configuration mode.

	Command or Action	Purpose

Step 7	**voice echo-canceller extended**
	or
	voice echo-canceller extended [**codec small** *codec* **large** *codec*]
	
	Example:
	`Router(config)# voice echo-canceller extended`
	
	Example:
	`Router(config)# voice echo-canceller extended codec small g711 large fax-relay`
		Enables the extended echo canceller with no restriction on codecs.
		or
		Enables the extended echo canceller with restricted codecs.
		• The following codec choices are valid:
		– Small footprint codec—G.711 or G.726.
		– Large footprint codec—G.729, G.726, G.728, G.723, fax-relay, GSM FR, or GSM EFR.
		Note The **voice echo-canceller extended** command enables the extended EC without codec restrictions on a Cisco AS5300 with C549 DSP firmware. One channel of voice is supported per DSP. Any codec is supported.
		Note The **voice echo-canceller extended codec** command restricts codecs on the Cisco AS5300 with C542 and C549 DSP firmware. Two channels of voice are supported per DSP. Only specific codecs are supported.

Step 8	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode.

Modifying Echo Cancellation Default Settings

Perform this task to modify the default settings for echo cancellation parameters.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot*/*port***:***ds0-group-number*
4. **echo-cancel enable**
5. **echo-cancel coverage** {**24** | **32** | **48** | **64** | **80** | **96** | **112** | **128**}
6. **echo-cancel erl worst-case** {**0** | **3** | **6**}
7. **non-linear**
8. **echo-cancel suppressor** *seconds*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot/port*:*ds0-group-number* **Example:** `Router(config)# voice-port 1/0:0`	Enters voice-port configuration mode on the selected slot, port, and DS0 group. **Note** The syntax of this command is platform-specific. For the syntax for your platform, refer to the *Cisco IOS Voice Command Reference*.
Step 4	**echo-cancel enable** **Example:** `Router(config-voiceport)# echo-cancel enable`	Enables echo cancellation. • Echo cancellation is enabled by default. Use the **no** form of this command to disable echo cancellation. • The extended G.168 EC is the default EC for all supported platforms in Cisco IOS Release 12.3(4)XD and later releases, except the Cisco AS5300. • On the Cisco AS5300, the Cisco G.165 EC is enabled by default with echo suppression disabled. **Note** This command is supported only when the **echo-cancel coverage** command is enabled.
Step 5	**echo-cancel coverage** {**24** | **32** | **48** | **64** | **80** | **96** | **112** | **128**} **Example:** `Router(config-voiceport)# echo-cancel coverage 64`	Adjusts the size of the echo canceller (echo path capacity coverage). • **128** is the default in Cisco IOS Release 12.4(20)T and later releases. Prior to this release, the default is **64** ms. **Note** This command is supported only when echo cancellation is enabled. See Step 4 of this procedure.

	Command or Action	Purpose
Step 6	**echo-cancel erl worst-case** [**0** | **3** | **6**] **Example:** `Router(config-voiceport)# echo-cancel erl worst-case 6`	(Optional) Determines worst-case ERL in dB. • Worst-case ERL is the minimum expected attenuation in the voice path. For example, if you have a worst-case ERL of 6 (**erl worst-case 6**), when you speak into the phone you can expect at least 6 dB of attenuation on the signal by the time it gets back to the original source (echo). In general you do not need to change this value from 6, which is the default. Worst-case ERL does not directly modify the inbound or outbound signals. This is purely a configuration parameter for the EC to help it distinguish between echo and a new signal. **Note** This command is supported for the extended G.168 EC only; it is not supported for the G.165 EC.
Step 7	**non-linear** **Example:** `Router(config-voiceport)# non-linear`	(Optional) Selects nonlinear processing (residual echo suppression) in the EC, which either shuts off any signal or mixes in comfort noise if no near-end speech is detected. **Note** This command is supported only when echo cancellation is enabled. See Step 4. • Nonlinear processing is enabled when the extended G.168 EC is enabled. Use the **no** form of this command to disable the NLP.

	Command or Action	Purpose
Step 8	**echo-cancel suppressor** *seconds* **Example:** `Router(config-voiceport)# echo-cancel suppressor 10`	(Optional) Applies echo suppression for the number of seconds specified when using the G.165 EC. • This command cannot be used with the extended G.168 EC in Cisco IOS Release 12.2(15)ZJ or later releases, or on NextPort (Cisco AS5350 and Cisco AS5400) platforms. **Note** This command is required to configure the Extended ITU-T standard G.168 Echo Cancellation feature in Cisco IOS Release 12.2(13)T. • For the AS5300, the Cisco G.165 EC is enabled by default with echo suppression disabled. The echo suppressor can be used only on T1 DSPs when the default Cisco G.165 EC is used. • This command enables echo cancellation for voice that is sent out an interface and received back on the same interface within the configured amount of time. • This command reduces the initial echo before the echo canceller can converge. In case of double-talk in the first number of seconds, the code automatically disables the suppressor. • The **echo-cancel suppressor** command is visible when the G.168 extended EC is selected but it has no effect. **Note** This command is supported only when echo cancellation is enabled.
Step 9	**end** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and completes the configuration.

Configuration Examples for Extended G.168 Echo Cancellation

This section includes the following examples:

- Enabling the Extended EC on the Cisco 7200 Series or Cisco 7500 Series: Example, page 21
- Enabling the Extended Echo Canceller on the Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco VG200 in Releases Prior to Cisco IOS Release 12.3(4)XD: Example, page 20
- Enabling the Extended EC on the Cisco 7200 Series or Cisco 7500 Series: Example, page 21
- Enabling the Extended Echo Canceller on the Cisco AS5300: Example, page 21
- Adjusting the Echo Canceller Size: Example, page 22
- Worst-Case Echo Return Loss: Example, page 22

Enabling the Extended EC on the Cisco 1700 Series and Cisco ICS 7750: Example

The following example enables the G.168 extended EC on a Cisco 1700 series or a Cisco ICS 7750. The extended EC is enabled by default when the **medium** keyword is used in Cisco IOS Release 12.2(13)ZH and later.

```
voice-card 1
 codec complexity medium
```

Enabling the Extended Echo Canceller on the Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco VG200 in Releases Prior to Cisco IOS Release 12.3(4)XD: Example

The following example shows that the echo canceller has been changed from the default proprietary Cisco EC to the extended EC in Cisco IOS releases prior to 12.3(4)XD.

Note The extended G.168 EC is the only EC in Cisco IOS Release 12.3(4)XD and later releases. Because it is enabled by default, it does not display in the configuration output in Cisco IOS Release 12.3(4)XD and later releases.

The following is example output from an originating Cisco 3640:

```
.
.
.
voice-card 1
 codec complexity high ecan-extended
.
.
.
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
!
voice-port 1/0:23
.
.
.
dial-peer voice 104001 voip
 destination-pattern 104001
 session target ipv4:10.2.0.104
 dtmf-relay cisco-rtp
 codec g711alaw
 fax rate 14400
 fax protocol cisco
.
.
.
```

Enabling the Extended EC on the Cisco 7200 Series or Cisco 7500 Series: Example

The following example changes codec complexity on a Cisco 7200 series or 7500 series:

```
dspint dspfarm 2/0
 codec medium ecan-extended
```

Enabling the Extended Echo Canceller on the Cisco AS5300: Example

The following example enables the extended G.168 EC with restricted codecs on the Cisco AS5300 with C542 or C549 DSP firmware:

```
!
version 12.3
no service pad
service timestamps debug datetime msec
service timestamps log uptime
no service password-encryption
service internal
!
hostname router
!
boot-start-marker
boot-end-marker
!
enable secret 5 $123
enable password temp
!
!
resource-pool disable
!
no aaa new-model
ip subnet-zero
ip rcmd rcp-enable
ip rcmd rsh-enable
ip domain name cisco.com
ip host router1 10.10.101.14
!
!
isdn switch-type primary-5ess
!
voice echo-canceller extended codec small g711 large fax-relay
!
!
!
fax interface-type fax-mail
!
!
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 pri-group timeslots 1-24
.
.
.
```

Adjusting the Echo Canceller Size: Example

The following example adjusts the size of the extended EC to 64 ms on Cisco 3600 series routers:

```
voice-port 1/0:0
 echo-cancel coverage 64
```

Worst-Case Echo Return Loss: Example

The following example sets the worst-case echo return loss to 3:

```
voice-port 0:D
 echo-canceller erl worst-case 3
 playout-delay mode fixed
 no comfort-noise
```

Configuring Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards

The multiflex trunk (MFT) dedicated echo cancellation modules (dedicated ECAN modules) are daughter cards that attach to the second generation multiflex voice/WAN interface cards (MFT VWIC2 family). The dedicated ECAN modules are available in 32-channel and 64-channel configurations (EC-MFT-32 and EC-MFT-64), which match the requirements of the 1- and 2-port T1/E1 MFT VWIC2s, respectively. This chapter describes the configuration to enable additional echo cancellation effectiveness:

- Control of the echo canceller provided through the size of the echo cancellation buffer, ranging from 24 milliseconds (ms) to 128 ms
- Processing and memory resources to ensure robust echo canceller coverage independent from the configuration of the echo canceller or the demand placed on the general voice DSP resources

Prerequisites for Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards

Cisco IOS Image

To run hardware echo cancellation on T1/E1 interfaces, you must install an IP Plus or IP Voice image (minimum) of Cisco IOS Release 12.3(14)T or a later release.

Baseboard and Daughter Card Configuration

Hardware echo cancellation is restricted to the same baseboard voice/WAN interface card (VWIC) on which the daughter card (EC-MFT-32 and EC-MFT-64) is installed and cannot be shared by other T1/E1 controllers.

Restrictions for Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards

Hardware Echo Cancellation Tail Length

If you are using hardware echo cancellation, the value for tail length is set to 128 ms. This is not configurable and cannot be changed.

Accurate TDM ERL Readings for Echo Cancellation

To ensure accurate statistics for network monitoring and troubleshooting, an estimate of the quality of the TDM connection and the ECAN's ability to discern and cancel out echo might be necessary. To ensure accurate readings, you must configure software-based echo cancellation by entering the **echo-cancel enable type software** command (Step 6 in the procedure in "How to Configure Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards" section on page 4). If you accept the default (hardware echo cancellation) or enter the **echo-cancel enable type hardware** command, the output for the **show voice call** command always displays "TDM ERL Level(dBm0): +6.0."

If you enter the **echo-cancel enable type software** command to enable software-based echo cancellation, the **show voice call** command output displays accurate real-time TDM ERL measurements. The sample output examples provided in the following sections demonstrate the difference:

- Sample Output of the show voice call command without Hardware Echo Cancellation
- Sample Output of the show voice call command with Hardware Echo Cancellation

Sample Output of the show voice call command without Hardware Echo Cancellation

The following is sample output of software-enabled echo cancellation—hardware echo cancellation is disabled. Note the different values for the TDM ERL levels.

```
Router# show voice call 0/0/0:23.1

0/0/0:23 1
      vtsp level 0 state = S_CONNECT
callid 0x0001 B01 state S_TSP_CONNECT clld 9011204 cllg 9011200
Router#  ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 3563, Tx Sig Pkts: 0, Tx Comfort Pkts: 4
Tx Dur(ms): 80150, Tx Vox Dur(ms): 71200, Tx Fax Dur(ms): 0
.
.
.
        ***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -12.5 from PBX/Phone, Tx -16.4 to PBX/Phone
TDM ACOM Levels(dBm0): +27.0, TDM ERL Level(dBm0): +27.0
TDM Bgd Levels(dBm0): -84.4, with activity being silence
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0

Router# show voice call 0/0/0:23.2

0/0/0:23 2
      vtsp level 0 state = S_CONNECT
callid 0x0002 B02 state S_TSP_CONNECT clld 9011202 cllg 9011205
```

```
Router#  ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 1800, Tx Sig Pkts: 0, Tx Comfort Pkts: 0
Tx Dur(ms): 36000, Tx Vox Dur(ms): 36000, Tx Fax Dur(ms): 0
.
.
.
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -23.5 from PBX/Phone, Tx -36.5 to PBX/Phone
TDM ACOM Levels(dBm0): +6.0, TDM ERL Level(dBm0): +6.0
TDM Bgd Levels(dBm0): +0.0, with activity being silence
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
```

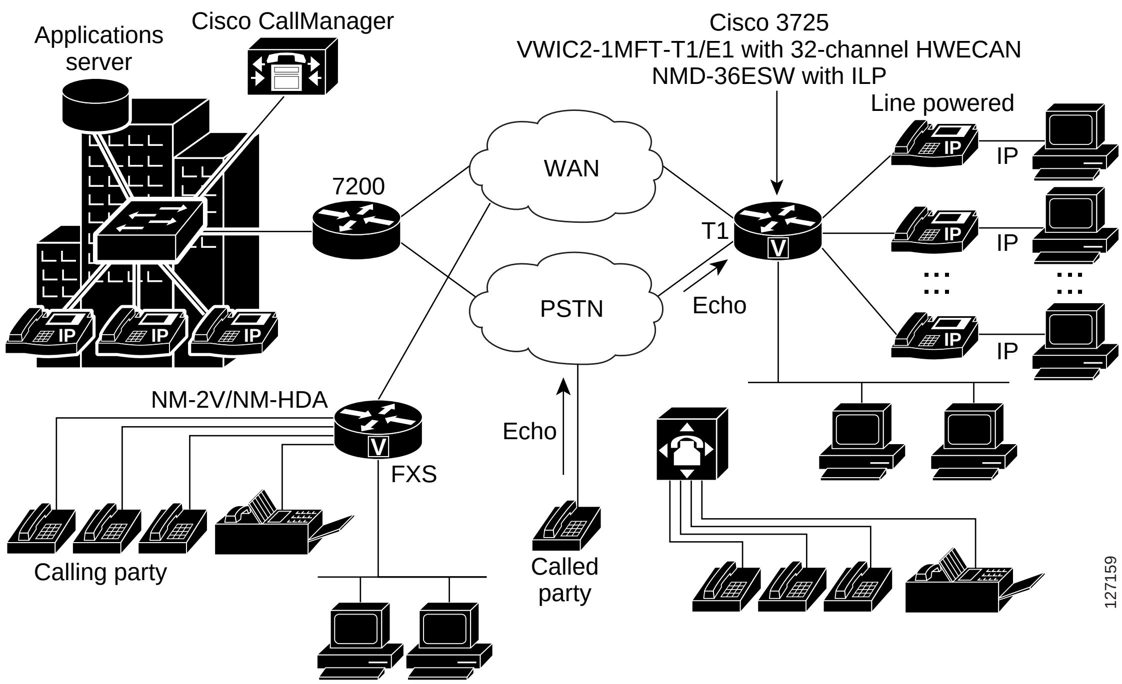
Sample Output of the show voice call command with Hardware Echo Cancellation

The following is sample output showing hardware echo cancellation—note that the TDM ERL level is +6.0 in both cases.

```
Router# show voice call 0/0/0:23.1

0/0/0:23 1
      vtsp level 0 state = S_CONNECT
callid 0x0002 B01 state S_TSP_CONNECT clld 9011204 cllg 9011200
Router#
        ***HARDWARE ECHO CANCELLER STATISTICS***
        Echo Canceller: On   Tail-length: 128ms
        H-Register: Update   Modem tone disable: Ignore 2100Hz tone
        Worst ERL : 6dB     Residual Control: Comfort noise
        High level compensation: Off
        Tx Power = 0.0dB  Tx Avg Power = 0.0dB
        Rx Power = 0.0dB  Rx Avg Power = 0.0dB
        ERL = 27.0dB     ACOM = 0.0
        3 Reflectors(Tails) = (1, 0, 0)Ms, Max Reflector = 1Ms
        Ecan Status words 0x7C, 0x1001
        EC Lib version: 9183.890
.
.
.
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -12.4 from PBX/Phone, Tx -15.1 to PBX/Phone
TDM ACOM Levels(dBm0): +6.0, TDM ERL Level(dBm0): +6.0
TDM Bgd Levels(dBm0): -84.4, with activity being silence
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0


Router# show voice call 0/0/0:23.2

0/0/0:23 2
      vtsp level 0 state = S_CONNECT
callid 0x0004 B02 state S_TSP_CONNECT clld 9011202 cllg 9011205
cmohanan-3845#
        ***HARDWARE ECHO CANCELLER STATISTICS***
        Echo Canceller: On   Tail-length: 128ms
        H-Register: Update   Modem tone disable: Ignore 2100Hz tone
        Worst ERL : 6dB     Residual Control: Comfort noise
        High level compensation: Off
        Tx Power = 0.0dB  Tx Avg Power = 0.0dB
        Rx Power = 0.0dB  Rx Avg Power = 0.0dB
        ERL = 6.0dB     ACOM = 0.0
        3 Reflectors(Tails) = (4, 0, 0)Ms, Max Reflector = 4Ms
```

```
        Ecan Status words 0x7C, 0x1001
        EC Lib version: 9183.890
.
.
.
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx -24.9 from PBX/Phone, Tx -35.7 to PBX/Phone
TDM ACOM Levels(dBm0): +6.0, TDM ERL Level(dBm0): +6.0
TDM Bgd Levels(dBm0): +0.0, with activity being silence
        ***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
```

Figure 1 Sample Network Topology for the T1/E1 Multiflex Voice/WAN Interface Cards with Echo Cancellation Module



How to Configure Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards

To configure hardware echo cancellation on T1/E1 multiflex voice/WAN interface cards, complete the following tasks.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **card type** {**e1** | **t1**} *slot subslot*
4. **voice-card** *slot*

5. **voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*}
6. **echo-cancel enable type** [**hardware** | **software**]
7. **echo-cancel coverage** {**24** | **32** | **48** | **64** | **80** | **96** | **112** | **128**}
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`card type {e1	t1}`** *`slot subslot`*  **Example:** `Router(config)# card type t1 1 0`	Sets or changes the card type to E1 or T1. • *slot*—Specifies the slot number. Range can be 0 to 6, depending on the platform. • *subslot*—Specifies the VWIC slot number. Range can be 0 to 3, depending on the host module or platform. • When the command is used for the first time, the configuration takes effect immediately. • A subsequent change in the card type will not take effect unless you enter the **reload** command or reboot the router. **Note** When you are using the **card type** command to change the configuration of an installed card, you must enter the **no card type** {**e1** | **t1**} *slot subslot* command first. Then enter the **card type** {**e1** | **t1**} *slot subslot* command for the new configuration information.
Step 4	**`voice-card`** *`slot`* **Example:** `Router(config)# voice card 1`	Enters voice card configuration mode. • Specify the slot location using a value from 0 to 5.

	Command or Action	Purpose
Step 5	**voice-port** {*slot-number*/*subunit-number*/*port* | *slot*/*port*:*ds0-group-number*} **Example:** `Router(voice-card)# voice-port 3/0:0`	Enters voice port configuration mode and specifies the voice port. • The *slot-number* argument identifies the slot where the voice interface card (VIC) is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed. • The *subunit-number* identifies the subunit on the VIC where the voice port is located. Valid entries are 0 or 1. • The *port* argument identifies the voice port number. Valid entries are 0 and 1. or • The *slot* argument is the slot in which the voice port adapter is installed. Valid entries are from 0 to 3. • The *port* argument is the voice interface card location. Valid entries are 0 to 3. • The *ds0-group-number* argument indicates the defined DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card. **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 to determine the syntax choices that are available.
Step 6	**echo-cancel enable type** [**hardware** | **software**] **Example:** `router(config-voiceport)# echo-cancel enable type hardware`	Enables hardware echo cancellation. • **The hardware keyword** is the default. Echo cancel coverage is hardcoded for 128 ms. • This command is needed only to configure the **software keyword** to effect software-based (DSP) echo cancellation or to restore the **hardware** default. **Note** The **hardware** and **software** keywords are available only when the optional hardware echo cancellation module (EC-MFT-32 or EC-MFT-64) is installed on the multiflex VWIC. **Note** If you need to obtain accurate, real-time readings for the quality of the TDM connection and the echo canceller's ability to discern and cancel out echo, you should enter the **echo-cancel enable type software** command. See the "Restrictions for Hardware Echo Cancellation on T1/E1 Multiflex Voice/WAN Interface Cards" section on page 2 for more information.

	Command or Action	Purpose
Step 7	**echo-cancel coverage {24 | 32 | 48 | 64 | 80 | 96 | 112 | 128}** **Example:** `Router (config-voiceport) # echo-cancel coverage 96`	Adjusts the echo canceller by the specified number of milliseconds. • These coverage options are applicable only if you configured the **echo-cancel enable type software** command in the previous step. • If you configured the **echo-cancel enable type hardware** command in the previous step, this value is set to 128 ms. • Beginning with Release 12.4(20)T, the default for software echo cancellation is 128 ms. Prior to Release 12.4(20)T, the default is 64 ms.
Step 8	**exit** **Example:** `Router(config-voiceport)# exit`	Exits controller cofiguration mode and returns the router to privileged EXEC mode.

Examples

This section provides the following examples for verifying echo cancellation:

- show echo-cancel hardware status: Example, page 7
- show call active voice echo-canceller summary: Example, page 7
- show call active voice echo-canceller CallID: Example, page 8

show echo-cancel hardware status: Example

The output in this section shows that hardware echo cancellation is enabled on slot 1.

```
Router_3725# show echo-cancel hardware status 1

VWIC HWECAN 1/0 is UP.
Software version:4.4.803 , Date:Feb  6 16:58:57 2004
Tail length:128     Tone disabler type:G.165    Fax notify: Off
Device:VWIC_8MBPS_1TIEC_TL128_MS_1P  Max Channels:32
Only Port0 have Local HWECAN Connectivity.


 ECAN CH   ASSIGNED      DSP ID     VOICEPORT      EC        NLP     COV     LAW
 ==========================================================================================
 1         yes           1/1        1/0:1.1        on        off     on      u-Law



Total assigned channel(s):1
Total device(s) in the slot 1
```

show call active voice echo-canceller summary: Example

The output in this section shows summary information for the hardware echo cancellation.

```
Router_3725# show call active voice echo-canceller summary
```

```
Call ID    Port        DSP/Ch     Codec      Ecan-type   Tail    Called #    Dial-peers
=========================================================================================
0xE71      1/0:1.1     1/1        g729r8     HW          128ms   1000        1/10


1 active call found
number of hardware ecan channels:1
number of software ecan channels:0
```

show call active voice echo-canceller CallID: Example

The output in this section shows hardware echo canceller information for an active voice call.

```
Router# show call active voice echo-canceller E71

Device:VWIC HWECAN 1/0 Channel Id = 1 Tail = 128Ms
Software version:4.4.803 , Date:Feb  6 16:58:57 2004
        Echo Canceller:On   Tail-length:128ms
        H-Register:Update   Modem tone disable:Ignore 2100Hz tone
        Worst ERL :6dB     Residual Control:Cancel only
        High level compensation:Off
        Tx Power = 0.0dB  Tx Avg Power = 0.0dB
        Rx Power = 0.0dB  Rx Avg Power = 0.0dB
        ERL = 1.0dB     ACOM = 0.0
        3 Reflectors(Tails) = (90, 0, 0)Ms, Max Reflector = 90Ms
        Ecan Status words 0x1C, 0x00
        EC Lib version:9155
```

More detailed syntax information about the commands used in this chapter is documented in the *Cisco IOS Voice Command Reference*.

NextPort-Based Voice Tuning and Echo Cancellation

This chapter describes how to dynamically configure voice services on the NextPort-based platforms: Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850. This chapter contains the following sections:

Prerequisites for Nextport-Based Voice Tuning and NextPort Dual-Filter G.168 Echo Canceller

To use the Nextport-Based Voice Tuning and Background Noise Statistics feature, you must be running NextPort service processing element (SPE) firmware version 8.8.1 or a later version and Cisco IOS Release 12.3(4)T or a later release.

To use the NextPort dual-filter G.168 echo canceller, you must be running SPE firmware version 10.2.2 or a later version and Cisco IOS Release 12.3(11)T or a later release.

Information About NextPort Voice Services

To configure the features presented in this chapter, you should understand the following concepts:

NextPort Dual-Filter G.168 Echo Canceller

The CSMV6 dial feature card (DFC) for NextPort platforms offers dual-filter G.168 echo canceller capability. The NextPort dual-filter G.168 echo canceller (EC) improves voice quality in VoIP connections by providing relatively less residual echo leakage, better nonlinear processing (NLP) timing, less clipping, and better comfort noise generation (CNG) in most environments.

The dual-filter G.168 echo canceller features two concurrently operating adaptive filters (which control echo tail coverage) and two double-talk detection functions. In addition, the comfort noise model uses "Hoth noise" spectrum shaping to better replicate the true noise spectrum.

Note Detailed information for all Cisco IOS commands mentioned in this section can be found in the *Cisco IOS Voice Command Reference*.

The NextPort dual-filter G.168 echo canceller uses the same voice-tuning (VC tune) interface for configuring voicecap parameters as the Cisco-proprietary G.164 echo canceller. To adjust the dual-filter echo canceller, use a voicecap or the Cisco IOS command-line interface (CLI) during configuration. You can also adjust settings while the system is running by using the **show port log** and **show port operational-status** commands. However because of the differences in internal operation of these ECs, there are some changes in the set of available parameters for voice tuning.

See the **echo-cancel coverage** command for updated Cisco IOS command usage with this feature. The NextPort dual-filter G.168 echo canceller adds the following benefits on NextPort platforms:

- Configurable parameters—Range checking that is performed on the voicecap parameters in the I960 NextPort layer has been updated. (Voicecap parameters in "raw mode" are never range-checked.)
- Up to 128 ms of echo tail coverage—Beginning with Cisco IOS Release 12.4(20)T, the NextPort dual-filter G.168 echo canceller supports echo tails from 24-ms to 128-ms in 16-ms increments. The **echo-cancel coverage** command limits the echo canceller coverage to 128-ms on NextPort platforms. For backward compatibility, a voicecap used in "raw mode" will still configure older SPEware to settings greater than 64-ms when used with newer releases of Cisco IOS software. For situations when new SPEware is loaded onto an older Cisco IOS release, the NextPort dual-filter G.168 echo canceller automatically sets coverage time to 64 ms.
- Updated set of reported statistics—Text in the **show voice port** command output has been changed to describe voicecap parameters and reported statistics. The **show port operational-status** command output has been updated to report TX/RX mean speech level statistics.
- Power statistics (RX and TX)—These statistics average only the power that is received during signal periods that are classified as speech.
- Unchanged configuration steps—Use voicecaps and the **echo-cancel coverage** command to configure this feature. See the "Voicecap Strings" section on page 3.
- SPE firmware and Cisco IOS software packaging support—The SPEware that contains the dual-filter G.168 echo canceller is field-upgradeable and can be used interchangeably with previous firmware versions with no effect on platform call density. The new SPEware interoperates with any Cisco IOS software release that supports voicecaps.

Note When older Cisco IOS software releases are used, voicecaps must be used in raw mode for some parameters. Some statistics may not be displayed or recorded properly with older software releases.

NextPort SPE Firmware

NextPort SPE firmware is software that drives the digital signal processor (DSP) portion of the NextPort dial feature cards (DFCs). NextPort firmware is bundled with Cisco IOS software.

NextPort SPE firmware runs on the NextPort DFC60, DFC108, 1 CT3_UPC 216, and UPC324 DFCs on the Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850 platforms. The ports on these modules can support modem, voice, fax, and digital services and can be aggregated at any of the following levels:

- Slot level of the NextPort module
- SPE level within the NextPort module
- Individual port level

Note To use the NextPort Voice Tuning and Background Noise Statistics feature, you must use the default bundled NextPort SPE firmware code that runs with Cisco IOS software. The NextPort-Based Voice Tuning and Background Noise feature uses SPE firmware version 8.8.1 or a later version. The NextPort dual-filter G.168 echo canceller uses NextPort firmware version 10.2.2, which is bundled with Cisco IOS Release 12.3(11)T. NextPort firmware version 10.2.2 can be used with Cisco IOS Release 12.3(7)T, 12.3(10), and later releases.

For more information about NextPort SPE firmware, see the NextPort SPE Release Notes on Cisco.com.

Voicecap Strings

Additional configuration of voice services on NextPort DFCs is achieved by configuring the voice tuning configuration capability (called voicecaps) using voicecap strings. Voicecap strings are created with the the **voicecap entry** command and are applied with the **voicecap configure** command.

Voice Tuning

Note You must have specific knowledge of the behavior of the telephone network in order to use these voicecap capabilities. See the *Nextport-Based Voice Tuning White Paper* on Cisco.com for a brief description of telephone network behavior.

This feature allows the following parameters, among others, to be configured:

- PSTN gains—PSTN gains adjust the power levels at the PSTN side of a VoIP connection to make up for loss plan imbalances and to ensure minimum echo return losses (ERLs) in a call. PSTN gain is configured with the CLI rather than with voicecaps.
- IP gains—IP gains adjust IP-side levels and are applied to the signal before it is propagated through the echo canceller. This point is also known as the reference signal.
- Dynamic attenuation—Dynamic attenuation mitigates low volume calls when attenuation has been added on the PSTN call leg to compensate for low ERL calls.
- Comfort noise generation—CNG enables and disables EC comfort noise.
- Minimum ERL—Minimum ERL switches off near-end talker clipping and poor echo canceller performance.

Background Noise

The NextPort Voice Tuning and Background Noise Statistics feature reports EC background noise level, voice activity detection (VAD) background noise level, ERL level, and Acombined (ACOM) statistics by averaging the combined values that are computed over the duration of the call. These statistics are appended to the end of each entry in the voice log, which you can see in the output from the **show port log** and **show port operational-status** commands.

How to Configure Nextport-Based Voice Tuning and the NextPort Dual-Filter G.168 Echo Canceller

To configure the Nextport-Based Voice Tuning and the NextPort Dual-Filter G.168 Echo Canceller feature, complete the following tasks:

- Downloading NextPort SPE Firmware, page 101 (required)
- Creating and Applying Voicecaps, page 103 (required)
- Setting Voice Tuning Parameters with V Registers, page 105 (optional)

Downloading NextPort SPE Firmware

To download NextPort SPE firmware, use the following commands:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **spe** {*first slot* | *first slot*/*spe*} {*last slot* | *last slot*/*spe*}
4. **firmware location** [*IFS*:[/]]*filename*
5. **end**
6. **copy running-config startup-config**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	**spe** {*first slot* | *first slot/spe*} {*last slot* | *last slot/spe*} **Example:** `Router(config)# spe 1 1/17`	Enters SPE configuration mode and sets the range of SPEs. • *first slot* and *last slot*—Identifies slots for the range. For the Cisco AS5350, slot values range from 1 to 3. For the Cisco AS5400, slot values range from 1 to 7. All ports on the specified slot are affected. • *first slot/spe* and *last slot/spe*—Identifies slots for the range. For the Cisco AS5350, slot values range from 1 to 3. For the Cisco AS5400, slot values range from 1 to 7. SPE values range from 1 to 17. You must include the slash mark. All ports on the specified slot and SPE are affected.
Step 4	**firmware location** [*IFS*:[/]]*filename* **Example:** `Router(config-spe)# firmware location flash:np.8.8.1.spe`	Downloads SPE modem code to all modems in a particular slot (that is, all modems on a feature card that contains 18 6-port modem modules). • *IFS*—(Optional) Cisco IOS file specification (IFS), which can be any valid IFS on any local file system. Examples of legal specifications include: – **bootflash:**—Loads the firmware from a separate flash memory device. – **flash:**—Loads the firmware from the flash NVRAM located within the router. – **null**:—Specifies a firmware file from null: File System. – **system:/**—Loads the firmware from a built-in file within the Cisco IOS image. The optional forward slash (/) and system path must be entered with this specification. • *filename*—The firmware filename. When the filename is entered without an IFS specification, this name defaults to the file in flash memory. • Use the **dir all-filesystems** EXEC command to display legal IFSs. • The **no** form of the command reverts the router back to the system-embedded default. When the access server is booted, the **firmware location** command displays the location for the firmware that is embedded in the Cisco IOS image. If the **firmware location** command is issued to download a firmware image from flash and then the **no** version of the exact command is subsequently issued, then the **firmware location** command downloads the embedded firmware in Cisco IOS software.

	Command or Action	Purpose
Step 5	**end** **Example:** `Router(config-spe)# end`	Completes the download and exits SPE configuration mode.
Step 6	**copy running-config startup-config** **Example:** `Router# copy running-config startup-config`	Copies the configuration from running RAM into NVRAM. • The download occurs when the modems become available and the display shows the SPE firmware upgrade option defined (default: busyout). • The **spe** command generates NVRAM modem download and configuration file entries. **Note** If the configuration is not saved as described in this step, download of the firmware specified with the **spe** command will not occur after the next reboot. • For detailed information on the **spe** command, see the following Cisco document: *SPE and Firmware Download Enhancements*.

Creating and Applying Voicecaps

Voicecaps can be used to configure any of the voice service parameters. Voicecaps are, however, primarily used to configure only those parameters that do not have associated Cisco IOS commands.

Restrictions

- Voicecaps are configured in global configuration mode. A maximum of five voicecap entries can be defined.
- Applying a voicecap is possible only in voice-port configuration mode. Once applied to a voice port, the voicecap affects all calls associated with that voice port.
- To achieve the specified functionality, an SPE image capable of voice tuning must be used in conjunction with the Cisco IOS software and module controller software.
- For backward compatibility, a voicecap used in raw mode will configure older SPEware to allow echo canceller coverage settings greater than 64 ms when the older SPEware is used with newer releases of Cisco IOS software. For situations when new SPEware is loaded onto an older Cisco IOS software release, the NextPort dual-filter G.168 echo canceller automatically sets coverage time to 64 ms.

Note Voicecap parameters in raw mode are never range-checked.

For a list of available voicecap parameters and code words that are used with the NextPort dual-filter G.168 echo canceller feature, see the *NextPort Dual-Filter G.168 Echo Canceller White Paper*.

Setting Voice Tuning Parameters with V Registers

The following sections contain information about voice tuning parameters with V registers:

Note All data specified in dB is entered in the form (dB * 10). So, for example, to specify 6.0 dB, 60 must be entered.

Set PSTN Gains

Note Too much attenuation may cause some calls to have low volume speech. For more information, see the dynamic attenuation feature described in the *Nextport-Based Voice Tuning White Paper*.

To fix poor loss plans and to ensure minimum ERLs, use PSTN gains to adjust the power levels on the PSTN call leg. To adjust these levels, make sure that the sum of the output attenuation, the input attenuation, and the lowest expected functional ERL is greater than the MinERL parameter setting described below. You should balance the power levels of the far-end talker and near-end talker as seen at the echo canceller.

Note Use the Cisco IOS CLI, not voicecap indexes, to set PSTN gains.

Set IP Gains

To adjust IP-side levels that are applied to the signal before it is propagated through the echo canceller, use IP gains. IP gains are controlled with the following V registers. The valid range for both input and output gain is –14 dB to 14 dB.

- v261—IP output gain.
- v263—IP input gain.

Note There have been some instances where the IP-side power has been too high. Using index v263 can mitigate this problem.

Set Dynamic Attenuation

To mitigate low volume calls when attenuation has been added on the PSTN call leg to compensate for low ERL calls, use dynamic attenuation. Dynamic attenuation is controlled with the following V registers:

- v289—Dynamic EC Attenuation Feature Enable. Set to 1 to enable. Set to 0 to disable.

- v290—Dynamic EC Attenuation Minimum ERL Value. Valid range is from 0 dB to 60 dB.
- v291—Dynamic EC Attenuation Final Rout Gain. Set to the lowest level desired for PSTN output attenuation. This value is usually set to 0. Valid range is from –14 dB to 6 dB.
- v292—Dynamic EC Attenuation Final Sin Gain. Set to the lowest level desired for PSTN input attenuation. This value is usually set to 0. Valid range is from –14 dB to 6 dB.

Set Comfort Noise Generation

During periods of far-end single talk, the echo canceller engages the non-linear processor (NLP) to suppress residual echo. However, it will also suppress any noise signal that is coming from the near-end side. This can lead to dead silence, which the listener may confuse with a dropped call. To overcome this condition, comfort noise generation (CNG) is added.

To choose between NLP silence or background noise reproduction, use comfort noise generation. CNG is controlled with the following V register:

- v294—CNG Enable. Set to 1 to enable. Set to 0 to disable.

Set Minimum ERL

The echo canceller uses the minimum ERL (MinERL) value to decide whether the incoming signal on the PSTN call leg is an echo or a near-end talker. If this value is too high, the echo canceller will not properly identify echo and will not adapt. If this value is too low, clipping of the near-end talker may occur.

To reduce near-end talker clipping and poor echo canceller performance, use minimum ERL. MinERL is controlled with the following V register:

v270—Sets the level that the echo canceller expects the lowest ERL of the PSTN to be. The valid range is from 0 dB to 20 dB. The default is 6.

Creating and Applying Voice Caps

To create and apply voice caps and voice cap entries, complete the following tasks:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voicecap entry** *name string*
4. **voice-port** *slot*/*port***:D**
5. **voicecap configure** *name*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voicecap entry`** *`name string`* **Example:** `Router(config)# voicecap entry qualityERL v270=120`	Creates a voicecap. • The *name* argument is a word that uniquely identifies this voicecap. • The *string* argument is a string of characters composed of a series of voicecap register entries, similar to a modemcap. Each entry is of the form vINDEX=VALUE, where INDEX refers to a specific V register, and VALUE designates the value to set that V register to. • For more information about V_register entries, see the "Setting Voice Tuning Parameters with V Registers" section on page 7. • The example creates a simple voicecap string named "qualityERL" with V register 270 set to 120.
Step 4	**`voice-port`** *`slot/port`*`:D` **Example:** `Router(config)# voice-port 3/0:D`	Enters voice-port configuration mode on the selected slot and port.
Step 5	**`voicecap configure`** *`name`* **Example:** `Router(config-voiceport)# voicecap configure qualityERL`	Applies a voicecap. • The *name* argument designates which of the newly created voicecaps to use on this voice port. This character value must be identical to the value entered when you created the voicecap entry. In this case, the value is "qualityERL." **Note** To configure multiple voice ports, repeat Step 4 and Step 5 for each voice port.
Step 6	**`exit`** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Verifying Voicecap Configurations

Use the following **show** commands in privileged EXEC mode to verify your configuration. Relevant fields are shown in bold.

SUMMARY STEPS

1. **show voice port**
2. **show port operational-status** *slot*/*port*
3. **show port voice log**

DETAILED STEPS

Step 1 **show voice port**

Use this command to display configured voicecaps, for example:

```
Router# show voice port

ISDN 2/0:D - 2/0:D
 Type of VoicePort is ISDN
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
.
.
.
 Station name None, Station number None
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Voicecap:EXAMPLE
```

Step 2 **show port operational-status** *slot*/*port*

Use this command to display background noise level information on current calls. Significant fields are shown in bold in the following example:

```
Router# show port operational-status 1/0

Slot/SPE/Port -- 1/0
Service Type                            :Voice service
Voice Codec                             :G.711 u-law
Echo Canceler Length                    :8 ms
Echo Cancellation Control               :Echo cancellation       - enabled
                                          Echo update             - enabled
                                          Non-linear processor    - enabled
                                          Echo reset coefficients - disabled
                                          High pass filter enable - disabled
Digit detection enable                  :DTMF signaling          - enabled
Voice activity detection                :Disabled
Comfort noise generation                :Generate comfort noise
Digit relay enable                      :OOB Digit relay         - disabled
                                          IB Digit relay          - disabled
Information field size                  :20 ms
Playout de-jitter mode                  :adaptive
Encapsulation protocol                  :RTP
Input Gain                              :0.0 dB
Output Gain                             :0.0 dB
Tx/Rx SSRC                              :20/0
```

```
Current playout delay                    :65 ms
Min/Max playout delay                    :65/105 ms
Clock offset                             :142003 ms
Predictive concealment                   :0 ms
Interpolative concealment                :0 ms
Silence concealment                      :0 ms
Buffer overflow discards                 :1
End-point detection errors               :0
Tx/Rx Voice packets                      :1337/1341
Tx/Rx signaling packets                  :0/0
Tx/Rx comfort noise packets              :0/0
Tx/Rx duration                           :26745/26745 ms
Tx/Rx voice duration                     :0/0 ms
Out of sequence packets                  :0
Bad protocol headers                     :0
Num. of late packets                     :0
Num. of early packets                    :1
Tx/Rx Power                            :-87.0/-57.3 dBm
Tx/Rx Talker Level                       :-86.3/-57.0 dBm
TX/RX Mean Speech level                  :-86.3/-57.0 dBm
VAD Background noise level               :6.2 dBm
ERL level                                :127.0 dB
ACOM level                               :127.0 dB
Tx/Rx current activity                   :silence/silence
Tx/Rx byte count                         :213920/214240
ECAN Background noise level              :-83.4 dBm
Latest SSRC value                        :391643394
Number of SSRC changes                   :1
Number of payload violations             :0
```

Step 3 **show port voice log**

Use this command to display background noise level information on completed calls. Significant fields are shown in bold in the following example:

```
Router# show port voice log

Port 1/00 Events Log
  *Aug 22 07:59:27.515:Voice Terminate event:
    Disconnect Reason                   :  normal call clearing (16)
    Call Timer                          :  57  secs
    Current playout delay               :  65  ms
    Min/Max playout delay               :  65/105  ms
    Clock offset                        :  142003  ms
    Predictive concealment              :  0  ms
    Interpolative concealment           :  0  ms
    Silence concealment                 :  0  ms
    Buffer overflow discards            :  1
    End-point detection errors          :  0
    Tx/Rx Voice packets                 :  2813/2816
    Tx/Rx signaling packets             :  0/0
    Tx/Rx comfort noise packets         :  0/0
    Tx/Rx duration                      :  56260/56260  ms
    Tx/Rx voice duration                :  0/0  ms
    Out of sequence packets             :  0
    Bad protocol headers                :  0
    Num. of late packets                :  0
    Num. of early packets               :  1
    Tx/Rx Power                         :  -87.0/-57.3  dBm
    Tx/Rx Mean Speech Level             :  -86.7/-57.0  dBm
    Tx/Rx Talker Level                  :  -86.3/-57.0 dBm
    Average VAD Background noise level :  6.2  dBm
    Average ERL level                   :  127.0  dB
    Average ACOM level                  :  127.0  dB
```

```
    Tx/Rx current activity             :  silence/silence
    Tx/Rx byte count                   :  450080/450240
    Average ECAN Background noise level:  -83.4  dBm
   *Aug 22 07:59:27.515:Voice SSRC change events:
    Latest ssrc value                  :  391643394
    Total ssrc changes                 :  1
```

Troubleshooting NextPort Voicecaps

Use the following **debug** and **show** commands in privileged EXEC mode to debug the application of a voicecap and to check debugging output:

SUMMARY STEPS

1. **debug nextport vsmgr detail**
2. **debug dspapi detail**
3. **show debug**

DETAILED STEPS

Step 1 **debug nextport vsmgr detail**

Use this command to turn on debugging for NextPort voice services, for example:

```
Router# debug nextport vsmgr detail

NextPort Voice Service Manager:
  NP Voice Service Manager Detail debugging is on
.
.
.
```

Step 2 **debug dspapi detail**

Use this command to turn on debugging for DSP API message event details, for example:

```
Router# debug dspapi detail

DSP API:
  DSP API Command debugging is on
  DSP API Detail debugging is on
.
.
.
```

Step 3 **show debug**

Use this command to check voicecap application debugging. The significant field in the output is highlighted in bold in the following example:

```
Router# show debug

NextPort Voice Service Manager:
  NP Voice Service Manager Detail debugging is on

DSP API:
  DSP API Command debugging is on
```

```
  DSP API Detail debugging is on
*Aug 22 08:34:47.399:dspapi [2/1:1 (4)] dsp_init
*Aug 22 08:34:47.399:dspapi [2/1:1 (4)] dsp_voice_config_params:10 params
*Aug 22 08:34:47.399:  [0] ENCAP RTP:t_ssrc=20 r_ssrc=0 t_vpxcc=0 r_vpxcc=0
                        ifp_payload_type=122 sid_support=19 tse_payload=101
seq_num_start=6303 redundancy=0 cc_payload_typ
Router# e=125 fax_payload_type=122 alaw_pcm_switchover=8 mulaw_pcm_switchover=0
dtmf_payload_type=121, nte_rcv_payload_type=101dynamic_payload=0, codec=5
*Aug 22 08:34:47.399:  [1] PO_JITTER:mode=2 initial=60 max=200 min=40 fax_nom=300
*Aug 22 08:34:47.399:  [2] INBAND_SIG:mode=0x1 enable
*Aug 22 08:34:47.399:  [3] ECHO_CANCEL:flags=0x17 echo_len=256
*Aug 22 08:34:47.399:  [4] IDLE_CODE_DET:enable = 0 code=0x0 duration=6000
*Aug 22 08:34:47.403:  [5] GAIN:input=0 output=0
*Aug 22 08:34:47.403:  [6] CNG:
Router# 1
*Aug 22 08:34:47.403:  [7] INFO_FIELD_SIZE:160
*Aug 22 08:34:47.403:  [8] DIGIT_RELAY:2
*Aug 22 08:34:47.403:  [9] VOICECAP:EXAMPLE
*Aug 22 08:34:47.403:dspapi [2/1:1 (4)] dsp_start_service:G711_U (5)
*Aug 22 08:34:47.403:Matched voicecap:v0=0 v1=1
*Aug 22 08:34:47.403:msg length = 0x001D
*Aug 22 08:34:47.403:session ID = 0x006D
*Aug 22 08:34:47.403:   msg tag = 0x0000
*Aug 22 08:34:47.403:    msg ID = 0xF201
```

Configuring the NextPort Dual-Filter G.168 Echo Canceller

The NextPort dual-filter G.168 echo canceller is enabled by default on NextPort platforms in Cisco IOS Release 12.3(11)T and later releases. However, you can adjust the echo canceller tail coverage time at the voice-port interface by completing the following tasks:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *slot*/*port***:D**
4. **echo-cancel coverage** {**24** | **32** | **48** | **64** | **80** | **96** | **112** | **128**}
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice-port** *slot/port*:**D** **Example:** `Router(config)# voice-port 3/0:D`	Enters voice-port configuration mode on the selected slot and port.
Step 4	**echo-cancel coverage** {**24** | **32** | **48** | **64** | **80** | **96** | **112** | **128**} **Example:** `Router(config-voiceport)# echo-cancel coverage 64`	Adjusts the size of the echo canceller (EC) and selects the extended EC when the Cisco default EC is present. • Starting with Cisco IOS Release 12.4(20)T, the default coverage time and maximum possible coverage is 128 ms for both 8.x and 10.2.2 SPEware versions.
Step 5	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode and completes the configuration.

Configuration Examples for Nextport-Based Voice Tuning and the NextPort Dual-Filter G.168 Echo Canceller

The following sections contain examples that could be used to optimize a NextPort-based gateway for a given telephone network:

High ERL in the Network: Example

Register v270 is used to set the limit for the minimum expected ERLs that the gateway will encounter. If the gateway encounters ERLs that are lower than the v270 setting, the echo canceller performance will be suboptimal.

The default setting for v270 is 6 dB. This setting should work well for usual telephone networks. However, when the gateway is used on a well-managed telephone network with organized loss plans in place, the ERL is often greater than 6 dB. In these cases, v270 can be raised. Making this change reduces any clipping of the near-end signal; however, it will underperform if a low ERL is encountered.

In this example, the network is designed to have an ERL of 12 dB or greater. In this case, the following voicecap may improve performance. Notice that the value for v270 is entered as decibels multiplied by 10.

```
Router> enable
Router# configure terminal
Router(config)# voicecap entry qualityERL v270=120
Router(config)# end
```

Low ERL in the Network: Example

Unlike the scenario described in the previous example, some telephone networks may not always produce sufficient ERLs to meet the default setting of 6 dB. The best way to solve this problem would be to institute better loss plans on the telephone network. However, because this is not always possible, a voicecap can be used to alleviate the problem.

A low ERL means that the echo canceller must do a much deeper cancellation to remove sufficient echo. Also, lowering the minimum ERL can increase the incidence of clipping. The best way to improve this situation is to make up for the telephone network's lack of loss plan by adding some loss in the gateway. If the lowest ERL seen is 4 dB, adding 1 dB of output attenuation and –1 dB of input gain will ensure that the echo canceller never sees more than a 6-dB effective ERL.

Adding this attenuation can be done by entering a voicecap, but using the CLI is the recommended approach in this example. The following commands set the gains that are needed for this example:

```
Router> enable
Router# configure terminal
Router(config)# voice-port 3/0:D
Router(config-voiceport)# output attenuation 1
Router(config-voiceport)# input gain -1
Router(config-voiceport)# end
```

Clipped or Squelched Speech and Low ERL in the Network: Example

In this example, a network in which signal level imbalance is already causing clipping to occur with a MinERL setting of 6 dB and in which ERLs of less than 6 dB are already occurring, a dual approach must be taken. To stop the clipping, the MinERL setting should be lowered to 12 dB. If 4-dB ERLs are occurring, 4 dB of attenuation must be added to the input and the output to ensure that there is 12 dB of effective ERL at the echo canceller (4 dB of real ERL, plus 4 dB of output attenuation, plus 4 dB of input attenuation equals 12 dB of effective ERL). To create these settings, the following commands are used:

```
Router> enable
Router# configure terminal
Router(config)# voicecap entry qualityERL v270=120
Router(config)# voice-port 3/0:D
Router(config-voiceport)# voicecap configure qualityERL
Router(config-voiceport)# output attenuation 4
Router(config-voiceport)# input gain -4
Router(config-voiceport)# end
```

Dynamic Attenuation: Example

It is possible that worst-case settings are only required when the primary telephone circuits are all used and an alternate carrier with a poor loss plan must be used instead. For these cases, the dynamic attenuation feature removes the attenuation when the ERL is sufficient. In the following example, 4 dB of input and output attenuation is to be removed when it is not necessary to ensure the minimum ERL setting. To do this, the dynamic attenuation feature (using v289) is enabled. The required ERL must be set before attenuation is removed (using v290), and minimum attenuation levels for the input and output (using v291 and v292) must also be set. In this example, attenuation is set to 15 dB and then removed.

```
Router> enable
Router# configure terminal
Router(config)# voicecap entry dynatten v270=120 v289=1 v290=15 v291=0 v292=0
Router(config)# voice-port 3/0:D
Router(config-voiceport)# voicecap configure dynatten
Router(config-voiceport)# output attenuation 4
Router(config-voiceport)# input gain -4
```

```
Router(config-voiceport)# end
```

Echo Canceller Tail Coverage: Example

The following example adjusts echo canceller tail coverage to 64 ms on the Cisco AS5400:

```
Router(config) voice-port 1/0:0
Router(config-voiceport)# echo-cancel coverage 64
```

Enabling NextPort Echo Canceller Control for G.711 Encoded VoIP Packets

This section describes how to enable NextPort echo canceller control on the Cisco AS5350, AS5400, AS5400HPX, and AS5850 universal gateways when these gateways detect 2100 Hz tones, received in G.711 encoded VoIP packets. You can enable NextPort voicecaps to control the echo canceller from either the PSTN or IP side of the network.

Note NextPort control over the echo canceller is possible only in G.711 codec modes. Cisco recommends that you do not enable NextPort control over the echo canceller in conjunction with modem pass-through.

IP tone detection and NextPort control over the echo canceller is enabled using the command-line interface. Use the following commands to enable NextPort control over the echo canceller by creating a voicecap entry and applying it to the voice port.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voicecap entry** *name string*
4. **voice-port** *slot/port*
5. **voicecap configure** *name*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	`voicecap entry` *name string* **Example:** `Router(config)# voicecap entry npecho_ctrl v2=512 v51=32769`	Creates a voicecap entry. • The *name* argument is a word that uniquely identifies this voicecap. • The *string* argument is a series of voicecap register entries, similar to a modemcap. Each entry is of the form vINDEX=VALUE, where INDEX refers to a specific V register, and VALUE designates the value to which the V register should be set. • Example settings: – The **v2=512** setting enables the 250-ms silence detection. This setting is optional. When this setting is used in conjunction with the **v51 = 32769** setting, NextPort restores the echo canceller to its original state after it detects the 250-ms silence. – The **v51=32769** setting enables IP side tone detection/notification and allows NextPort to disable the NLP or the echo canceller upon reception of 2100 Hz answer tones from the IP side. This setting is required in Cisco IOS Release 12.3T.
Step 4	`voice-port` *slot/port* **Example:** `Router(config)# voice-port 3/0`	Enters voice-port configuration mode on the selected slot and port.
Step 5	`voicecap configure` *name* **Example:** `Router(config-voiceport)# voicecap configure npecho_ctrl`	Applies a voicecap entry to the voice port. • The *name* argument designates which of the newly created voicecaps to use on this voice port. This character value must be identical to the value entered when you created the voicecap entry. **Note** To configure multiple voice ports, repeat Step 4 and Step 5 for each voice port.

Step 6	`exit`
	
	Example:
	`Router(config-voiceport)# exit`
		Exits voice-port configuration mode.

Troubleshooting NextPort Echo Canceller Control for G.711 Encoded VoIP Packets

You can display the EST trace messages that show the tone detections and the resultant echo operations if you enter the **debug trace module f080 0010** *s/d/m* command. NextPort enables and disables the NLP and the echo canceller based on reception of 2100 Hz answer tones from the IP side or PSTN side and generates EST trace messages for each tone detected and its echo operation. NextPort also detects the 250 ms of silence and generates EST trace messages to indicate such detection and to indicate that the echo state has been restored.

Step 1 Router# **debug trace module f080 0010 s/d/m**

Use this command to display the EST trace messages. In the command, **s/d/m** is defined as follows:

- s = slot
- d = dfc
- m = module number

When the default configuration values for Index 51 and Index 52 are used, IP tone detection and notification are disabled, and all existing features continue to function normally.

The following example shows EST trace messages collected from the console:

```
Router#
*Apr 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:      Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:51.735:      Data Len   : 56
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received Early ANS tone 0x01 from
IP side
*Apr 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:
Router#  Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:51.735:      Data Len   : 63
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received Tone Off ntf for code 0x01
from IP side
*Apr 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:      Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
Router#*Apr 26 21:40:51.735:      Data Len   : 45
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received ANS tone 0x03 from IP
*Apr 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:      Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:51.735:      Data Len   : 47
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Non-linear Processor Is Disabled
*Apr
Router# 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:      Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:51.735:      Data Len   : 63
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received Tone Off ntf for code 0x03
from IP side
*Apr 26 21:40:51.735:   00:00:14: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:
Router#   Address    : 0x3000000
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:51.735:      Data Len   : 47
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received ANSam tone 0x07 from IP
*Apr 26 21:40:51.735:   00:00:13: Port Trace Event:
*Apr 26 21:40:51.735:      Port       : 3/00
*Apr 26 21:40:51.735:      Address    : 0x3000000
```

```
*Apr 26 21:40:51.735:      Trace Event: 0x2
*Apr 26 21:40:51.735:      Data Format: ASCII
*Apr 26 21:40:5
Router#1.735:      Data Len   : 63
*Apr 26 21:40:51.735:      Data       : Session 0x0144 Received Tone Off ntf for code 0x07
from IP side
*Apr 26 21:40:51.739:   00:00:13: Port Trace Event:
*Apr 26 21:40:51.739:      Port       : 3/00
*Apr 26 21:40:51.739:      Address    : 0x3000000
*Apr 26 21:40:51.739:      Trace Event: 0x2
*Apr 26 21:40:51.739:      Data Format: ASCII
*Apr 26 21:40:51.739:      Data Len   : 48
*Apr 26 21:40:51.739:      Data       : Session 0x0144 Received /ANSam tone 0x0f from IP
Router#*Apr 26 21:40:51.739:   00:00:13: Port Trace Event:
*Apr 26 21:40:51.739:      Port       : 3/00
*Apr 26 21:40:51.739:      Address    : 0x3000000
*Apr 26 21:40:51.739:      Trace Event: 0x2
*Apr 26 21:40:51.739:      Data Format: ASCII
*Apr 26 21:40:51.739:      Data Len   : 31
*Apr 26 21:40:51.739:      Data       : Session 0x0144 ECAN Is Disabled
*Apr 26 21:40:51.739:   00:00:04: Port Trace Event:
*Apr 26 21:40:51.739:      Port       : 3/00
*Apr 26 21:40:51.739:      Address    : 0x3000000
Router#*Apr 26 21:40:51.739:      Trace Event: 0x2
*Apr 26 21:40:51.739:      Data Format: ASCII
*Apr 26 21:40:51.739:      Data Len   : 63
*Apr 26 21:40:51.739:      Data       : Session 0x0144 Received Tone Off ntf for code 0x0f
from IP side

*Apr 26 21:46:36.431:   00:00:08: Port Trace Event:
*Apr 26 21:46:36.431:      Port       : 3/00
*Apr 26 21:46:36.431:      Address    : 0x3000000
*Apr 26 21:46:36.431:      Trace Event: 0x2
*Apr 26 21:46:36.431:      Data Format: ASCII
*Apr 26 21:46:36.431:      Data Len   : 43
*Apr 26 21:46:36.431:      Data       : Session 0x0144 detected 250 msec of silence
*Apr 26 21:46:36.431:   00:00:08: Port Trace Event:
*Apr 26 21:46:36.431:      Port       : 3/00
*Apr 26 21:46:36.431:      Address    : 0x3000000
*Apr 26 21:46:36.435:      Trace Event: 0x2
*Apr 26 21:46:36.435:      Data Format: ASCII
*Apr 26 21:46:36.435:      Data Len   : 41
*Apr 26 21:46:36.435:      Data       : Session 0x0144 Ecan State 0x0007 Restored
```

Verifying Analog and Digital Voice-Port Configurations

This chapter describes some basic procedures and specific CLI commands you can use to verify the set up and configuration of the analog and digital voice ports on the routers in your voice network.

Information About Verifying Analog and Digital Voice-Port Configurations

After configuring the voice ports on your router, there are two simple verifications you can use to check the operation of the telephony handset. There are also eight CLI commands you can use to verify proper operation of the configuration.

How to Verify Analog and Digital Voice-Port Configurations

To verify the operability of the analog and digital voice-port configurations on your voice network, complete the following tasks.

SUMMARY STEPS

1. Check for a dial tone.
2. Check for DTMF detection.
3. **show voice port summary**
4. **show voice port**
5. **show running-config**
6. **show controller**
7. **show voice dsp**
8. **show voice call summary**
9. **show call active voice**

10. **show call history voice**

DETAILED STEPS

Step 1 Check for a dial tone.

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

Step 2 Check for Dual-Tone Multifrequency (DTMF) detection.

If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is probably configured properly.

Step 3 **show voice port summary**

Use this command to identify the port numbers of voice interfaces installed in your router. For examples of the output, see the "Examples" section on page 3.

Step 4 **show voice port**

Use this command to verify voice-port parameter settings. Refer to Table 2 for the appropriate syntax for your platform. For sample output, see the "show voice port Command: Examples" section on page 4.

Table 2 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 Cisco 3700 series	Analog	**show voice port** [*slot*/*port* | **summary**]
	Digital	**show voice port** [*slot*/*port*:*ds0-group-number* | **summary**]
Cisco MC3810	Analog	**show voice port** [*slot*/*port* | **summary**]
	Digital	**show voice port** [*slot*:*ds0-group-number* | **summary**]
Cisco AS5300	Digital	**show voice port** [*controller*:{*ds0-group-number* | **D**}] [**summary**]
Cisco AS5350 Cisco AS5400 Cisco AS5850	Digital	**show voice port** [*slot*/*controller*:{*ds0-group-number* | **D**}] [**summary**]
Cisco AS5800	Digital	**show voice port** {*shelf*/*slot*/*port*:*ds0-group-number*}
Cisco 7200 series	Digital	**show voice port** {*slot*/*port-adapter*:*ds0-group-number*}
Cisco 7500 series	Digital	**show voice port** {*slot*/*port-adapter*/*slot*:*ds0-group-number*}

Step 5 **show running-config**

Use this command to verify the codec complexity setting for digital T1/E1 connections. If medium complexity is specified for the voice card, the codec complexity command is not displayed. If high complexity is specified, the **codec complexity high** command is displayed. The following example shows output when high complexity is specified:

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

voice-card 0
```

```
 codec complexity high
.
.
.
```

Step 6 **show controller**

Use this command to verify that the digital T1/E1 controller is up and that no alarms have been reported, and to display information about clock sources and other controller settings. For output examples, see the "show controller Command: Examples" section on page 7.

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

Step 7 **show voice dsp**

Use this command to display voice-channel configuration information for all DSP channels. For output examples, see the "show voice dsp Command: Examples" section on page 8.

```
Router# show voice dsp
```

Step 8 **show voice call summary**

Use this command to verify the call status for all voice ports. For output examples, see the "show voice call summary Command: Examples" section on page 9.

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

Step 9 **show call active voice**

Use this command to display the contents of the active call table, which shows all of the calls currently connected through the router or concentrator. For output examples, see the "show call active voice Command: Example" section on page 10.

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

Step 10 **show call history voice**

Use this command to display the contents of the call history table. To limit the display to the most recent calls connected through this router, use the **last** keyword and define the number of calls to display with the *number* argument. 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 11.

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

Examples

This section contains output examples for the following commands on different platforms and for different configurations:

- show voice port summary Command: Examples
- show voice port Command: Examples
- show controller Command: Examples
- show voice dsp Command: Examples
- show voice call summary Command: Examples
- show call active voice Command: Example

- show call history voice Command: Example

show voice port summary Command: Examples

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 Digital Voice Port

The following output is from a Cisco MC3810:

```
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

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

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 Foreign Exchange Service (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 T1 CAS Voice Port

The following output is from a Cisco AS5300 T1 channel-associated signaling (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

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 E1 Controller

The following output is from a Cisco MC3810 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 T1 Controller

The following output is from a Cisco AS5800 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

Digital Voice Port on a Cisco 3640

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

Cisco MC3810 Analog Voice Port

The following output is from a Cisco MC3810:

```
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

Cisco 7200 Series Router

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=172.16.235.18

RemoteUDPPort=16580

RoundTripDelay=29 ms

SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:172.16.235.18
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

Cisco 7200 Series Router

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=172.16.235.18

RemoteUDPPort=16552

RoundTripDelay=23 ms

SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:172.16.235.18
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

This chapter provides information to assist you in analyzing and troubleshooting voice port problems.

Troubleshooting Chart

Table 1 lists some problems that you might encounter after configuring voice ports. It also provides some suggested remedies.

Table 1 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 in the Verifying Analog and Digital Voice-Port Configurations chapter for output.

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Table 1 Troubleshooting Voice Port Configurations (continued)

Problem	Suggested Action
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 in the Verifying Analog and Digital Voice-Port Configurations chapter for sample output.
Telephony device buzzes or does not ring	Use the **show voice port** command to confirm that the **ring frequency** command is configured correctly. It must match the connected telephony equipment and may be country-dependent.
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 configured level for the **music-threshold** command.
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 milliseconds (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. (Refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.)
Clipped or fuzzy speech	• Reduce input gain. (Refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.) • Change the voice activity detection (VAD) level. Sometimes VAD cuts the sound too early and the speaker's voice is clipped. You can also change the time that VAD waits for silence.

Table 1 Troubleshooting Voice Port Configurations (continued)

Problem	Suggested Action
Clipped speech	Reduce the input level at the listener's router. (Refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.)
Volume too low or missed Dual-Tone Multifrequency (DTMF)	Increase speaker's output level or listener's input level. (Refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.)
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 "Configuring Echo Cancellation" section on page 75.)
Too much echo	Reduce the output level at the speaker's voice port. (Refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.)

Note For information on **test** commands that force voice ports into specific states for testing refer to the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.