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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.

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 Feature Information, page 1
- Voice Port Configuration Overview, page 1

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

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.

- The "Telephone to WAN" figure shows one voice port connecting a telephone to the WAN through the router.
- The "Telephone to PSTN" figure shows one voice port connected to the PSTN and another to a telephone; the router acts like a small PBX.
- The "PBX-to-PBX over a WAN" figure 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. The table below 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

<table>
<thead>
<tr>
<th>Platform</th>
<th>Analog</th>
<th>Digital</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco 880 series (includes IAD881B, IAD881F, C881SRST, IAD888B, IAD888F, and C888SRST)</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco 1750</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco 3600 series</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco 7200 series</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco 7500 series</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco AS5300</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco AS5350</td>
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<td>Yes</td>
</tr>
<tr>
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<td>Yes</td>
</tr>
<tr>
<td>Cisco AS5850</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Cisco MC3810</td>
<td>Yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>

### 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 "FXS and FXO Interfaces" and the Telephony Signaling Interfaces. 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 the "FXS Signaling Interfaces" figure, FXS signaling is used for end-user telephony equipment, such as a telephone or fax machine. The "FXS and FXO Signaling Interfaces" figure 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 the "E&M Signaling Interfaces" figure, 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 and 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 telecommunication 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 and 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 telecommunication, 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 the table below. Cisco voice implementation supports E&M Types I, II, III, and V.
### Table 2: E&M Wiring and Signaling Methods

<table>
<thead>
<tr>
<th>E&amp;M Type</th>
<th>E-Lead Configuration</th>
<th>M-Lead Configuration</th>
<th>Signal Battery Lead Configuration</th>
<th>Signal Ground Lead Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>Output, relay to ground</td>
<td>Input, referenced to ground</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>II</td>
<td>Output, relay to SG</td>
<td>Input, referenced to ground</td>
<td>Feed for M, connected to -48V</td>
<td>Return for E, galvanically isolated from ground</td>
</tr>
<tr>
<td>III</td>
<td>Output, relay to ground</td>
<td>Input, referenced to ground</td>
<td>Connected to -48V</td>
<td>Connected to ground</td>
</tr>
<tr>
<td>V</td>
<td>Output, relay to ground</td>
<td>Input, referenced to -48V</td>
<td>--</td>
<td>--</td>
</tr>
</tbody>
</table>

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 "DSO 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.

- Finding Feature Information, page 9
- Prerequisites for Configuring Analog Voice Ports, page 9
- Information About Analog Voice Hardware, page 10
- How to Configure Analog Voice Ports, page 13

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

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, on page 10.

• 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 Guides. More current information may be available in the release notes for the Cisco IOS software you are using.

---

**Information About Analog Voice Hardware**

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

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**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.
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 occurring 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 Cisco 3600 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:

- Cisco 2600 Series Routers Hardware Installation Guide
- Cisco 3600 Series Routers Hardware Installation Guide
- 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.

Basic Parameters on Analog FXO FXS or E and 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, on page 9 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 on the Cisco MC3810, on page 17.

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 Configuration"
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.

How to Configure Analog Voice Ports

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

Perform this task to configure basic parameters:
**SUMMARY STEPS**

1. enable
2. configure terminal
3. Do one of the following:
   - `voice-port slot / port`
4. Do one of the following:
   - `signal {loop-start | ground-start}`
5. `cptone locale`
6. `dial-type {dtmf | pulse}`
7. `operation {2-wire | 4-wire}`
8. `type {1 | 2 | 3 | 5}`
9. Do one of the following:
   - `ring frequency {25 | 50}`
   - ``
   - `ring frequency {20 | 30}`
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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Do one of the following:</td>
<td>Enters voice-port configuration mode.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td>• voice-port slot / port</td>
<td>Note: The slash must be entered between slot and port. Valid entries vary by router platform; enter the <code>show voice port summary</code> command for available values.</td>
</tr>
<tr>
<td>Example: Router(config)# voice-port 1/0</td>
<td>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.</td>
</tr>
<tr>
<td>voice-port slot/subunit/port</td>
<td></td>
</tr>
<tr>
<td>Example: Router(config)# voice-port 1/0/0</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Do one of the following:</td>
</tr>
<tr>
<td>• signal {loop-start</td>
<td>ground-start}</td>
</tr>
<tr>
<td>Example: Router(config-voiceport)# signal ground-start</td>
<td>Note: Configuring the <code>signal</code> 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.</td>
</tr>
<tr>
<td>signal {wink-start</td>
<td>immediate-start</td>
</tr>
<tr>
<td>Example: Router(config-voiceport)# signal wink-start</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td><code>cptone locale</code></td>
</tr>
<tr>
<td>Example: Router(config-voiceport)#cptone us</td>
<td>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 <code>cptone</code> command. The default is <code>us</code>.</td>
</tr>
<tr>
<td>Step 6</td>
<td>`dial-type {dtmf</td>
</tr>
<tr>
<td>Example: Router(config-voiceport)# dial-type dtmf</td>
<td></td>
</tr>
<tr>
<td>Step</td>
<td>Command or Action</td>
</tr>
<tr>
<td>------</td>
<td>------------------</td>
</tr>
</tbody>
</table>
| 7    | operation {2-wire | 4-wire} | (E&M only) Specifies the number of wires used for voice transmission at this interface (the audio path only, not the signaling path).  
  - The default is 2-wire. |
|      | Example: Router(config-voiceport)# operation 4-wire |         |
| 8    | type {1 | 2 | 3 | 5} | (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. |
|      | Example: Router(config-voiceport)# type 2 |         |
| 9    | Do one of the following:  
  - ring frequency {25 | 50}  
  -  
  - ring frequency {20 | 30} | (FXS only) Selects the ring frequency, in hertz, used on the FXS interface. This number must match the connected telephony equipment and may be country-dependent. If 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.  
  -  
  -  |
|      | Example: Router(config-voiceport)# ring frequency 50 |         |
|      | Example: Router(config-voiceport)# ring frequency 30 |         |
| 10   | ring number number | (FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router.  
  - The default is 1. |
|      | Example: Router(config-voiceport)# ring number 1 |         |
| 11   | ring cadence {{pattern01 | pattern02 | pattern03 | pattern04 | pattern05 | pattern06 | pattern07 | pattern08 | pattern09 | pattern10 | pattern11 | pattern12} | [define pulse interval]} | (FXS only) Specifies an existing pattern for ring, or defines a new one. Each pattern specifies a ring-pulse time and a ring-interval time.  
  - The default is the pattern specified by the cптone locale that has been configured. |
|      | Example: Router(config-voiceport)# ring cadence pattern01 |         |
### Configuring Codec Complexity on the Cisco MC3810

To configure codec complexity for analog voice ports on the Cisco MC3810 using High-Performance Compression Modules (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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1**

  - enable
  - Example:
    
      ```
      Router> enable
      ```

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 2**

  - show voice dsp
  - Example:
    
      ```
      Router# show voice dsp
      ```

  Enables privileged EXEC mode.

  - Enter your password if prompted.

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.
## Configuring Codec Complexity on the Cisco MC3810

### Command or Action

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 3</td>
<td><code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td><code>voice-card 0</code></td>
<td>Enters voice-card configuration mode and specifies voice card 0.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice-card 0</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td>`codec complexity {high</td>
<td>medium}`</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voicecard)# codec complexity high</td>
<td></td>
</tr>
</tbody>
</table>
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.

- Finding Feature Information, page 20
- Prerequisites for Configuring Digital Voice Ports, page 20
- Information About Digital Voice Hardware, page 22
- How to Configure Digital T1 E1 Voice Ports, page 28

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

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, on page 22.

- (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.
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 3620 and Cisco 3610 series routers) 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

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
```
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.

**Information About Digital Voice Hardware**

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.

The C881 and C888 SRST models automatically detect a failure occurring 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.

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.
Cisco 2600 Cisco 3600 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
- T1/E1 Moderate-Capacity and High-Capacity Digital Voice Port Adapter Installation and Configuration

---

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, on page 40).

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

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 the table below.

Table 3: CMM Port Adapters

<table>
<thead>
<tr>
<th>CMM Port Adapters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>• WS-SVC-CMM-6T1</td>
<td>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.</td>
</tr>
<tr>
<td>• WS-SVC-CMM-6E1</td>
<td>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.</td>
</tr>
<tr>
<td>WS-SVC-CMM-24FXS</td>
<td>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.</td>
</tr>
</tbody>
</table>
CMM Port Adapters | Description
--- | ---
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, on page 20 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

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:

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. 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 page 29.

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>enable</strong></td>
</tr>
</tbody>
</table>
### Configuring Codec Complexity on Digital T1 E1 Voice Ports

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
</tbody>
</table>

**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

To change codec complexity on Cisco 880 Series, Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, and Cisco MC3810 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. Do one of the following:
   • no ds0-group ds0-group-number
   •
   •
   • 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. Do one of the following:
   • 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}
   •
   •
   • pri-group timeslots timeslot-list
14. exit
15. Repeat Step 3, then continue with Step 16.
16. no shutdown
17. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
</tbody>
</table>

- Enter your password if prompted.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice-port slot /port:ds0-group-number</td>
<td>Enters voice-port configuration mode on the selected slot, port, and DS0 group.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# voice-port 1/0:23</td>
<td></td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> The syntax of this command is platform-specific. For the syntax for your platform, refer to theCisco IOS Voice Command Reference.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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.</td>
</tr>
<tr>
<td><strong>Step 4</strong> shutdown</td>
<td>Shuts down all voice ports assigned to the T1 interface on the voice card.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-voiceport)# shutdown</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> exit</td>
<td>Exits voice-port configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-voiceport)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> controller {t1</td>
<td>e1} slot/port</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# controller t1 1/0</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> Do one of the following:</td>
<td>Removes the related DS0 groups, or Removes the related PRI group.</td>
</tr>
<tr>
<td>• no ds0-group ds0-group-number</td>
<td></td>
</tr>
<tr>
<td>•</td>
<td></td>
</tr>
<tr>
<td>• no pri-group timeslots timeslot-list</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-controller)# no ds0-group 1</td>
<td></td>
</tr>
</tbody>
</table>
### Purpose

**Command or Action**

**Example:**

```bash
Router(config-controller)# no pri-group
timeslots 1,7,9
```

**Step 8**

**exit**

**Example:**

```bash
Router(config-controller) exit
```

Exits controller configuration mode and returns to global configuration mode.

**Step 9**

**voice-card slot**

**Example:**

```bash
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.

**Step 10**

**codec complexity**

- **high**
- **medium**

**[ecan-extended]**

**Example:**

```bash
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:**

```bash
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**

Do one of the following:

- **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**

**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.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>•</td>
<td>Specifies an ISDN PRI on a channelized T1 or E1 controller.</td>
</tr>
<tr>
<td>•</td>
<td><strong>Note</strong> When configuring PRI groups, you must also configure the <strong>isdn switch-type</strong> command. Also, only one PRI group can be configured on a controller.</td>
</tr>
<tr>
<td>• pri-group timeslots  <em>timeslot</em> - list</td>
<td></td>
</tr>
</tbody>
</table>

**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
```

**Step 14**

`exit`

**Example:**

```
Router(config-controller)# exit
```

**Step 15**

Repeat Step 3, then continue with Step 16.

**--**

**Step 16**

`no shutdown`

**Example:**

```
Router(config-controller)# no shutdown
```

**Step 17**

`end`

**Example:**

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

### Configuring the Flex Option on Codec Complexity

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

On the Cisco 2600XM, 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).

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. The table below summarizes the codec combinations for DSP-sharing.

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.

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.

<table>
<thead>
<tr>
<th>Local DSP Resource (Import)</th>
<th>Remote DSP Resource (Export)</th>
</tr>
</thead>
<tbody>
<tr>
<td>High complexity</td>
<td>Medium complexity</td>
</tr>
<tr>
<td>High complexity</td>
<td>Yes</td>
</tr>
<tr>
<td>Medium complexity</td>
<td>Yes</td>
</tr>
<tr>
<td>Flexible complexity</td>
<td>Yes</td>
</tr>
</tbody>
</table>

Table 4: Codec Complexity Settings for DSP Resource Sharing Between Local and Remote Sources

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice-card slot</td>
<td>Enters voice-card configuration mode and specifies the slot location.</td>
</tr>
<tr>
<td>Example: Router(config)# voice-card 1</td>
<td>• For the slot argument, specify a value from 1 to 4, depending on your router.</td>
</tr>
<tr>
<td>Step 4 codec complexity flex [reservation - fixed {high</td>
<td>medium}]</td>
</tr>
<tr>
<td>Example: Router(config-voicecard)# codec complexity flex</td>
<td>• 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 applications 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Step 5 voice local-bypass</td>
<td>Configures local calls to bypass the DSP. This is the default.</td>
</tr>
</tbody>
</table>

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 page 29.
### Configuring Codec Complexity

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

**Note**
Use the `show interfaces dsfpfarm` 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. Do one of the following:
   - `dspint dsfpfarm  slot /0`
   - `dspint dsfpfarm  slot /port-adapter /port`
4. `codec {high | medium} [ecan-extended]`
5. `exit`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>

---

**Example:**

```bash
Router(config-voicecard)# voice local-bypass
```

- 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>exit</code></td>
<td>Exits voice-card configuration mode and returns the router to global configuration mode.</td>
</tr>
</tbody>
</table>

**Example:**

```bash
Router(config-voicecard)# exit
```
### Command or Action

- **Example:**
  
  ```
  Router> enable
  ```

### Purpose

- Enter your password if prompted.

#### Step 2

**configure terminal**

- **Example:**
  
  ```
  Router# configure terminal
  ```

### Purpose

- Enters global configuration mode.

#### Step 3

Do one of the following:

- **Example:**
  
  ```
  Router(config)# dspint dspfarm 2/0
  ```

### Purpose

- Enters DSP interface configuration mode for the Cisco 7200 series.
- Enters DSP interface configuration mode for the Cisco 7500 series.

#### Step 4

**codec {high | medium} [ecan-extended]**

- **Example:**
  
  ```
  Router(config-dspfarm)# codec medium ecan-extended
  ```

### Purpose

- 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
  ```

### Purpose

- Exits to global configuration mode.

---

**What to Do Next**

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.

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.

The figure below 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.
All controller commands shown in the figure below, other than ds0-group, apply to all time slots in the T1 line.

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

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 because its 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 the figure below. 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 8: 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 the figure below). The PLL clocking is driven by the clock reference on the receive (Rx) side of the digital line connection.

**Figure 9: 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 the figure below.

**Figure 10: 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.
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):

```plaintext
controller E1 1/0
  framing crc4
  linecoding hdb3
  clock source line primary
do-config timeslots 1-15 type e&m-wink-start
controller E1 1/1
  framing crc4
  linecoding hdb3
  clock source line
do-config 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.

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 the figure below). If controller E1 0 fails, the PLL internally generates the clock reference to drive E1 1.

![Figure 11: Dual E1 Ports--One Receiving and One Providing Clocking](image)

The following configuration sets up this clocking method:

```plaintext
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 the figure below).

![Figure 12: Dual E1 Ports--Both Clocks from the Router](image)

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 &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 &m-wink` command creates logical voice port 0:1.

To map individual DS0s, define additional DS0 groups under the T1/E1 controller, specifying different timeslots. 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:
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.

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.

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
4. Do one of the following:
   • controller \{t1 | e1\} slot / port
   • controller \{t1 | e1\} number
   • controller \{t1 | e1\} shelf / slot / port
5. Do one of the following:
   • framing \{sf | esf\}
   • crc4 | no-crc4 [australia]
6. clock source \{line [primary | secondary] | internal\}
7. Do one of the following:
   • linecode \{ami | b8zs\}
   • 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-cana | fxo-ground-start | fxo-loop-start | fxs-ground-start | fxs-loop-start\}
9. no shutdown

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 3</strong> card type {t1</td>
<td>e1} slot</td>
</tr>
<tr>
<td>Example: Router(config)# card type t1 0</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> Do one of the following: • controller {t1</td>
<td>e1} slot / port</td>
</tr>
<tr>
<td>• controller {t1</td>
<td>e1} number</td>
</tr>
<tr>
<td>• controller {t1</td>
<td>e1} shelf / slot / port</td>
</tr>
<tr>
<td>Example: Router(config)# controller t1 1/0</td>
<td>• For the Cisco AS5800 and Cisco 7500 series, identifies the shelf, slot, and port number.</td>
</tr>
<tr>
<td>Example: Router(config)# controller t1 1</td>
<td></td>
</tr>
<tr>
<td>Example: Router(config)# controller t1 1/0/0</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> Do one of the following: • framing {sf</td>
<td>esf}</td>
</tr>
<tr>
<td>• framing {crc4</td>
<td>no-crc4} [australia]</td>
</tr>
<tr>
<td>Example: Router(config-controller)# framing esf</td>
<td>• For E1, the frame type can be crc4 or no crc4 or australia. Default for E1 is crc4.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# framing crc4</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> clock source {line [primary</td>
<td>secondary]</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# clock source line primary</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> Do one of the following:</td>
<td>Specifies the line encoding to use for T1 or E1 line.</td>
</tr>
<tr>
<td>• linecode {ami</td>
<td>b8zs}</td>
</tr>
<tr>
<td>• linecode {ami</td>
<td>hdb3}</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# linecode b8zs</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# linecode hdb3</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> ds0-group ds0-group-number timeslots timeslot-list type {e&amp;m-delay-dial</td>
<td>e&amp;m-fgd</td>
</tr>
<tr>
<td>Note</td>
<td>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.</td>
</tr>
</tbody>
</table>
**Step 9**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>no shutdown</td>
<td>Activates the controller.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config-controller)# no shutdown
```

---

**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.

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.

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

For more information on 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.

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

1. enable
2. configure terminal
3. Do one of the following:
   - voice-port port
   - voice-port slot / port:ds0-group-number
   - voice-port slot / port-adapter:ds0-group-number
   - voice-port slot / port-adapter/slot:ds0-group-number
   - voice-port controller:{ds0-group-number | D}
   - voice-port slot / controller:{ds0-group-number | D}
   - 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>----------------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
</tbody>
</table>

**Step 3**

Do one of the following:

- `voice-port port`
- `voice-port slot / port:ds0-group-number`
- `voice-port slot / port-adapter:ds0-group-number`
- `voice-port slot / port-adapter/slot:ds0-group-number`
- `voice-port controller:{ds0-group-number | D}`
- `voice-port slot / controller:{ds0-group-number | D}`
- `voice-port shelf / slot / port:ds0-group-number`

Example:

Router(config)# voice-port 1:0

Example:

Router(config)# voice-port 1/1:0

Example:

Router(config)# voice-port 1/1/1
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice-port 1:1</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice-port 1/0 D</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice-port 1/2/0:1</td>
<td></td>
</tr>
</tbody>
</table>

**Step 4**

| type {1 | 2 | 3 | 5} | (E&M only) Specifies the type of E&M interface to which this voice port is connected. See Table 3 in the "Voice Port Configuration Overview" chapter for an explanation of E&M types. |
|--------|---------------------------------------------------------------|
| Example: | Router(config-voiceport)# type 1                           |

**Step 5**

<table>
<thead>
<tr>
<th>cptone locale</th>
<th>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.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# cptone us</td>
</tr>
</tbody>
</table>

**Step 6**

| compand-type {u-law | a-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. |
|-------------|----------------------------------------|
| Example:    | Router(config-voiceport)# compand-type 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.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 7</strong></td>
<td></td>
</tr>
</tbody>
</table>
| `ring frequency` {25 | 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. |
| Example:               |                                                                          |
| `Router(config-voiceport)# ring frequency 50` |                                                                          |
| **Step 8**             |                                                                          |
| `ring number number`  | (FXO only) Specifies the maximum number of rings to be detected before an incoming call is answered by the router.  
  - Default is 1. |
| Example:               |                                                                          |
| `Router(config-voiceport)# ring number 1` |                                                                          |
| **Step 9**             |                                                                          |
| `ring cadence {[pattern01 | pattern02 | pattern03 | pattern04 | pattern05 | pattern06 | pattern07 | pattern08 | pattern09 | pattern10 | pattern11 | pattern12} [define pulse interval]` | (FXS only) Specifies an existing pattern for ring, or defines a new one. Each pattern specifies a ring-pulse time and a ring-interval time. The keywords and arguments are as follows:  
  - `pattern01` through `pattern12`--Specifies preset ring cadence patterns. Enter `ring cadence ?` to see ring pattern explanations.  
  - `define pulse interval` --Specifies a user-defined pattern as follows:  
    - `pulse` is a number (1 or 2 digits from 1 to 50) specifying ring pulse (on) time in hundreds of milliseconds.  
    - `interval` is a number (1 or 2 digits from 1 to 50) specifying ring interval (off) time in hundreds of milliseconds.  
    - The default is the pattern specified by the configured `cptonelocale` command. |
| Example:               |                                                                          |
| `Router(config-voiceport)# ring cadence pattern01 define 12 15` |                                                                          |
| **Step 10**            |                                                                          |
| `description string`  | Attaches a text string to the configuration that describes the connection for this voice port. This description appears in various displays and is useful for tracking the purpose or use of the voice port. The `string` argument is a character string from 1 to 255 characters in length.  
  - The default is that no description is attached to the configuration. |
| Example:               |                                                                          |
| `Router(config-voiceport)# description 1` |                                                                          |
| **Step 11**            |                                                                          |
| `no shutdown`          | Activates the voice port.                                              |
| Example:               |                                                                          |
| `Router(config-voiceport)# no shutdown` |                                                                          |
CHAPTER 4

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.

• Finding Feature Information, page 55
• Information About Fine-Tuning Analog and Digital Voice Ports, page 55
• How to Configure Fine-Tuning Features for Voice Ports, page 56

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Information About Fine-Tuning Analog and Digital Voice Ports

• 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 Voice Ports**

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

<table>
<thead>
<tr>
<th>Note</th>
</tr>
</thead>
<tbody>
<tr>
<td>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 <em>Cisco IOS Voice Command Reference</em>.</td>
</tr>
</tbody>
</table>

**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 | fxs-ground-start | fxs-loop-start | fxo-ground-start | fxo-loop-start}`
5. `exit`
6. `voice-port slot / port`
7. `operation {2-wire | 4-wire}`
8. `type {1 | 2 | 3 | 5}`
9. Do one of the following:
   - `signal {loop-start | ground-start}`
   - `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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td><code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td></td>
</tr>
<tr>
<td>`controller {t1</td>
<td>e1} slot/port`</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>(T1 or E1) and a slot and port for configuration commands that</td>
</tr>
<tr>
<td>Router(config)# controller t1 1/0</td>
<td>specifically apply to the T1 or E1 interface.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td></td>
</tr>
<tr>
<td>`ds0-group ds0-group-number timeslots timeslot-list type {e&amp;m-delay-dial</td>
<td>Defines the T1 or E1 channels for use by compressed voice calls and</td>
</tr>
<tr>
<td>timeslots 1 type {e&amp;m-wink-start</td>
<td>the signaling method the router uses to connect to the PBX or central</td>
</tr>
<tr>
<td>fxo-ground-start</td>
<td>fxo-loop-start}</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-controller)# ds0-group 1</td>
<td>• The <code>ds0-group</code> command automatically creates a logical voice port.</td>
</tr>
<tr>
<td>timeslots 1 type e&amp;m-wink-start</td>
<td>• <code>ds0-group-number</code> --Value from 0 to 23 that identifies the DS0 group.</td>
</tr>
<tr>
<td></td>
<td>• <code>timeslot-list</code> --Single number, numbers separated by commas, or a</td>
</tr>
<tr>
<td></td>
<td>pair of numbers separated by a hyphen to indicate a range of</td>
</tr>
<tr>
<td></td>
<td>timeslots. For T1, allowable values are 1 to 24; for E1,</td>
</tr>
<tr>
<td></td>
<td>allowable values are 1 to 31.</td>
</tr>
<tr>
<td><strong>The signaling method selection for</strong></td>
<td>The signaling method selection for <code>type</code> depends on the connection</td>
</tr>
<tr>
<td><strong>type</strong></td>
<td>that you are making:</td>
</tr>
<tr>
<td></td>
<td>• Ear and Mouth (E&amp;M) connects PBX trunk lines (tie lines) and</td>
</tr>
<tr>
<td></td>
<td>telephone equipment. The wink and delay settings both specify</td>
</tr>
<tr>
<td></td>
<td>confirming signals between the sending and receiving ends, or the</td>
</tr>
<tr>
<td></td>
<td>immediate setting stipulates no special off-hook/on-hook signal.</td>
</tr>
<tr>
<td></td>
<td>• FXO connects a CO to a standard PBX interface where permitted by</td>
</tr>
<tr>
<td></td>
<td>local regulations.</td>
</tr>
<tr>
<td></td>
<td>• FXS connects basic telephone equipment and PBXs.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 5</strong> exit</td>
<td>Exits controller configuration mode and returns to global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# exit</td>
</tr>
<tr>
<td><strong>Step 6</strong> voice-port slot / port</td>
<td>Enters voice-port configuration mode and identifies a slot and port for configuration parameters.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice-port 2/1</td>
</tr>
<tr>
<td><strong>Step 7</strong> operation {2-wire</td>
<td>4-wire}</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# operation 4-wire</td>
</tr>
<tr>
<td><strong>Step 8</strong> type {1</td>
<td>2</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# type 2</td>
</tr>
<tr>
<td><strong>Step 9</strong> Do one of the following:</td>
<td>Defines the signal type to be used.</td>
</tr>
<tr>
<td>• signal {loop-start</td>
<td>ground-start}</td>
</tr>
<tr>
<td>• signal {wink-start</td>
<td>immediate</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# signal loop-start</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# signal wink-start</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td><strong>exit</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# <strong>exit</strong></td>
</tr>
<tr>
<td></td>
<td>Exits voice-port configuration mode and returns to global configuration mode.</td>
</tr>
<tr>
<td><strong>Step 11</strong></td>
<td>**connect connection-name voice-port voice-port-number {t1</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# <strong>connect connect1 voice-port 1/1/0 t1 1/0 0</strong></td>
</tr>
<tr>
<td></td>
<td>Creates a named connection between two voice ports associated with T1 or E1 interfaces where you have already defined the groups by using the <strong>ds0-group</strong> command.</td>
</tr>
<tr>
<td><strong>Step 12</strong></td>
<td><strong>exit</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# <strong>exit</strong></td>
</tr>
<tr>
<td></td>
<td>Exits the current configuration session and returns to privileged EXEC mode.</td>
</tr>
</tbody>
</table>

## 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>enable</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; <strong>enable</strong></td>
</tr>
</tbody>
</table>
|                  | Enables privileged EXEC mode.  
|                  | • Enter your password if prompted. |
Fine-Tuning Analog and Digital Voice Ports

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>terminal</td>
<td>Example: Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong> voice-port</td>
<td>Enters voice-port configuration mode.</td>
</tr>
<tr>
<td>slot/port</td>
<td>Example: Router(config)# voice-port 3/0</td>
</tr>
<tr>
<td><strong>Step 4</strong> condition</td>
<td>Manipulates sent or received bit patterns to match expected patterns on</td>
</tr>
<tr>
<td>{tx-a-bit</td>
<td>tx-b-bit</td>
</tr>
<tr>
<td>Example: Router(config-voiceport)# condition tx-a-bit on</td>
<td>• The default is that the signaling format is not manipulated (for all transmit or receive A, B, C, and D bits).</td>
</tr>
<tr>
<td><strong>Step 5</strong> define</td>
<td>Defines specific transmit or receive signaling bits to match the bit</td>
</tr>
<tr>
<td>{tx-bits</td>
<td>rx-bits} {seize</td>
</tr>
<tr>
<td>Example: Router(config-voiceport)# define tx-bits seize 0000</td>
<td>• Also specifies which bits a voice port monitors and which bits it ignores, if patterns that are different from the defaults are required.</td>
</tr>
<tr>
<td><strong>Step 6</strong> ignore</td>
<td>Configures the voice port to ignore the specified receive bit for North</td>
</tr>
<tr>
<td>{rx-a-bit</td>
<td>rx-b-bit</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step 2</th>
<th>configure terminal</th>
<th>Enters global configuration mode.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 3</th>
<th>voice-port slot / port</th>
<th>Enters voice-port configuration mode.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config)# voice-port 3/0</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 4</th>
<th>calling-number outbound range string1 string2</th>
<th>(Cisco AS5300 only) Specifies ANI to be sent out when the T1-CAS <strong>fgd-eana</strong> command is configured as signaling type. The <code>string1</code> and <code>string2</code> arguments are valid E.164 telephone number strings. Both strings must be of the same length and cannot be more than 32 digits long.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# calling-number outbound range 3000 4000</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 5</th>
<th>calling-number outbound sequence [string1] [string2] [string3] [string4] [string5]</th>
<th>(Cisco AS5300 only) Specifies ANI to be sent out when the T1-CAS <strong>fgd-eana</strong> command is configured as signaling type. This option configures a sequence of discrete strings (<code>string1...string5</code>) 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.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# calling-number outbound sequence 2000 3000 4000</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 6</th>
<th>calling-number outbound null</th>
<th>(Cisco AS5300 only) Suppresses ANI. No ANI is passed when this voice port is selected.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# calling-number outbound null</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 7</th>
<th>exit</th>
<th>Exits voice-port configuration mode and completes the configuration.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# exit</td>
<td></td>
</tr>
</tbody>
</table>
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, on page 67.

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td></td>
</tr>
<tr>
<td>configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 3</td>
<td></td>
</tr>
<tr>
<td>voice-port slot / port</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Note</td>
<td></td>
</tr>
<tr>
<td>The syntax of this command is platform-specific. For the syntax for your platform, refer to the <em>Cisco IOS Voice Command Reference</em>.</td>
<td></td>
</tr>
<tr>
<td>Step 4</td>
<td></td>
</tr>
<tr>
<td>no battery-reversal</td>
<td>(Analog only) Enables battery reversal. The default is that battery reversal is enabled.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Note</td>
<td></td>
</tr>
<tr>
<td>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 <strong>no battery-reversal</strong> command when a connected FXO port does not support battery reversal detection.</td>
<td></td>
</tr>
<tr>
<td>Step 5</td>
<td></td>
</tr>
<tr>
<td>no supervisory disconnect</td>
<td>(FXO only) Enables the PBX or PSTN switch to provide STD. The <strong>supervisory disconnect</strong> command is enabled by default.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Step 6</td>
<td></td>
</tr>
<tr>
<td>disconnect-ack</td>
<td>(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 <strong>no disconnect-ack</strong> command prevents the FXS port from responding to the on-hook disconnect with a removal of line power.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>
### 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.

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:

---

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 7</strong></td>
<td></td>
</tr>
<tr>
<td><code>exit</code></td>
<td>Exits voice-port configuration mode and completes the configuration.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router(config-voiceport)# exit
```
### 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 time
10. cadence-min-on-time time
11. cadence-max-off-time time
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 time
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
</tbody>
</table>

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

<p>| <strong>Step 3</strong> voice class dualtone tag| 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. |
| <strong>Example:</strong>                       |         |
| Router(config)# voice class dualtone 1 | • For more information about configuring voice classes, refer to &quot;Dial Peer Configuration on Voice Gateway Routers&quot;. |</p>
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 4</strong></td>
<td>freq-pair tone-id frequency-1 frequency-2</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-pair 16 300 0</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>Repeat this command for each additional tone to be specified.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>freq-max-deviation hertz</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-max-deviation 10</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>freq-max-power dBmO</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-max-power 20</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>freq-min-power dBmO</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-min-power 35</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td>freq-power-twist dBmO</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-power-twist 15</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td>freq-max-delay time</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# freq-max-delay 10</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td>cadence-min-on-time time</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voice-class)# cadence-min-on-time 10</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 11</strong></td>
<td>cadence-max-off-time time</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# cadence-max-off-time 2000</td>
</tr>
<tr>
<td>Specifies the maximum tone off time that will be detected, in 10-millisecond increments. Range is 0 to 5000 (0 ms to 50 seconds).</td>
<td></td>
</tr>
<tr>
<td><strong>Step 12</strong></td>
<td>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]</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# cadence-list 1 0 1000</td>
</tr>
<tr>
<td>(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:</td>
<td></td>
</tr>
<tr>
<td>• cadence-id -- Range is 1 to 10. There is no default.</td>
<td></td>
</tr>
<tr>
<td>• cycle-N-on-time -- Range is 0 to 1000 (0 ms to 10 seconds). Default is 0.</td>
<td></td>
</tr>
<tr>
<td>• cycle-N-off-time -- Range is 0 to 1000 (0 ms to 10 seconds). Default is 0.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 13</strong></td>
<td>cadence-variation time</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# cadence-variation 200</td>
</tr>
<tr>
<td>(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.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 14</strong></td>
<td>exit</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# exit</td>
</tr>
<tr>
<td>Exits voice class configuration mode.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 15</strong></td>
<td>voice-port slot / subunit / port</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# voice-port 0/1/0</td>
</tr>
<tr>
<td>Enters voice-port configuration mode.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 16</strong></td>
<td>supervisory disconnect dualtone {mid-call</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# supervisory disconnect dualtone mid-call voice-class 1</td>
</tr>
<tr>
<td>Assigns an FXO supervisory disconnect tone voice class to the voice port.</td>
<td></td>
</tr>
<tr>
<td><strong>Step 17</strong></td>
<td>supervisory disconnect anytone</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# supervisory disconnect anytone</td>
</tr>
<tr>
<td>Configures the voice port to disconnect on receipt of any tone.</td>
<td></td>
</tr>
</tbody>
</table>
### 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>enable</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>configure terminal</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Enters voice-port configuration mode.</td>
</tr>
<tr>
<td>voice-port slot / port</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice-port 3/0</td>
</tr>
<tr>
<td>Note</td>
<td>The syntax of this command is platform-specific. For the syntax for your platform, refer to the <em>Cisco IOS Voice Command Reference</em>.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-----------------------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td><strong>timeouts call-disconnect</strong> <em>seconds</em></td>
</tr>
<tr>
<td>Example:</td>
<td>Configures the call disconnect timeout value in seconds. Range is 0 to 120. Default is 60.</td>
</tr>
<tr>
<td>Router(config-voiceport)# timeouts call-disconnect 60</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>timeouts initial</strong> <em>seconds</em></td>
</tr>
<tr>
<td>Example:</td>
<td>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.</td>
</tr>
<tr>
<td>Router(config-voiceport)# timeouts initial 10</td>
<td>• The <em>seconds</em> argument is the initial timeout duration. Range is 0 to 120. Default is 10.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>timeouts interdigit</strong> <em>seconds</em></td>
</tr>
<tr>
<td>Example:</td>
<td>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).</td>
</tr>
<tr>
<td>Router(config-voiceport)# timeouts interdigit 10</td>
<td>• The <em>seconds</em> argument is the interdigit timeout wait time in seconds. Range is 0 to 120. Default is 10.</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td><strong>timeouts ringing</strong> { <em>seconds</em></td>
</tr>
<tr>
<td>Example:</td>
<td>Specifies the duration that the voice port allows ringing to continue if a call is not answered.</td>
</tr>
<tr>
<td>Router(config-voiceport)# timeouts ringing infinity</td>
<td>• Default for <em>seconds</em> is 180.</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td><strong>timeouts wait-release</strong> { <em>seconds</em></td>
</tr>
<tr>
<td>Example:</td>
<td>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.</td>
</tr>
<tr>
<td>Router(config-voiceport)# timeouts wait-release 30</td>
<td>• Default for <em>seconds</em> is 30.</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td><strong>exit</strong></td>
</tr>
<tr>
<td>Example:</td>
<td>Exits voice-port configuration mode and completes the configuration.</td>
</tr>
<tr>
<td>Router(config-voiceport)# exit</td>
<td></td>
</tr>
</tbody>
</table>

**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 milliseconds
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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td>Enters voice-port configuration mode.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice-port slot / port</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice-port 3/0</td>
<td></td>
</tr>
</tbody>
</table>

Note: The syntax of this command is platform-specific. For the syntax for your platform, refer to the Cisco IOS Voice Command Reference.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 4** timing clear-wait **milliseconds** | (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. |
| Example:                          |.Router(config-voiceport)# timing clear-wait 200                                                                                       |
| **Step 5** timing delay-duration **milliseconds** | (E&M only) Specifies the delay signal duration for delay-dial signaling, in milliseconds.  
  • Range is 100 to 5000. Default is 200. |
| Example:                          |.Router(config-voiceport)# timing delay-duration 100                                                                                   |
| **Step 6** 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. |
| Example:                          |.Router(config-voiceport)# timing delay-start milliseconds                                                                                  |
| **Step 7** timing delay-with-integrity **milliseconds** | (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. |
| Example:                          |.Router(config-voiceport)# timing delay-with-integrity 0                                                                                |
| **Step 8** timing dial-pulse min-delay **milliseconds** | 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. |
| Example:                          |.Router(config-voiceport)# timing dial-pulse min-delay 300                                                                             |
| **Step 9** timing dialout-delay **milliseconds** | (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. |
| Example:                          |.Router(config-voiceport)# timing dialout-delay 100                                                                                     |
| **Step 10** timing digit **milliseconds** | Specifies the DTMF digit signal duration in milliseconds.  
  • Range is 50 to 100. Default is 100. |
| Example:                          |.Router(config-voiceport)# timing digit 50                                                                                             |
| **Step 11** timing guard-out **milliseconds** | (FXO ports only) Specifies the duration in milliseconds of the guard-out period that prevents this port from seizing a remote FXS port before the remote port detects a disconnect signal.  
  • Range is 300 to 3000. Default is 2000. |
<p>| Example:                          |.Router(config-voiceport)# timing guard-out milliseconds                                                                                  |</p>
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 12</strong></td>
<td><strong>Purpose</strong></td>
</tr>
</tbody>
</table>
| timing hookflash-out milliseconds | Specifies the duration, in milliseconds, of the hookflash.  
• Range is 50 to 500. Default is 300. |
| Example: Router(config-voiceport)# timing hookflash-out 500 |  |
| **Step 13** | **Purpose** |
| timing interdigit milliseconds | Specifies the dual-tone multifrequency (DTMF) interdigit duration, in milliseconds.  
• Range is 50 to 500. Default is 100. |
| Example: Router(config-voiceport)# timing interdigit 100 |  |
| **Step 14** | **Purpose** |
| timing percentbreak percent | (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. |
| Example: Router(config-voiceport)# timing percentbreak 20 |  |
| **Step 15** | **Purpose** |
| timing pulse pulses-per-second | (FXO and E&M only) Specifies the pulse dialing rate in pulses per second.  
• Range is 10 to 20. Default is 20. |
| Example: Router(config-voiceport)# timing pulse 20 |  |
| **Step 16** | **Purpose** |
| timing pulse-digit milliseconds | (FXO only) Configures the pulse digit signal duration.  
• Range is 10 to 20. Default is 20. |
| Example: Router(config-voiceport)# timing pulse-digit 10 |  |
| **Step 17** | **Purpose** |
| timing pulse-interdigit milliseconds | (FXO and E&M only) Specifies pulse dialing interdigit timing in milliseconds.  
• Range is 100 to 1000. Default is 500. |
| Example: Router(config-voiceport)# timing pulse-interdigit 500 |  |
| **Step 18** | **Purpose** |
| timing wink-duration milliseconds | (E&M only) Specifies maximum wink-signal duration, in milliseconds, for a wink-start signal.  
• Range is 100 to 400. Default is 200. |
| Example: Router(config-voiceport)# timing wink-duration 200 |  |
### 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><code>Router&gt; enable</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td><code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td><code>Router# configure terminal</code></td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Configures a T1 controller and enters controller configuration mode.</td>
</tr>
<tr>
<td>controller T1 number</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# controller T1 1</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Configures channelized T1 time slots, which enables a Cisco AS5300 modem to answer and send an analog call.</td>
</tr>
<tr>
<td>ds0-group channel-number timeslots range type signaling-type dtmf dnis</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# ds0-group 0 timeslots 1-4 type e&amp;m-immediate-start dtmf dnis</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Enters cas-controller configuration mode and customizes signaling parameters for a particular E1 or T1 channel group on a channelized line.</td>
</tr>
<tr>
<td>cas-custom channel</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-controller)# cas-custom 2</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Configures the DTMF interdigit timer for a DS0 group.</td>
</tr>
<tr>
<td>dtmf timer-inter-digit milliseconds</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(conf-ctrl-cas)# dtmf timer-inter-digit 100</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td>Exits cas-controller configuration mode and completes the configuration.</td>
</tr>
<tr>
<td>exit</td>
<td></td>
</tr>
<tr>
<td>Example:</td>
<td>Router(conf-ctrl-cas)# exit</td>
</tr>
</tbody>
</table>

**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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> dial-peer voice tag voip</td>
<td>Enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# dial-peer voice 555 voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> vad [aggressive]</td>
<td>Enables VAD for calls using this dial peer.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-dial-peer)# vad</td>
<td><strong>Note</strong> VAD is enabled by default. Use the vad command only if you have previously disabled the feature by using the no vad command.</td>
</tr>
<tr>
<td>Step 5</td>
<td>Command or Action</td>
</tr>
<tr>
<td>--------</td>
<td>------------------</td>
</tr>
<tr>
<td></td>
<td>exit</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-dial-peer)# exit</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 6</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>voice vad-time milliseconds</td>
<td>Modifies the minimum silence detection time for VAD.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice vad-time 500</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 7</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>voice-port slot / port</td>
<td>Enters voice-port configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice-port 3/0</td>
<td>Note: The syntax of this command is platform-specific. For information, refer to the Cisco IOS Voice Command Reference.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 8</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>music-threshold number</td>
<td>Specifies the minimal decibel level of music played when calls are put on hold. The decibel level affects how VAD treats the music data.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# music-threshold -70</td>
<td>• 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.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 9</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>comfort-noise</td>
<td>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.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# comfort-noise</td>
<td>• Comfort noise is enabled by default.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 10</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>exit</td>
<td>Exits voice-port configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# exit</td>
<td></td>
</tr>
</tbody>
</table>
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.

- Finding Feature Information, page 81
- Information About PSTN Fallback, page 82
- Restrictions for PSTN Fallback, page 82
- How to Configure PSTN Fallback, page 83
- How to Verify and Monitor the PSTN Fallback Feature, page 97
- What To Do Next, page 98

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

The Cisco SAA functionality in Cisco IOS software was formerly known as Response Time Reporter (RTR). In the How to Configure PSTN Fallback, on page 83 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.

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

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. configure terminal
3. call fallback active
4. call fallback key-chain name-of-chain

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 call fallback active</td>
<td>Enables the PSTN fallback feature to alternate dial peers in case of network congestion.</td>
</tr>
<tr>
<td>Example: Router(config)# call fallback active</td>
<td></td>
</tr>
</tbody>
</table>
Purpose
Command or Action | Purpose
--- | ---
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. configure terminal
3. call fallback monitor

**DETAILED STEPS**

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

**Configuring Call Fallback Cache Parameters**

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

1. enable
2. configure terminal
3. call fallback cache-size \textit{number}
4. call fallback cache-timeout \textit{seconds}
5. clear call fallback cache [\textit{ip-address}]

### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> call fallback cache-size \textit{number}</td>
<td>Specifies the call fallback cache size.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# call fallback cache-size 5</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> call fallback cache-timeout \textit{seconds}</td>
<td>Specifies the time after which the cache entry is purged, in seconds. Default: 600.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# call fallback cache-timeout 300</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> clear call fallback cache [\textit{ip-address}]</td>
<td>Clears the current ICPIF estimates for all IP addresses or a specific IP address in the cache.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# clear call fallback cache 10.1.1.1</td>
<td></td>
</tr>
</tbody>
</table>

---

**Configuring Call Fallback Jitter-Probe Parameters**

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

1. **enable**
2. **configure terminal**
3. **call fallback jitter-probe num-packets**  \textit{number-of-packets}
4. **call fallback jitter-probe precedence**  \textit{precedence}
5. **call fallback jitter-probe priority-queue**

### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> call fallback jitter-probe num-packets \textit{number-of-packets}</td>
<td>Specifies the number of packets for jitter. Default: 15.</td>
</tr>
<tr>
<td>Example: Router(config)# call fallback jitter-probe num-packets 10</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> call fallback jitter-probe precedence \textit{precedence}</td>
<td>Specifies the treatment of the jitter-probe transmission. Default: 2.</td>
</tr>
<tr>
<td>Example: or</td>
<td>Specifies the differentiated services code point (dscp) packet of the jitter-probe transmission.</td>
</tr>
<tr>
<td>Example: Router(config)# call fallback jitter-probe dscp \textit{dscp-number}</td>
<td></td>
</tr>
</tbody>
</table>

**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.
### 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 *percent*

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
</tbody>
</table>

---

### Example:

**Step 5**

call fallback jitter-probe priority-queue

Assigns a priority to the queue for jitter probes.

**Example:**

Router(config)# call fallback jitter-probe dscp 2
### 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`

#### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **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`  
**Example:**  
`or` | Specifies fallback threshold to use packet delay and loss values. No defaults. |

---

### Configuring Call Fallback Threshold Parameters (continued)

#### Command or Action | Purpose
---|---
**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 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>* Enter your password if prompted.</td>
</tr>
</tbody>
</table>

**Example:**

```bash
Router(config)# call fallback threshold delay 100 loss 150
```

**Example:**

```bash
or
```

**Example:**

```bash
Router(config)# call fallback threshold icpif 100
```

### Notes

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 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router# configure terminal
```

**Step 2**

Configures the waiting timeout interval for a response to a probe in milliseconds. Default: 300 milliseconds.

**Example:**

```
Router(config)# call fallback
wait-timeout 200
```

**Step 3**

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.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong>&lt;br&gt;Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
</tbody>
</table>

**Step 2**

configure terminal<br><br>**Example:**<br>Router# configure terminal

Enters global configuration mode.

**Step 3**

call fallback active<br><br>**Example:**<br>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<br><br>**Example:**<br>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. Do one of the following:
   • call fallback map map target ip-address address-list ip-address1 ip-address2 ... ip-address7
   • call fallback map map target ip-address subnet ip-network netmask

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> Do one of the following:</td>
<td>Specifies the call fallback router to keep a cache table (by IP addresses) of distances for several destination peers sitting behind the router.</td>
</tr>
<tr>
<td>• call fallback map map target ip-address address-list ip-address1 ip-address2 ... ip-address7</td>
<td></td>
</tr>
<tr>
<td>• call fallback map map target ip-address subnet ip-network netmask</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Specifies the call fallback router to keep a cache table (by subnet addresses) of distances for several destination peers sitting behind the router.</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

**Dial Peer Configuration**

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} [ip address]`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
| **Example:** Router> enable | • Enter your password if prompted. |
| **Step 2** configure terminal | Enters global configuration mode. |
| **Example:** Router# configure terminal | |
| **Step 3** dial-peer voice tag voip | Enters dial peer configuration mode, specifies the method of voice encapsulation, and defines a particular dial peer:  
| **Example:** Router(config)# dial-peer voice 10 voip | `tag` --Digits that define a particular dial peer. Range is from 1 to 2147483647. |
| **Step 4** call fallback [icmp-ping] rtr | Configures dial-peer parameters for pings to IP destinations:  
| **Example:** Router(config-dial-peer)# call fallback icmp-ping | • `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. |
Purpose

Command or Action | Purpose
--- | ---

**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**

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 bytes] interval seconds [loss number] [timeout milliseconds]*

### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>
| Example:
 Router> enable | |
| **Step 2** configure terminal | Enters global configuration mode. |
| Example:
 Router# configure terminal | |
### PSTN Fallback

#### Configuring ICMP Pings to Monitor IP Destinations

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| 3    | `call fallback active [icmp-ping] rtr` | 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]` | 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

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

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>
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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>
# How to Verify and Monitor the PSTN Fallback Feature

## 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.
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, on page 92 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.
CHAPTER 6

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:

• Finding Feature Information, page 99
• Information About Echo Cancellation, page 99
• How to Configure the Extended G.168 Echo Canceller, page 109
• Restrictions for G.168 Extended Echo Canceller, page 109
• Configuration Examples for Extended G.168 Echo Cancellation, page 118

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Information About Echo Cancellation

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.

The figure below 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 13: Transmit and Receive Paths in a Voice Network](image)

**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.

The figure below shows a common voice network where echo cancellation might be used.

![Figure 14: Echo Cancellation Network](image)

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.

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.

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.

The figure below shows a typical DSP channel configured for voice processing.

Figure 15: 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.

• A combined (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 5: Feature History for Extended ITU-T G.168 Echo Cancellation**

<table>
<thead>
<tr>
<th>Release</th>
<th>Modification</th>
</tr>
</thead>
<tbody>
<tr>
<td>12.2(13)T</td>
<td>This feature was introduced.</td>
</tr>
<tr>
<td>12.2(13)ZH</td>
<td>The extended G.168 EC became the default on the Cisco 1700 series and the Cisco ICS 7750.</td>
</tr>
<tr>
<td>12.2(15)ZJ</td>
<td>The extended G.168 EC became the default on the Cisco 2600 series, Cisco 3600 series, Cisco 3700 series.</td>
</tr>
<tr>
<td></td>
<td><strong>Note</strong> 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.</td>
</tr>
<tr>
<td>12.3(1)</td>
<td>The extended G.168 EC became the default on the Cisco IAD2420, Cisco MC3810, and Cisco VG200.</td>
</tr>
<tr>
<td>12.3(4)T</td>
<td>The extended G.168 EC became the default on the Cisco 7200 series and Cisco Catalyst 4000 AGM.</td>
</tr>
</tbody>
</table>
## 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

The table below contains comparison information for G.165 and G.168 echo cancellation.

Table 6: Echo Canceller Comparison

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

Extended Echo Canceller Support by Platform

The table below 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 7: Extended Echo Canceller Algorithm Coverage by Platform

<table>
<thead>
<tr>
<th>Platform</th>
<th>Network Module</th>
<th>High Complexity Codec</th>
<th>Medium Complexity Codec</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Digital</td>
<td>Analog</td>
<td>Digital</td>
</tr>
<tr>
<td>Cisco 1700 series</td>
<td>--</td>
<td>12.2(8)YN</td>
<td>12.2(8)YN</td>
<td>12.2(8)YN</td>
</tr>
<tr>
<td></td>
<td></td>
<td>12.2(13)T</td>
<td>12.2(13)T</td>
<td>12.2(13)T</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>12.2(13)T</td>
<td>12.2(13)T</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Flexi6 support in Cisco IOS Release 12.2(8)YN.</td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>NM-HDV</td>
<td>--</td>
<td>12.2(13)T</td>
<td>12.2(13)T</td>
</tr>
<tr>
<td>Cisco 2600XM</td>
<td>(C549)</td>
<td></td>
<td></td>
<td>Full support.</td>
</tr>
<tr>
<td>Cisco 3600 series</td>
<td>Cisco 3700 series</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td>Cisco VG200</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>NM-1V, NM-2V</td>
<td>--</td>
<td>--</td>
<td>12.2(13)T</td>
</tr>
<tr>
<td>Cisco 2691 Cisco</td>
<td>(C542)</td>
<td></td>
<td>--</td>
<td></td>
</tr>
<tr>
<td>Cisco 3600 series</td>
<td>Cisco 3700 series</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td>Cisco VG200</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Not supported.</td>
</tr>
<tr>
<td>Platform</td>
<td>Network Module</td>
<td>High Complexity Codec</td>
<td>Medium Complexity Codec</td>
<td>Comments</td>
</tr>
<tr>
<td>----------</td>
<td>----------------</td>
<td>------------------------</td>
<td>-------------------------</td>
<td>----------</td>
</tr>
<tr>
<td>Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series</td>
<td>NM-HDxx</td>
<td>12.3(4)XD</td>
<td>12.3(4)XD</td>
<td>12.3(4)XD</td>
</tr>
<tr>
<td>Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series</td>
<td>AIM-Voice (C5421), AIM-Voice-30 (C542)</td>
<td>--</td>
<td>12.2(15)ZJ</td>
<td>--</td>
</tr>
<tr>
<td>Cisco 2600XM Cisco 2691 Cisco 3640 Cisco 3660 Cisco 3700 series</td>
<td>NM-HDA (C5421)</td>
<td>12.2(15)ZJ 12.3(4)T</td>
<td>--</td>
<td>12.2(15)ZJ 12.3(4)T</td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>NM-HDA (C5421)</td>
<td>12.3(9)</td>
<td>--</td>
<td>12.3(9)</td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>AIM-Voice (C5421)</td>
<td>--</td>
<td>12.3(9)</td>
<td>--</td>
</tr>
<tr>
<td>Cisco 7200 series</td>
<td>PA-VXx-2TE1+ and PA-MCX-nTE1</td>
<td>--</td>
<td>12.2(13)T</td>
<td>--</td>
</tr>
<tr>
<td>Cisco 7500 series</td>
<td>--</td>
<td>--</td>
<td>12.2(13)T</td>
<td>--</td>
</tr>
<tr>
<td>Cisco 7600 series</td>
<td>Communication Media Module (WS-SVC-CMM) with one of the following port adapters: WSSVCCMM6T1 WSSVCCMM4E1 WSSVCCMM24FXS</td>
<td>--</td>
<td>12.3(8)XY 12.3(14)T</td>
<td>--</td>
</tr>
<tr>
<td>Platform</td>
<td>Network Module</td>
<td>High Complexity Codec</td>
<td>Medium Complexity Codec</td>
<td>Comments</td>
</tr>
<tr>
<td>------------------</td>
<td>------------------</td>
<td>-----------------------</td>
<td>-------------------------</td>
<td>--------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Cisco AS5300</td>
<td>--</td>
<td>--</td>
<td>12.2(13)T (restricted)</td>
<td>12.3(3) (unrestricted)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>1-channel DSP on C549 with extended EC, any codec (unrestricted).</td>
</tr>
<tr>
<td>Cisco AS5350</td>
<td>NextPort DFC</td>
<td>--</td>
<td>Digital - 12.3(11)T</td>
<td>12.3(11)T</td>
</tr>
<tr>
<td>Cisco AS5400</td>
<td>modules: DFC60</td>
<td></td>
<td></td>
<td>See the &quot;NextPort-Based Voice Tuning and Echo Cancellation&quot; chapter in this guide.</td>
</tr>
<tr>
<td>Cisco AS5850</td>
<td>DFC108 1 CT3_UPC</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco Catalyst 4000</td>
<td>AGM</td>
<td>12.3(4)T</td>
<td>--</td>
<td>12.3(4)T High-complexity analog and medium-complexity digital is planned.</td>
</tr>
<tr>
<td>Cisco Catalyst 6000</td>
<td>Cisco 6624</td>
<td>A002040- 00002</td>
<td>--</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Cisco 6608</td>
<td>--</td>
<td>A004040- 00002</td>
<td></td>
</tr>
<tr>
<td>Cisco Catalyst 6500 series</td>
<td>Communication Media Module (WS-SVC-CMM) with one of the following port adapters: WSSVCCMM6F1 WSSVCCMM6E1</td>
<td>12.3(8)XY 12.3(14)T</td>
<td>--</td>
<td></td>
</tr>
<tr>
<td></td>
<td>WSSVCCMM6F1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>WSSVCCMM6E1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco IAD2420</td>
<td>--</td>
<td>12.2(13)T</td>
<td>12.3(1) mainline</td>
<td>12.3(1) mainline</td>
</tr>
<tr>
<td>Cisco IAD243x</td>
<td>VIC2-4FXO onboard T1</td>
<td>12.3(4)XD</td>
<td>12.3(4)XD</td>
<td>12.3(4)XD</td>
</tr>
<tr>
<td>Cisco MC3810</td>
<td>HCM 549</td>
<td>12.2(13)T</td>
<td>12.3(1) mainline</td>
<td>12.3(1) mainline</td>
</tr>
</tbody>
</table>
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.

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.

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

Table 8: Cisco IOS Commands for Selecting Extended E.168 EC by Platform and Cisco IOS Release

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco IOS Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco 1700 Series and Cisco ICS 7750</td>
<td></td>
</tr>
<tr>
<td>12.2(13)T</td>
<td>Router(config)# voice echo-canceller extended</td>
</tr>
<tr>
<td>12.2(13)ZH 12.2(15)ZJ 12.3(1)</td>
<td>Router(voice-card)# codec complexity medium</td>
</tr>
<tr>
<td>12.3(4)T and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
<tr>
<td>Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series, Cisco MC3810, Cisco VG200</td>
<td></td>
</tr>
<tr>
<td>12.2(13)T 12.2(13)ZH 12.3(1)</td>
<td>Router(voice-card)# codec complexity medium or codec complexity high ecan-extended</td>
</tr>
<tr>
<td></td>
<td>Router(voice-card)# codec complexity medium</td>
</tr>
<tr>
<td>12.2(15)ZJ 12.3(4)T</td>
<td>Router(voice-card)# codec complexity medium</td>
</tr>
<tr>
<td>12.3(4)XD and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
</tbody>
</table>
### How to Configure the Extended G.168 Echo Canceller

#### 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.

<table>
<thead>
<tr>
<th>Cisco IOS Release</th>
<th>Cisco IOS Command</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco 7200 Series and Cisco 7500 Series</td>
<td></td>
</tr>
<tr>
<td>12.2(13)T</td>
<td><code>Router(config-dspfarm)# codec complexity medium ecan-extended</code></td>
</tr>
<tr>
<td>12.2(13)ZH and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
<tr>
<td>Cisco 7600 Series with Communication Media Modules</td>
<td></td>
</tr>
<tr>
<td>12.3(8)XY and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
<tr>
<td>Cisco AS5300</td>
<td></td>
</tr>
<tr>
<td>12.2(13)T</td>
<td><code>Router(config)# voice echo-canceller extended codec small codec large codec</code></td>
</tr>
<tr>
<td>12.3(3)</td>
<td><code>Router(config)# voice echo-canceller extended [codec small codec large codec]</code></td>
</tr>
<tr>
<td>Cisco Catalyst 6500 Series with Communication Media Modules</td>
<td></td>
</tr>
<tr>
<td>12.3(8)XY and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
<tr>
<td>Cisco Catalyst 4000 AGM</td>
<td></td>
</tr>
<tr>
<td>12.3(4)T and later</td>
<td>No configuration necessary. G.168 EC enabled by default.</td>
</tr>
</tbody>
</table>
Changing Echo Cancellers on Digital Voice Ports

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

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


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 the table above for extended EC algorithm coverage by platform.

Enabling the Extended G.168 EC 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>2. configure terminal</td>
<td></td>
</tr>
<tr>
<td>3. voice echo-canceller</td>
<td></td>
</tr>
<tr>
<td>4. exit</td>
<td></td>
</tr>
</tbody>
</table>

**Example:**

```
Router> enable
```

- Enter your password if prompted.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice echo-canceller extended</td>
<td>Enables the G.168 extended echo canceller on the Cisco 1700 series or</td>
</tr>
<tr>
<td>Example: Router(config)# voice echo-canceller extended</td>
<td>Cisco ICS 7750.</td>
</tr>
<tr>
<td></td>
<td>• You do not have to shut down all the voice ports on the Cisco</td>
</tr>
<tr>
<td></td>
<td>1700 or Cisco ICS 7750 to switch the echo canceller, but you</td>
</tr>
<tr>
<td></td>
<td>should make sure that when you switch the echo canceller, there</td>
</tr>
<tr>
<td></td>
<td>are no active calls on the router.</td>
</tr>
<tr>
<td></td>
<td>• To return to the proprietary Cisco G.165 default EC, use the no</td>
</tr>
<tr>
<td></td>
<td>form of the command.</td>
</tr>
<tr>
<td>Step 4 exit</td>
<td>Exits global configuration mode.</td>
</tr>
<tr>
<td>Example: Router(config)# exit</td>
<td></td>
</tr>
</tbody>
</table>

**Enabling the Extended G.168 EC 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 medium
5. end
## DETAILED STEPS

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

### 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.
**Before You Begin**

Extended EC with Restricted Codecs on C542 or C549 DSP:

- Create a backup of your router configuration.
- Determine which codecs are required when enabling the extended echo canceller. For this information, see the codec restriction options for the `voice echo-canceler extended` command.
  - 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.
  - 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.
- 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.
- 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:

- Create a backup of your router configuration.
- 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.

---

**Note**

- 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. Do one of the following:
   • voice echo-canceller extended
   • voice echo-canceller extended [codec small codec large codec]
8. exit

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> no dial-peer voice tag voip</td>
<td>(Optional) Removes VoIP dial peers, one dial peer at a time.</td>
</tr>
<tr>
<td>Example: Router(config)# no dial-peer voice 1 voip</td>
<td>• When configuring the extended EC in global configuration mode, you must remove or modify all existing VoIP dial peers before the <strong>voice echo-canceller extended</strong> command is accepted.</td>
</tr>
<tr>
<td><strong>Step 4</strong> dial-peer voice tag voip</td>
<td>Enters dial-peer configuration mode so that you can modify a codec type.</td>
</tr>
<tr>
<td>Example: Router(config)# dial-peer voice 1 voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> codec {g711alaw</td>
<td>g711ulaw</td>
</tr>
</tbody>
</table>
### 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables higher privilege levels, such as privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong> voice-port slot / port:ds0-group-number</td>
<td>Enters voice-port configuration mode on the selected slot, port, and DS0 group.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice-port 1/0:0</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>The syntax of this command is platform-specific. For the syntax for your platform, refer to the Cisco IOS Voice Command Reference.</td>
</tr>
<tr>
<td><strong>Step 4</strong> echo-cancel enable</td>
<td>Enables echo cancellation.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voiceport)# echo-cancel enable</td>
</tr>
<tr>
<td></td>
<td>• Echo cancellation is enabled by default. Use the no form of this command to disable echo cancellation.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• On the Cisco AS5300, the Cisco G.165 EC is enabled by default with echo suppression disabled.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>This command is supported only when the echo-cancel coverage command is enabled.</td>
</tr>
<tr>
<td>Step</td>
<td>Command or Action</td>
</tr>
<tr>
<td>-------</td>
<td>-------------------</td>
</tr>
</tbody>
</table>
| 5     | `echo-cancel coverage` {24 | 32 | 48 | 64 | 80 | 96 | 112 | 128} | 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. |
| 6     | `echo-cancel erl worst-case` {0 | 3 | 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. |
| 7     | `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. |
| 8     | `echo-cancel suppressor seconds` | (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)Z1 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 purpose of the echo-cancel suppressor command is visible when the G.168 extended EC is selected but it has no effect. This command is supported only when echo cancellation is enabled.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• End</td>
<td>Exits voice-port configuration mode and completes the configuration.</td>
</tr>
</tbody>
</table>

### Configuration Examples for Extended G.168 Echo Cancellation

**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 EC 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. This example applies to the Cisco 2600 Series, Cisco 3600 Series, Cisco 3700 Series routers, and Cisco VG200.

```
voice-card 1
  codec complexity high ecan-extended
```

**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
```
Enabling the Extended EC on the Cisco 7200 and Cisco 7500 Series Example

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

```
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 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
    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
!
isdnst switch-type primary-5ess
!
voice echo-canceller extended codec small g711 large fax-relay
!
!
fax interface-type fax-mail
```

voice-port configuration guide, Cisco IOS Release 15M&T
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
```
Pulse Code Modulation (PCM) Audio Capture

The Pulse Code Modulation (PCM) Audio capture feature is used for debugging audio quality issues. PCM capture refers to an existing Digital Signal Processor (DSP) feature by which the digital audio signal at various nodes in the audio signal processing path of a voice channel may be intercepted and uploaded to the host router using specialized DSP-to-host message packets. Cisco IOS file services allow a file containing interleaved audio and debug data (.dat) to be created in the local file system or a remote TFTP server. This .dat file is then decoded and deinterleaved into separate, synchronized .wav files for each of the signal interception nodes. This feature is typically employed for capture of audio test signals in troubleshooting specific voice issues such as echo. Signals may be captured at any or all of the defined nodes, including the input-output nodes of an echo canceller (Rin, Sin, Sout), the Acoustic Shock Protection circuit, and the Noise Reduction module. Additional nodes of interest will be added as new signal processing features are introduced.

- Finding Feature Information, page 121
- Information about PCM Audio Capture, page 122
- How to Configure PCM Audio Capture, page 122
- Additional References for Cisco UBE Serviceability, page 125
- Feature Information for Pulse Code Modulation (PCM) Audio Capture, page 126

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.
Information about PCM Audio Capture

PCM Audio Capture

The following are the enhancements to the PCM Audio Capture feature:

• Separate PCM capture and Banjo logger feature so that they do not share the same data (.dat) file; they have their own data file.

• One PCM call per data file is generated dynamically. The filename contains information such as voice port type and number, call ID, calling and called number, GUID, DSP channel number, and time stamp.

• A user on the TDM-TDM or TDM-VoIP call can dynamically enable and disable PCM capture by entering predefined start and stop Dual Tone Multi-Frequency (DTMF) digits.

• More test points or streams can be captured.

Note

PCM capture is a CPU-intensive feature, and you must not enable several PCM capture sessions while running heavy traffic.

How to Configure PCM Audio Capture

Configuring PCM Audio Capture

SUMMARY STEPS

1. enable
2. configure terminal
3. voice pcm capture buffer number
4. voice pcm capture destination url
5. voice pcm capture on-demand-trigger
6. voice pcm capture user-trigger-string start-string stop-string stream bitmap duration call-duration
7. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
</tbody>
</table>
### Purpose

**Command or Action**

**Example:**

```
Router> enable
```

- Enter your password if prompted.

**Step 2**

**configure terminal**

**Example:**

```
Router# configure terminal
```

Enters global configuration mode.

**Step 3**

**voice pcm capture buffer** *number*

**Example:**

```
Router(config)# voice pcm capture buffer 10
```

Configures the number of PCM capture buffers. The Range is from 0 to 200000. To change the PCM capture buffer size, you must first configure it with 0 and then configure it with the desired number.

**Step 4**

**voice pcm capture destination** *url*

**Example:**

```
Router(config)# voice pcm capture destination tftp://10.10.1.2/acphan/
```

Configures or changes the destination URL for storing captured data.

**Step 5**

**voice pcm capture on-demand-trigger**

**Example:**

```
Router(config)# voice pcm capture on-demand-trigger
```

Configures user-triggered PCM capture.

**Step 6**

**voice pcm capture user-trigger-string**

**Example:**

```
Router(config)# voice pcm capture #132 #543 stream ff duration 230
```

Changes the default user trigger PCM capture start and stop string, stream, and duration.

- The start and stop string must have different values.
- PCM stream bitmap is in hexadecimal. The range is from 1 to FFFFFFFF.
  - The stream bitmap definitions are as follows:
    - bit 0—Rin
    - bit 1—Sin
    - bit 2—Sout
    - bit 3—nonNLP Sout
    - bit 4—fax modem in
    - bit 5—fax modem out
### Verifying PCM Audio Capture

Perform this task to verify the configuration for the PCM Audio Capture feature.

**SUMMARY STEPS**

1. `enable`
2. `show voice pcm capture`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td><code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td><code>show voice pcm capture</code></td>
<td>PCM Capture is on and is logging to URL tftp://10.10.1.2/acphan/ 50198 messages sent to URL, 0 messages dropped Message Buffer (total:inuse:free) 200000:0:200000</td>
</tr>
</tbody>
</table>
Buffer Memory: 68000000 bytes, Message size: 340 bytes

Displays the configured PCM capture buffer and destination, number of saved messages/packets, number of dropped messages/packets, and number of buffers allocated, both used and free.

## Additional References for Cisco UBE Serviceability

### Related Documents

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco IOS commands</td>
<td>Cisco IOS Master Command List, All Releases</td>
</tr>
<tr>
<td>Voice commands</td>
<td>• Cisco IOS Voice Command Reference - A through C</td>
</tr>
<tr>
<td></td>
<td>• Cisco IOS Voice Command Reference - D through I</td>
</tr>
<tr>
<td></td>
<td>• Cisco IOS Voice Command Reference - K through R</td>
</tr>
<tr>
<td></td>
<td>• Cisco IOS Voice Command Reference - S Commands</td>
</tr>
<tr>
<td></td>
<td>• Cisco IOS Voice Command Reference - T through Z Commands</td>
</tr>
</tbody>
</table>

### Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support and Documentation website provides online resources to</td>
<td><a href="http://www.cisco.com/cisco/web/support/index.html">http://www.cisco.com/cisco/web/support/index.html</a></td>
</tr>
<tr>
<td>download documentation, software, and tools. Use these resources to install</td>
<td></td>
</tr>
<tr>
<td>configure the software and to troubleshoot and resolve technical issues</td>
<td></td>
</tr>
<tr>
<td>with Cisco products and technologies. Access to most tools on the Cisco</td>
<td></td>
</tr>
<tr>
<td>Support and Documentation website requires a Cisco.com user ID and password.</td>
<td></td>
</tr>
</tbody>
</table>
Feature Information for Pulse Code Modulation (PCM) Audio Capture

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

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

**Table 9: Feature Information for Pulse Code Modulation (PCM) Audio Capture**

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pulse Code Modulation (PCM) Audio Capture</td>
<td>15.2(2)T</td>
<td>The PCM Capture feature is used for debugging audio quality issues.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>In Cisco IOS Release 15.2(2)T, this feature was implemented on the Cisco Unified Border Element.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>The following commands were introduced or modified: show voice pcm capture, voice pcm capture.</td>
</tr>
<tr>
<td>Pulse Code Modulation (PCM) Audio Capture</td>
<td>Cisco IOS XE Release 3.6S</td>
<td>The PCM Capture feature is used for debugging audio quality issues.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>In Cisco IOS XE Release 3.6S, this feature was implemented on the Cisco Unified Border Element (Enterprise)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>The following commands were introduced or modified: show voice pcm capture, voice pcm capture.</td>
</tr>
</tbody>
</table>
Acoustic Shock Protection

Acoustic Shock Protection (ASP) is a voice circuit-breaker feature that is designed to protect users, especially those wearing headsets, from exposure to loud, sustained, and piercing tones, such as those produced by a fax machine. It is a workplace-safety feature for voice calls. When the tone is present at the input of the ASP module, the audio path in the affected direction is muted to protect the listener, and a gentle alert tone is played out for as long as the tone persists. ASP may be inserted in either or both directions of a call, that is, applied to incoming packets to protect the ears of a listener on the Time-Division Multiplexing (TDM) gateway, applied to incoming PSTN calls (microphone signal) to protect the ears of listeners at the other end of the call, or applied to both simultaneously.

- Finding Feature Information, page 127
- Restrictions for ASP, page 127
- Information About ASP, page 128
- How to Configure ASP, page 129
- Configuration Examples for the Acoustic Shock Protection Feature, page 134
- Feature Information for Acoustic Shock Protection, page 135

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Restrictions for ASP

- Supported on PVDM3 only.
- Supported only on flex codec complexity.
• No support for H.32x video call, complex forking calls, and fax and modem calls.
• No support for TDM hairpin call.
• The configuration under dial peer has higher priority than the configuration at the global level.
• No support for conference calls, IP/SIP phones, and the Skinny Client Control Protocol (SCCP).
• CLI supports enabling ASP but not disabling ASP.
• No support for dynamically enabling or disabling ASP during a call.

Information About ASP

Acoustic Shock Protection

Acoustic Shock Protection (ASP) is an adaptive signal processing algorithm on the Digital Signal Processor (DSP) that analyzes incoming audio for the presence of offending tones that might harm humans. Offending tones include signals that are:

• Loud
• Tonal (energy concentrated around a single frequency)
• Persistent (lasts longer than a few tens of milliseconds)

If an offending tone is present, the audio path in that direction is muted temporarily, and a quiet, alerting signal is played out to the listener side. The call is never dropped; only the audio is muted temporarily. If or when the tone disappears from the input, the mute is removed. ASP does not disrupt low-frequency tones (below 650 Hz) such as ringback, dial, and so forth. Since ASP is designed to mute only single-frequency tones, it allows multi-tone signals such as Dual Tone Multi-Frequency (DTMF) to pass unhindered. ASP is supported on TDM gateways (TDM-VoIP and TDM-TDM) and on the Cisco Unified Border Element (Cisco UBE).

Note

ASP is for voice calls only and not for faxes and modems.

Some of the best practices for ASP are as follows:

• Use default values
• Use ASP on dial peers where you are certain that people (not faxes) are listening.
• Do not use ASP on dial peers associated with fax machines, modems, or TTY/TDD devices. Use fax-relay or modem-relay modes on dial peers dedicated to such devices.
• ASP is designed for deployment in situations where customers have experienced acoustic shock safety issues. If there are issues like false triggering (for example, ASP alerts on regular voices), then you must turn off ASP. You can choose from three detector sensitivity modes: slow, auto, or fast. Fast mode is a highly sensitive hair-trigger. Auto mode is recommended. Slow mode lets more tone leak through, but has better rejection of false triggers.
How to Configure ASP

Creating the Media Profile for ASP

Perform this task to create a media profile to configure acoustic shock protection.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media profile asp** \textit{tag} \\
4. **mode** \textit{mode} \\
5. **end**

DETAILED STEPS

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>enable</strong></td>
<td>Enables privileged EXEC mode.</td>
<td></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Device&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>configure terminal</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>media profile asp</strong> \textit{tag}</td>
<td>Creates the media profile to configure ASP and enters media profile configuration mode. The range for the media profile tag is from 1 to 10000.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Device(config)# media profile asp 5</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td><strong>mode</strong> \textit{mode}</td>
<td>Sets the ASP sensitivity mode to \textit{preset = auto} (which is default). Auto mode provides a good tradeoff between ASP speed and false trigger rejection. The other modes are:</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Device(cfg-mediaprofile)# mode auto</td>
<td></td>
</tr>
</tbody>
</table>

- **slow**—Presets ASP sensitivity mode to 1. This mode provides slower detection speed for reduced chance of false triggers.
- **fast**—Presets ASP sensitivity mode to 2. This mode provides faster detection speed but higher chance of false triggers.
- **expert**—This mode exposes direct control of individual ASP parameters and is recommended for test use only.
Creating the Media Profile to Enable ASP

After the media profile is created, you must create a media class to enable acoustic shock protection. Perform this task to create a media class.

SUMMARY STEPS

1. enable
2. configure terminal
3. media class\ tag
4. asp profile\ tag
5. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>enable</strong></td>
</tr>
<tr>
<td></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td>device&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>configure terminal</strong></td>
</tr>
<tr>
<td></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>device# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>media class\ tag</strong></td>
</tr>
<tr>
<td></td>
<td>Creates the media class to enable the acoustic shock protection feature</td>
</tr>
<tr>
<td></td>
<td>and enters media class configuration mode. The range for the media</td>
</tr>
<tr>
<td></td>
<td>class tag is from 1 to 10000.</td>
</tr>
<tr>
<td>Example:</td>
<td>device(config)# media class 2</td>
</tr>
</tbody>
</table>
### Configuring the Media Class at a Dial Peer Level for ASP

#### SUMMARY STEPS

1. enable
2. configure terminal
3. dial-peer voice tag pots
4. media-class tag
5. end

#### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>enable</strong></td>
</tr>
<tr>
<td>Example:</td>
<td><strong>Device&gt; enable</strong></td>
</tr>
<tr>
<td></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>configure terminal</strong></td>
</tr>
<tr>
<td>Example:</td>
<td><strong>Device# configure terminal</strong></td>
</tr>
<tr>
<td></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>dial-peer voice tag pots</strong></td>
</tr>
<tr>
<td>Example:</td>
<td><strong>Device(config)# dial-peer voice 20 pots</strong></td>
</tr>
<tr>
<td></td>
<td>Defines a particular dial peer and enters dial-peer voice configuration mode. The range for the dial-peer voice tag is from 1 to 1073741823.</td>
</tr>
</tbody>
</table>
### Configuring the Media Class Globally for ASP

**SUMMARY STEPS**

1. enable
2. configure terminal
3. media service
4. enhancement
5. tdm  tag
6. end

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td></td>
</tr>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Device&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td></td>
</tr>
<tr>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td></td>
</tr>
<tr>
<td>media service</td>
<td>Enters media service configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(config)# media service</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td></td>
</tr>
<tr>
<td>media-class  tag</td>
<td>Applies the media class to the specific dial peer. The range for the media class tag number is from 1 to 10000.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(config-dial-peer)# media-class 2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td></td>
</tr>
<tr>
<td>end</td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Device(config-dial-peer)# end</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td>Step 4 enhancement</td>
<td>Enters the submode enhance of media service.</td>
</tr>
<tr>
<td>Example: Device(cfg-mediaservice)# enhancement</td>
<td></td>
</tr>
<tr>
<td>Step 5 tdm tag</td>
<td>Applies the TDM call globally. The range for the media class tag number is from 1 to 10000.</td>
</tr>
<tr>
<td>Example: Device(cfg-service-enhance)# tdm 2</td>
<td></td>
</tr>
<tr>
<td>Step 6 end</td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Device(config-dial-peer)# end</td>
<td></td>
</tr>
</tbody>
</table>

### Verifying ASP

Perform this task to verify the voice quality metrics.

**SUMMARY STEPS**

1. enable
2. show call active voice stats | b pid:

**DETAILED STEPS**

**Step 1 enable**

**Example:** Device> enable

Enables privileged EXEC mode.

**Step 2 show call active voice stats | b pid:**

**Example:**

Device# show call active voice stats | b pid:1300

11EC : 5 09:14:25.971 PDT Thu Jul 28 2011.1 +1130 pid:1300 Answer 1300 active dur 00:01:36 tx:17/321 rx:17/321 dscp:0 media:0 DSP/TX: PK=17, SG=0, NS=1, DU=90570, VO=320 DSP/RX: PK=17, SG=0, CF=1, RX=90570, VO=320, BS=0, BP=0, LP=0, EP=0

...
DSP/DL: RT=0, ED=0
MIC Direction:
DSP/NR: NR=1, ND=0, LV=257, IN=1, PN=0, ON=0
DSP/AS: AE=1, AD=0, AV=0, AM=0, NT=0, DT=0, TT=0, TD=0, LF=0, LD=0
EAR Direction:
DSP/NR: NR=0, ND=0, LV=0, IN=0, PN=0, ON=0
DSP/AS: AE=0, AD=0, AV=0, AM=0, NT=0, DT=0, TT=0, TD=0, LF=0, LD=0

Displays information about digital signal processing (DSP) voice quality metrics.

Troubleshooting Tips

The following commands can help troubleshoot ASP:

- debug voip hpi all
- debug voip dsmp all
- debug voip dsm all
- debug voip vtsp all
- debug vpm dsp all

Configuration Examples for the Acoustic Shock Protection Feature

Example: Enabling ASP Globally

```bash
media profile asp 6
!
media class 1
  asp profile 6
!
media service enhancement
tdm 1
```

Example: Enabling ASP on a Dial Peer

```bash
media profile asp 4
!
media class 1
  asp profile 4
!
dial-peer voice 2100 pots
destination-pattern 2100
incoming called-number 1100
```
Feature Information for Acoustic Shock Protection

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

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

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Acoustic Shock Protection</td>
<td>15.2(2)T, 15.2(3)T</td>
<td>Acoustic Shock Protection (ASP) is a voice circuit-breaker feature that is designed to protect users, especially those wearing headsets, from exposure to loud, sustained, and piercing tones, such as those produced by a fax machine. It is a workplace-safety feature for voice calls. ASP is supported on TDM gateways and on Cisco UBE. The following commands were introduced or modified: media profile asp, media service.</td>
</tr>
<tr>
<td>Acoustic Shock Protection</td>
<td>Cisco IOS XE Release 3.6S</td>
<td>Acoustic Shock Protection (ASP) is a voice circuit-breaker feature that is designed to protect users, especially those wearing headsets, from exposure to loud, sustained, and piercing tones, such as those produced by a fax machine. It is a workplace-safety feature for voice calls. ASP is supported on TDM gateways and on Cisco UBE. In Cisco IOS XE Release 3.6S, this feature was implemented on the Cisco Unified Border Element (Enterprise) The following commands were introduced or modified: media profile asp, media service.</td>
</tr>
</tbody>
</table>
Feature Information for Acoustic Shock Protection
Noise Reduction

Noise Reduction (NR) is a voice enhancement process that improves the quality of incoming speech that has already been corrupted with background noise; for example, a voice conference participant speaking on a cell-phone in a car. NR works best with steady state broadband noises like engine noise but not as well with impulsive noises like nearby chatter.

- Finding Feature Information, page 137
- Prerequisites for Noise Reduction, page 137
- Restrictions for NR, page 138
- Information About NR, page 138
- How to Configure NR, page 139
- Configuration Examples for the NR feature, page 144
- Feature Information for Noise Reduction, page 145

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Prerequisites for Noise Reduction

Cisco Unified Border Element

- Cisco IOS Release 15.2(2)T, or a later release must be installed and running on your Cisco Unified Border Element.
Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.6S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions for NR

- Supported only on PVDM3.
- Supported only on flex codec complexity.
- No support for H.32x video call, complex forking calls, and fax and modem calls.
- No support for Time-Division Multiplexing (TDM) hairpin call.
- Configurations under POTS dial peer has higher priority over VoIP dial peer for NR.
- Configurations under the dial peer has higher priority than configurations at the global level.
- No support for conference calls, IP/SIP phones, and the Skinny Client Control Protocol (SCCP).
- CLI supports enabling NR but not disabling NR.
- No support for dynamically enabling or disabling NR during a call.

Information About NR

Noise Reduction

Noise Reduction (NR) is an adaptive signal processing algorithm on the Digital Signal Processor (DSP) that analyzes incoming audio, extracts a fingerprint of the background noise during talker pauses, and then performs ongoing spectral subtraction of this noise after a short training period (a few seconds). NR constantly adapts to changes in background noises over time.

NR can affect music on hold signals by making the music quieter. NR may disrupt fax/modem/TDD devices, although it is designed to self-disable in those cases. Use modem-relay mode for reliable fax/modem transmission. NR is supported on TDM gateways (TDM-VoIP and TDM-TDM) and on the Cisco Unified Border Element (Cisco UBE).

Some of the best practices for NR are as follows:

- Use default values.
- Do not use NR on dial peers associated with fax machines. Use fax or modem-relay modes for those dial peers.
- NR, when used without dynamic user control of intensity (as is the case with gateways), must be used at a low intensity (default or lower) since it is always on. High intensity is dramatic for demonstrations with loud background noises, but the NR process itself will degrade "normal" calls if NR is run at high intensity.
How to Configure NR

Creating the Media Profile for NR

Perform this task to create a media profile to configure noise reduction parameters.

SUMMARY STEPS

1. enable
2. configure terminal
3. media profile nr tag
4. intensity level
5. noisefloor level
6. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Device&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> media profile nr tag</td>
<td>Creates the media profile to configure noise reduction parameters and enters media profile configuration mode. The range for the media profile tag is from 1 to 10000.</td>
</tr>
<tr>
<td>Example: Device(config)# media profile nr 2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> intensity level</td>
<td>Configures the intensity level or depth of the noise reduction process. The range is from 0 to 6.</td>
</tr>
<tr>
<td>Example: Device(cfg-mediaprofile)# intensity 2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> noisefloor level</td>
<td>Configures the noise level, in dBm, above which NR will operate. NR will allow noises quieter than this level to pass without processing. The range is from -58 to -20.</td>
</tr>
<tr>
<td>Example: Device(cfg-mediaprofile)# noisefloor -50</td>
<td></td>
</tr>
</tbody>
</table>
Creating the Media Class to Enable NR

After the media profile is created, you must create a media class to enable noise reduction. Perform this task to create a media class.

SUMMARY STEPS

1. enable
2. configure terminal
3. media class tag
4. nr profile tag
5. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>enable</td>
</tr>
<tr>
<td>Example:</td>
<td>Device&gt; enable</td>
</tr>
<tr>
<td></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>configure terminal</td>
</tr>
<tr>
<td>Example:</td>
<td>Device# configure terminal</td>
</tr>
<tr>
<td></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>media class tag</td>
</tr>
<tr>
<td>Example:</td>
<td>Device(config)# media class 2</td>
</tr>
<tr>
<td></td>
<td>Creates the media class to enable the noise reduction feature and enters media class configuration mode. The range for the media class tag is from 1 to 10000.</td>
</tr>
</tbody>
</table>
### Configuring the Media Class at a Dial Peer Level for NR

Perform this task to configure the media class for a dial peer.

**SUMMARY STEPS**

1. `enable`
2. `configure terminal`
3. `dial-peer voice tag pots`
4. `media-class tag`
5. `end`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><code>enable</code> Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Device&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><code>configure terminal</code> Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Device# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><code>dial-peer voice tag pots</code> Defines a particular dial peer and enters the dial-peer voice configuration mode. The range for the dial-peer voice tag is from 1 to 1073741823.</td>
</tr>
<tr>
<td>Example:</td>
<td>Device(config)# dial-peer voice 20 pots</td>
</tr>
</tbody>
</table>
### Purpose

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 4</strong> media-class <em>tag</em></td>
<td>Applies the media class to the specific dial peer. The range for the media class tag number is from 1 to 10000.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(config-dial-peer)# media-class 2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> end</td>
<td>Returns to the privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(config-dial-peer)# end</td>
<td></td>
</tr>
</tbody>
</table>

---

**Configuring the Media Class Globally for NR**

Perform this task to configure a media class globally.

**SUMMARY STEPS**

1. enable
2. configure terminal
3. media service
4. enhancement
5. tdm *tag*
6. end

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device&gt; enable</td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>-------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 3</strong> media service</td>
<td>Enters media service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(config)# media service</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> enhancement</td>
<td>Enters the submode enhance of media service.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(cfg-mediaservice)# enhancement</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> tdm tag</td>
<td>Applies the TDM call globally. The range for the media class tag number is from 1 to 10000.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(cfg-service-enhance)# tdm 2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> end</td>
<td>Returns to the privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(config-dial-peer)# end</td>
<td></td>
</tr>
</tbody>
</table>

**Verifying NR**

Perform this task to verify the voice quality metrics.

**SUMMARY STEPS**

1. enable
2. show call active voice stats | b pid:

**DETAILED STEPS**

**Step 1** enable

**Example:** Device> enable

Enables privileged EXEC mode.

**Step 2** show call active voice stats | b pid:
Example:
Device# show call active voice stats | b pid:1300

11EC : 5 09:14:25.971 PDT Thu Jul 28 2011.1 +1130 pid:1300 Answer 1300 active dur 00:01:36 tx:17/321 rx:17/321 dscp:0 media:0
DSP/TX: PK=17, SG=0, NS=1, DU=90570, VO=320
DSP/RX: PK=17, SG=0, CF=1, RX=90570, VO=320, BS=0, BP=0, LP=0, EP=0

... 
DSP/DL: RT=0, ED=0
MIC Direction:
DSP/NR: NR=1, ND=0, LV=257, IN=1, PN=0, ON=0
DSP/AS: AE=1, AD=0, AV=0, AM=0, NT=0, DT=0, TD=0, LF=0, LD=0
EAR Direction:
DSP/NR: NR=0, ND=0, LV=0, IN=0, PN=0, ON=0
DSP/AS: AE=0, AD=0, AV=0, AM=0, NT=0, DT=0, TD=0, LF=0, LD=0

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1

Displays information about digital signal processing (DSP) voice quality metrics.

---

Troubleshooting Tips

The following commands can help troubleshoot NR:

- `debug voip hpi all`
- `debug voip dsmp all`
- `debug voip dsm all`
- `debug voip vtsp all`
- `debug vpm dsp all`

---

Configuration Examples for the NR feature

Example: Enabling NR globally

```
media profile nr 1
   intensity 1
!
media profile nr 2
!
media profile nr 3
   intensity 2
!
media profile nr 4
   intensity 3
!
media profile nr 5
   intensity 2
```
Feature Information for Noise Reduction

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.
# Table 11: Feature Information for Noise Reduction

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Noise Reduction</td>
<td>15.2(2)T, 15.2(3)T</td>
<td>Noise Reduction (NR) is a voice enhancement or restoration process that improves the quality of incoming speech that has already been corrupted with background noise. NR is supported on TDM gateways and on the Cisco UBE. The following commands were introduced or modified: <code>intensity</code>, <code>media profile nr</code>, <code>media service</code>, and <code>noisefloor</code>.</td>
</tr>
<tr>
<td>Noise Reduction</td>
<td>Cisco IOS XE Release 3.6S</td>
<td>Noise Reduction (NR) is a voice enhancement or restoration process that improves the quality of incoming speech that has already been corrupted with background noise. NR is supported on TDM gateways and on Cisco UBE. In Cisco IOS XE Release 3.6S, this feature was implemented on the Cisco Unified Border Element (Enterprise). The following commands were introduced or modified: <code>intensity</code>, <code>media profile nr</code>, <code>media service</code>, <code>noisefloor</code>.</td>
</tr>
</tbody>
</table>
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

Finding Feature Information, page 147
Prerequisites for Hardware Echo Cancellation, page 148
Restrictions for Hardware Echo Cancellation, page 148
How to Configure Hardware Echo Cancellation, page 150

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.
Prerequisites for Hardware Echo Cancellation

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

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 page 150). 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

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
vtp level 0 state = S_CONNECT
callid 0x0001 B01 state S_TSP_CONNECT c1ld 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

***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

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

***HARDWARE ECHO CANCELLER STATISTICS***

Echo Cancellation: 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
Ecanc 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
cmohanann-3845#

***HARDWARE ECHO CANCELLER STATISTICS***

Echo Cancellation: 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
Ecanc Status words 0x7C, 0x1001
EC Lib version: 9183.890
How to Configure Hardware Echo Cancellation

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
- Enter your password if prompted. |
| **Example:**  
Router> enable | |
| **Step 2** configure terminal | Enters global configuration mode. |
| **Example:**  
Router# configure terminal | |
| **Step 3** card type {e1 | t1} slot subslot | 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. |
| **Example:**  
Router(config)# card type t1 1 0 | |
| **Step 4** voice-card slot | Enters voice card configuration mode.  
- Specify the slot location using a value from 0 to 5. |
| **Example:**  
Router(config)# voice card 1 | |
| **Step 5** voice-port {slot-number / subunit-number / port | slot / port : ds0-group-number} | 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. |
| **Example:**  
Router(voice-card)# voice-port 3/0:0 | |
### Purpose

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>• The <code>port</code> argument is the voice interface card location. Valid entries are 0 to 3.</td>
<td></td>
</tr>
<tr>
<td>• The <code>ds0-group-number</code> 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.</td>
<td></td>
</tr>
</tbody>
</table>

**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 page 148 for more information.

### 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 configuration 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**

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.
```

<table>
<thead>
<tr>
<th>ECAN CH</th>
<th>ASSIGNED</th>
<th>DSP ID</th>
<th>VOICEPORT</th>
<th>EC</th>
<th>NLP</th>
<th>COV</th>
<th>LAW</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>yes</td>
<td>1/1</td>
<td>1/0:1.1</td>
<td>on</td>
<td>off</td>
<td>on</td>
<td>u-Law</td>
</tr>
</tbody>
</table>

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`

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

1 active call found
number of hardware ecancel channels: 1
number of software ecanel channels: 0

**show call active voice echo-canceller CallID Example**

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

Router# `show call active voice echo-canceller 0xE71`

```
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
```

Voice Port Configuration Guide, Cisco IOS Release 15M&T
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:

- Finding Feature Information, page 155
- Prerequisites for NextPort Services, page 155
- Information About NextPort Voice Services, page 156
- How to Configure NextPort Services, page 158

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Prerequisites for NextPort Services

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

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 "Hot 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 1960 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, on page 157.
- 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.
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

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 **voicecap entry** command and are applied with the **voicecap configure** command.

**Voice Tuning**

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.

Note
You must have specific knowledge of the behavior of the telephone network in order to use these voice capabilities.

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 A combined (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 Services

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

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2</td>
<td>Command or Action</td>
</tr>
<tr>
<td>--------</td>
<td>------------------</td>
</tr>
<tr>
<td></td>
<td>configure terminal</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
</tr>
<tr>
<td></td>
<td>Router# configure terminal</td>
</tr>
</tbody>
</table>

**Step 3**  
**Command or Action:** `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**  
**Command or Action:** `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.  
  - `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.

**Step 5**  
**Command or Action:** `end`  
**Example:**  
Router(config-spe)# end  

Completes the download and exits SPE configuration mode.
**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-Based Voice Tuning and Echo Cancellation Guide".

**Setting Voice Tuning Parameters with V Registers**

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

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

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 and Echo Cancellation Guide".

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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td></td>
</tr>
</tbody>
</table>
|                   | • Enter your password if prompted.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><code>configure terminal</code></td>
<td>Example: Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>Creates a voicecap.</td>
</tr>
<tr>
<td><code>voicecap entry name string</code></td>
<td>Example: Router(config)# voicecap entry qualityERL v270=120</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>Enters voice-port configuration mode on the selected slot and port.</td>
</tr>
<tr>
<td><code>voice-port slot / port :D</code></td>
<td>Example: Router(config)# voice-port 3/0:D</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>Applies a voicecap.</td>
</tr>
<tr>
<td><code>voicecap configure name</code></td>
<td>Example: Router(config-voiceport)# voicecap configure qualityERL</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>Exits voice-port configuration mode and completes the configuration.</td>
</tr>
<tr>
<td><code>exit</code></td>
<td>Example: Router(config-voiceport)# exit</td>
</tr>
</tbody>
</table>

**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:

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:

Example:

```
Router# show port operational-status 1/0
Slot/SPE/Port -- 1/0
   Service Type :Voice service
   Voice Codec :G.711 u-law
   Echo Canceller Length :8 ms
   Echo Cancellation Control :Echo cancellation - enabled
   .
   .
   .
   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/Tx 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 : 0
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:

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 Talker Level : -86.3/-57.0 dBm
  TX/RX Mean Speech level : -86.7/-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 ssr src value : 391643394
  Total ssr src 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:

**Example:**

```bash
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:

**Example:**

```bash
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:

**Example:**

```bash
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
```
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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
</tbody>
</table>
### Configuration Examples for NextPort Services

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
```
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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>- Enter your password if prompted.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>configure terminal</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>voicecap entry name string</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# voicecap entry npecho_ctrl v2=512 v51=32769</td>
</tr>
<tr>
<td></td>
<td>Creates a voicecap entry.</td>
</tr>
<tr>
<td></td>
<td>• The name argument is a word that uniquely identifies this voicecap.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• Example settings:</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>voice-port slot / port</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# voice-port 3/0</td>
</tr>
<tr>
<td></td>
<td>Enters voice-port configuration mode on the selected slot and port.</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td>voicecap configure name</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# voicecap configure npecho_ctrl</td>
</tr>
<tr>
<td></td>
<td>Applies a voicecap entry to the voice port.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>To configure multiple voice ports, repeat Step 4 and Step 5 for each voice port.</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td>exit</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voiceport)# exit</td>
</tr>
<tr>
<td></td>
<td>Exits voice-port configuration mode.</td>
</tr>
</tbody>
</table>
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.

```plaintext
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** = dcf
- **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: 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
*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 26 21:40:51.735: 00:00:14: Port Trace Event:
```
NextPort-Based Voice Tuning and Echo Cancellation

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

*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 : 0x30000000
*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 : Session 0x0144 Received Tone Off ntf for code 0x07 from IP side

Router#*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 : 0x30000000
*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 : 48
*Apr 26 21:40:51.735: Data : Session 0x0144 Received /ANSam tone 0x0f from IP side

*Apr 26 21:40:51.735: 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 : 0x30000000
*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 : 0x30000000
*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

Router#*Apr 26 21:40:51.739: 00:00:08: Port Trace Event:
*Apr 26 21:40:51.739: Port : 3/00
*Apr 26 21:40:51.739: Address : 0x30000000
*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 : 0x30000000
*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 : 0x30000000
*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

Voice Port Configuration Guide, Cisco IOS Release 15M&T

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

• Finding Feature Information, page 175
• Information About Verifying Voice-Port Configurations, page 175
• How to Verify Voice-Port Configurations, page 175

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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

Information About Verifying 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 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 Dual-Tone Multifrequency (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.

Step 4
`show voice port`
Use this command to verify voice-port parameter settings. Refer to the table below for the appropriate syntax for your platform. For sample output, see the `show voice port Command Examples`, on page 179.

Table 12: Show Voice Port Command Syntax

<table>
<thead>
<tr>
<th>Platform</th>
<th>Voice Port Type</th>
<th>Command Syntax</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco 1750</td>
<td>Analog</td>
<td>`show voice port [slot / port</td>
</tr>
<tr>
<td>Cisco 2600 series</td>
<td>Analog</td>
<td>`show voice port [slot / port</td>
</tr>
<tr>
<td>Cisco 3600 series</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco 3700 series</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco MC3810</td>
<td>Analog</td>
<td>`show voice port [slot / port</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Digital</td>
<td>`show voice port [slot / port : ds0-group-number</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Digital</td>
<td>`show voice port [slot : ds0-group-number</td>
</tr>
</tbody>
</table>
### 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:

**Example:**

```
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`, on page 182.

**Example:**

```
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`, on page 183.
Example:

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, on page 184.

Example:

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, on page 184.

Example:

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, on page 185.

Example:

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**

**Cisco 3640 Router Analog Voice Port**

The following output is from a Cisco 3640 router:

```
Router# show voice port summary
```

<table>
<thead>
<tr>
<th>PORT</th>
<th>CH</th>
<th>SIG-TYPE</th>
<th>ADMIN</th>
<th>OPER</th>
<th>STATUS</th>
<th>STATUS</th>
<th>EC</th>
</tr>
</thead>
<tbody>
<tr>
<td>======</td>
<td>====</td>
<td>----------</td>
<td>-------</td>
<td>------</td>
<td>--------</td>
<td>--------</td>
<td>----</td>
</tr>
</tbody>
</table>
```
Cisco MC3810 Digital Voice Port

The following output is from a Cisco MC3810:

```plaintext
Router# show voice port summary

+---+---+--------+-------+---+---+---+---+
<table>
<thead>
<tr>
<th>IN</th>
<th>OUT</th>
</tr>
</thead>
<tbody>
<tr>
<td>----</td>
<td>----</td>
</tr>
<tr>
<td>-----</td>
<td>----</td>
</tr>
<tr>
<td>-----</td>
<td>----</td>
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<td>-----</td>
<td>----</td>
</tr>
<tr>
<td>-----</td>
<td>----</td>
</tr>
<tr>
<td>-----</td>
<td>----</td>
</tr>
</tbody>
</table>
```

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:

```plaintext
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 EandM Voice Port

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

Router# show voice port 1
/0:1
receive 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
Ringing Time Out is set to 60 s
Compaunding 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 EandM Voice Port

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

Router# show voice port 1

receive 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:
     REFROM 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
```
### 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:

<table>
<thead>
<tr>
<th>Router# show voice dsp</th>
<th>TYPE</th>
<th>DSP</th>
<th>CH</th>
<th>CODEC</th>
<th>VERS</th>
<th>STATE</th>
<th>STATE</th>
<th>RST</th>
<th>AI</th>
<th>PORT</th>
<th>TS</th>
<th>ABORT</th>
<th>TX/RX-PAK-CNT</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>DNA</td>
<td>C549</td>
<td>010</td>
<td>g729r8</td>
<td>3.3</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>1</td>
<td>0</td>
<td>67400/85384</td>
</tr>
<tr>
<td></td>
<td>DNA</td>
<td>01</td>
<td>g729r8</td>
<td>.8</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>7</td>
<td>0</td>
<td>67566/83623</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DNA</td>
<td>02</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>13</td>
<td>0</td>
<td>65675/81851</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DNA</td>
<td>03</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>20</td>
<td>0</td>
<td>65530/83610</td>
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</tr>
<tr>
<td></td>
<td>DNA</td>
<td>C549</td>
<td>011</td>
<td>g729r8</td>
<td>3.3</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>2</td>
<td>0</td>
<td>66820/84799</td>
</tr>
<tr>
<td></td>
<td>DNA</td>
<td>01</td>
<td>g729r8</td>
<td>.8</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>8</td>
<td>0</td>
<td>59028/66946</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DNA</td>
<td>02</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>14</td>
<td>0</td>
<td>65591/81084</td>
<td></td>
</tr>
<tr>
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<td>DNA</td>
<td>03</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
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<td>0</td>
<td>1/015</td>
<td>21</td>
<td>0</td>
<td>66336/82739</td>
<td></td>
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<tr>
<td></td>
<td>DNA</td>
<td>C549</td>
<td>012</td>
<td>g729r8</td>
<td>3.3</td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>3</td>
<td>0</td>
<td>59036/65245</td>
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<tr>
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<td>DNA</td>
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<td>.8</td>
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<td>65826/81950</td>
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<td>g729r8</td>
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<td>1/015</td>
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<td>65606/80733</td>
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<tr>
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<td>busy</td>
<td>idle</td>
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<td>0</td>
<td>1/015</td>
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<td>65577/83532</td>
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<td>DNA</td>
<td>C549</td>
<td>013</td>
<td>g729r8</td>
<td>3.3</td>
<td>busy</td>
<td>idle</td>
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<td>0</td>
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<td>g729r8</td>
<td>.8</td>
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<td>0</td>
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<td>10</td>
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<td>65647/82088</td>
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<td>g729r8</td>
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<td>idle</td>
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<td>g729r8</td>
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<td>busy</td>
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<td>.8</td>
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<td>1/015</td>
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<td>65664/81737</td>
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<td></td>
<td>DNA</td>
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<td>g729r8</td>
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<td>idle</td>
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<td>0</td>
<td>1/015</td>
<td>18</td>
<td>0</td>
<td>65607/81820</td>
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<tr>
<td></td>
<td>DNA</td>
<td>03</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>24</td>
<td>0</td>
<td>65589/83889</td>
<td></td>
</tr>
<tr>
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<td>DNA</td>
<td>C549</td>
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<td>g729r8</td>
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<td>busy</td>
<td>idle</td>
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<td>1/015</td>
<td>6</td>
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<td>66889/83331</td>
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<tr>
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<td>g729r8</td>
<td>.8</td>
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<td>idle</td>
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<td>0</td>
<td>1/015</td>
<td>12</td>
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<td>66690/81700</td>
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<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>19</td>
<td>0</td>
<td>66422/82099</td>
<td></td>
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<tr>
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<td>DNA</td>
<td>03</td>
<td>g729r8</td>
<td></td>
<td>busy</td>
<td>idle</td>
<td>0</td>
<td>0</td>
<td>1/015</td>
<td>25</td>
<td>0</td>
<td>65566/83852</td>
<td></td>
</tr>
</tbody>
</table>

### Verifying Analog and Digital Voice-Port Configurations

Examples
show voice call summary Command Examples

Cisco MC3810 Analog Voice Port

The following output is from a Cisco MC3810:

<table>
<thead>
<tr>
<th>PORT</th>
<th>CODEC</th>
<th>VAD</th>
<th>VTSP STATE</th>
<th>VPM STATE</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/1</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>FXSLS_CONNECT</td>
</tr>
<tr>
<td>1/2</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>FXSLS_ONHOOK</td>
</tr>
<tr>
<td>1/3</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>EM_ONHOOK</td>
</tr>
<tr>
<td>1/4</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>EM_ONHOOK</td>
</tr>
<tr>
<td>1/5</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>FXOLS_ONHOOK</td>
</tr>
<tr>
<td>1/6</td>
<td>-</td>
<td>-</td>
<td>-</td>
<td>FXOLS_ONHOOK</td>
</tr>
</tbody>
</table>

Cisco 3600 Series Router Digital Voice Port

The following output is from a Cisco 3600 series router:

<table>
<thead>
<tr>
<th>PORT</th>
<th>CODEC</th>
<th>VAD</th>
<th>VTSP STATE</th>
<th>VPM STATE</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/015.1</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.2</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.3</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.4</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.5</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.6</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.7</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.8</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.9</td>
<td>g729r8</td>
<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
<tr>
<td>1/015.10</td>
<td>g729r8</td>
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<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
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<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
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<td>y</td>
<td>S_CONNECT</td>
<td>S_TSP_CONNECT</td>
</tr>
</tbody>
</table>

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
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
DisconnectTime=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
cvVoIPCallHistoryICPf=0
This chapter provides information to assist you in analyzing and troubleshooting voice port problems.

- Troubleshooting Chart, page 187

**Troubleshooting Chart**

The table below lists some problems that you might encounter after configuring voice ports. It also provides some suggested remedies.

*Table 13: Troubleshooting Voice Port Configurations*

<table>
<thead>
<tr>
<th>Problem</th>
<th>Suggested Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>No connectivity</td>
<td>Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the <em>Cisco IOS IP Configuration Guide</em>.</td>
</tr>
<tr>
<td>Problem</td>
<td>Suggested Action</td>
</tr>
<tr>
<td>-------------------------------</td>
<td>--------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>No connectivity</td>
<td>Enter the <code>show controller t1</code> or <code>show controller e1</code> command with the controller number for the voice port you are troubleshooting. This will tell you:</td>
</tr>
<tr>
<td></td>
<td>• If the controller is up. If it is not, use the <code>no shutdown</code> command to make it active.</td>
</tr>
<tr>
<td></td>
<td>• Whether alarms have been reported.</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>See the &quot;show controller Command: Examples&quot; section in the &quot;Verifying Analog and Digital Voice-Port Configurations&quot; chapter for output.</td>
</tr>
<tr>
<td>No connectivity</td>
<td>Enter the <code>show voice port</code> command with the voice port number that you are troubleshooting, which will tell you:</td>
</tr>
<tr>
<td></td>
<td>• If the voice port is up. If it is not, use the <code>no shutdown</code> command to make it active.</td>
</tr>
<tr>
<td></td>
<td>• What parameter values have been set for the voice port, including default values (these do not appear in the output for the <code>show running-config</code> command). If these values do not match those of the telephony connection you are making, reconfigure the voice port.</td>
</tr>
<tr>
<td></td>
<td>See the &quot;show voice port Command: Examples&quot; section in the &quot;Verifying Analog and Digital Voice-Port Configurations&quot; chapter for sample output.</td>
</tr>
<tr>
<td>Telephony device buzzes or does not ring</td>
<td>Use the <code>show voice port</code> command to confirm that the <code>ring frequency</code> command is configured correctly. It must match the connected telephony equipment and may be country-dependent.</td>
</tr>
<tr>
<td>Distorted speech</td>
<td>Use the <code>show voice port</code> command to confirm the <code>cptone</code> keyword setting (also called <code>region tone</code>) is US.</td>
</tr>
<tr>
<td></td>
<td>Setting a wrong <code>cptone</code> could result in faulty voice reproduction during analog-to-digital or digital-to-analog conversions.</td>
</tr>
<tr>
<td>Music on hold is not heard</td>
<td>Reduce the configured level for the <code>music-threshold</code> command.</td>
</tr>
<tr>
<td>Problem</td>
<td>Suggested Action</td>
</tr>
<tr>
<td>------------------------------------------------------------------------</td>
<td>-------------------------------------------------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Background noise is not heard</td>
<td>Enable the <strong>comfort-noise</strong> command.</td>
</tr>
<tr>
<td>Long pauses occur in conversation; like speaking on a walkie-talkie</td>
<td>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.</td>
</tr>
<tr>
<td>Jerky or choppy speech</td>
<td>Variable delay, or jitter, is being introduced by congestion in the packet network. Two possible remedies are to:</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• Increase the size of the jitter buffer with the <strong>playout-delay</strong> command. (Refer to the <em>Cisco IOS Voice Troubleshooting and Monitoring Guide</em>.)</td>
</tr>
<tr>
<td>Clipped or fuzzy speech</td>
<td>• Reduce input gain. (Refer to the <em>Cisco IOS Voice Troubleshooting and Monitoring Guide</em>.)</td>
</tr>
<tr>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td>Clipped speech</td>
<td>Reduce the input level at the listener's router. (Refer to the <em>Cisco IOS Voice Troubleshooting and Monitoring Guide</em>.)</td>
</tr>
<tr>
<td>Volume too low or missed Dual-Tone Multifrequency (DTMF)</td>
<td>Increase speaker’s output level or listener’s input level. (Refer to the <em>Cisco IOS Voice Troubleshooting and Monitoring Guide</em>.)</td>
</tr>
<tr>
<td>Echo interval is greater than 25 ms (sounds like a separate voice)</td>
<td>Configure the <strong>echo-cancel enable</strong> command and increase the value for the <strong>echo-cancel coverage</strong> keyword. (See the &quot;Configuring Echo Cancellation&quot; section.)</td>
</tr>
<tr>
<td>Problem</td>
<td>Suggested Action</td>
</tr>
<tr>
<td>------------------</td>
<td>----------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Too much echo</td>
<td>Reduce the output level at the speaker's voice port. (Refer to the Cisco IOS Voice Troubleshooting and Monitoring Guide.)</td>
</tr>
</tbody>
</table>

**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.*
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