



Digital E1 Packet Voice Trunk Network Module Interfaces

This document describes how to configure digital E1 packet voice trunk network module interfaces on Cisco 2600 and 3600 series routers and includes the following sections:

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Feature Overview

Digital E1 packet voice trunk network modules for Cisco 2600 and 3600 series routers allow enterprises or service providers, using the equipped routers as customer premises equipment, to deploy digital voice and fax relay. These modules receive constant bit-rate telephony information over E1 interfaces and can convert that information to a compressed format, so that it can be transmitted as Voice over IP (VoIP), Voice over Frame Relay (VoFR), and Voice over ATM (VoATM).

Benefits

Digital E1 packet voice trunk network modules allow Cisco 2600 and 3600 series routers to provide E1 connectivity to private branch exchanges (PBXs) or to a central office (CO). With digital E1 connectivity, Cisco 2600 and 3600 series routers can provide greater voice density for enterprise and service provider VoIP networks. A digital E1 packet voice trunk network module is a complete solution, made up of a network module with installed packet voice data modules (PVDMs), and one E1 multiflex trunk voice/WAN interface card with either one or two E1 ports.

Restrictions

The following restrictions apply to digital E1 packet voice trunk network module configuration:

- Group 4 fax is not supported.
- The high-density voice network module has one slot for a voice/WAN interface card (VWIC); VWICs supply one or two ports. Only the dual-mode (voice/WAN) multiflex trunk cards are supported in the digital E1 packet voice trunk network module, not older VICs. For more information, see the “Prerequisites” section on page 4.
- Drop-and-Insert capability is supported only between two ports on the same multiflex card.
- Common-channel signaling (CCS) and Primary Rate Interface (PRI) are not supported.
- R2 signaling is not supported.
- Voice over ATM—including AAL5 encapsulation, circuit emulation service (CES), and AAL2—is not supported for VoATM on the Cisco 2600 series router.
- Digital E1 voice is manageable through Simple Network Management Protocol (SNMP) using Release 2.0 of Cisco Voice Manager.

Related Documents

The following documents can help you understand how to install Cisco 2600 and 3600 series routers:

- *Cisco 2600 Series Hardware Installation Guide*
- *Quick Start Guide Cisco 2600 Series Cabling and Setup*
- *Software Configuration Guide*
- *Cisco 3660 Router Cabling and Setup Quick Start Guide*
- *Cisco 3600 Series Hardware Installation Guide*
- *Cisco Network Modules Hardware Installation Guide for Cisco 3600 Series and Cisco 2600 Series Routers*

The following Cisco IOS Release 12.1 documents are also helpful:

- *Cisco IOS Multiservice Applications Configuration Guide*
- *Cisco IOS Multiservice Applications Command Reference*

The following documents can help you troubleshoot ISDN, PRI, and BRI connections:

- *Internetwork Troubleshooting Guide*
- *Cisco IOS Debug Command Reference*

For information about supported hardware on a Cisco 2600 or 3600 series router, go to the following URLs:

- http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis2600/index.htm
- http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3600/index.htm
- For the Voice over IP Quick Start Guides, go to:
http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis3600/voice/4936vqsg.htm

Supported Platforms

This feature is supported on the following platforms:

- Cisco 2610
- Cisco 2611
- Cisco 2612
- Cisco 2613
- Cisco 2620
- Cisco 2621
- Cisco 3620
- Cisco 3640
- Cisco 3662
- Cisco 3661

Supported Standards, MIBs, and RFCs

MIBs

- CISCO-ENTITY-VENDORTYPE-OID-MIB
- OLD-CISCO-CHASSIS-MIB
- CAS_INTF_MIB

Standards

- G.711 A Law at 64,000 bps
- G.711 u Law at 64,000 bps
- G.723.1 Annex A at 5,300 bps
- G.723.1 Annex A at 6,300 bps
- G.723.1 at 5,300 bps
- G.723.1 at 6,300 bps
- G.726 at 16,000 bps
- G.726 at 24,000 bps
- G.726 at 32,000 bps
- G.728 at 16,000 bps
- G.729 at 8,000 bps
- G.729 Annex A at 8,000 bps
- G.729 Annex B at 8,000 bps
- G.729 Annex B with Annex A at 8,000 bps

RFCs

- RFC 1890
- RFC 1889

Prerequisites

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

- Obtain E1 service from your service provider or PBX.
- Install Cisco IOS Release 12.1(2)T or a later release. The *minimum* DRAM memory requirements to support digital E1 packet voice trunk network modules are as follows:
 - 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 might 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.

- Install one of the following high-density E1 network modules in the router chassis:
 - Single-Port 30-Channel E1 High-Density Voice Network Module (NM-HDV-1E1-30)
 - Single-Port Enhanced 30-Channel E1 High-Density Voice Network Module (NM-HDV-1E130E)
 - Dual-Port 60-Channel High-Density Voice Network Module (NM-HDV-2E1-60)



Note You can install one module in a Cisco 2600 series router or a Cisco 3620 router. A Cisco 3640 router can support three modules, and you can install as many as six modules in a Cisco 3660 router.

- Install at least one PVDM-12 in the high-density digital E1 network module if it is not already equipped. The digital E1 packet voice trunk network module contains five 72-pin SIMM sockets or banks, numbered 0 through 4, for PVDMs. Each socket can be filled with a single 72-pin PVDM. A digital E1 packet voice trunk network module can support the following numbers of channels:
 - When the digital 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, G.729 Annex B, G.723.1, G.728, and fax relay.
 - When the digital 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.



Note Each PVDM holds three digital signal processors (DSPs). With five PVDM slots populated, a total of 15 DSPs are provided. High-complexity codecs support two simultaneous calls on each DSP, while medium-complexity codecs support four calls on each DSP.

- Install at least one dual-mode voice/WAN interface card (VWIC) for a voice connection if a VWIC was not included with the network module. You can install one VWIC (providing one or two line interfaces) in the digital E1 packet voice trunk network module. Only the one- and two-port E1 multiflex trunk interface cards (VWIC-1MFT-E1, VWIC-2MFT-E1, VWIC-2MFT-E1-DI) are supported.

For Drop-and-Insert capability, you must install a two-port Drop-and-Insert E1 multiflex trunk voice/WAN interface card (VWIC-2MFT-E1-DI). To install a VWIC in a network module, see *Cisco WAN Interface Cards Hardware Installation Guide*.

- Install at least one other network module or WAN interface card to provide the connection to the IP LAN or WAN.
- Establish a working IP, Frame Relay, or ATM network. For more information about configuring IP, refer to the *Cisco IOS IP and IP Routing Configuration Guide* for Cisco IOS Release 12.1.
- Complete your company's dial plan.
- Establish a working telephony network based on your company's dial plan.

Cisco IOS Multiservice Applications Configuration Guide and *Cisco IOS Multiservice Applications Command Reference* for Cisco IOS Release 12.1 provide information about setting up voice networks.

Configuration Tasks


Complete the following tasks to configure a digital E1 packet voice trunk network module:

- Set up voice cards and E1 controllers.
- Configure serial and LAN interfaces.
- Set up voice ports.
- Configure voice dial peers.

Configuring Voice Card and E1 Controller Settings

The following steps specify codec settings for voice cards and set up E1 controllers for clocking and other E1 parameters, as well as for DS0 groups that define the channels for compressed voice and TDM groups for Drop-and-Insert capability.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# voice-card slot	Enter voice card configuration mode and specify the slot location by using a value from 0 to 5, depending on your router.

Command	Purpose
Step 3 Router(config-voice-ca)# codec complexity {high medium}	<p>Specify the codec complexity based on the codec standard you are using. High-complexity codecs support lower call density than do medium-complexity codecs. The number of channels supported is based on the number of PVDMs installed and the codec complexity. Here is a guideline:</p> <ul style="list-style-type: none"> • When the digital 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, G.729 Annex B, G.723.1, G.723.1 Annex A, G.728, and fax relay. • When the digital 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 <p>All voice cards in a router must use the same codec complexity setting.</p> <p>The keyword that you specify for codec complexity affects the choice of codecs available using the codec dial-peer configuration command. See Step 7 in “Configuring Voice Dial Peers” on page 19.</p> <p> Note You cannot change codec complexity while DS0 groups are defined. If they are already set up, use the no ds0-group command before resetting the codec complexity. For more information about the pri-group command, see Step 9.</p>
Step 4 Router(config-voice-ca)# Exit	Exit the voice card configuration mode.
Step 5 Router(config)# controller E1 slot/port	Enter controller configuration mode for the E1 controller at the specified <i>slot/port</i> location. Valid values for <i>slot</i> and <i>port</i> are 0 and 1.

Command	Purpose
Step 6 Router(config-controller)# clock source { line [primary] internal }	Configure controller E1 1/0 to specify the clock source. The line keyword specifies that the clock source is derived from the active line—rather than from the free-running internal clock. This is the default setting and is generally more reliable. These rules apply to clock sourcing on the E1 controller ports: <ul style="list-style-type: none"> • 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. See “Verifying Voice Card and Controller Settings” on page 14, for more information about configurations for clocking.
Step 7 Router(config-controller)# framing crc4	Set the framing according to your service provider’s instructions. Choose cyclic redundancy check 4 (CRC4) format.
Step 8 Router(config-controller)# linecode hdb3	Set the line encoding according to your service provider’s instructions. E1 uses high density bipolar 3 (HDB3) encoding (similar to alternative mark inversion, or AMI).

Command	Purpose
<p>Step 9 Router(config-controller)# cablelength long {gain26 gain36} {-15db -22.5db -7.5db 0db}</p> <p>or</p> <p>cablelength short {133 266 399 533 655}</p>	<p>(E1 interfaces only) The cable length setting must conform to the actual cable length you are using. For example, if you attempt to enter the cablelength short command on a long-haul E1 link, the command is rejected.</p> <p>To set a cable length longer than 655 feet for an E1 link, use the cablelength long command. The keywords are as follows:</p> <ul style="list-style-type: none"> • gain26 specifies the decibel pulse gain at 26. This is the default pulse gain. • gain36 specifies the decibel pulse gain at 36. • -15db specifies the decibel pulse rate at -15 decibels. • -22.5db specifies the decibel pulse rate at -22.5 decibels. • -7.5db specifies the decibel pulse rate at -7.5 decibels. • 0db specifies the decibel pulse rate at 0 decibels. This is the default pulse rate. <p>To set a cable length 655 feet or less for an E1 link, use the cablelength short command. There is no default for cablelength short. The keywords are as follows:</p> <ul style="list-style-type: none"> • 133 specifies a cable length from 0-133 feet. • 266 specifies a cable length from 134-266 feet. • 399 specifies a cable length from 267-399 feet. • 533 specifies a cable length from 400-533 feet. • 655 specifies a cable length from 534-655 feet. <p>If you do not set the cable length, the system defaults to a setting of cablelength long gain26 0db.</p>
<p>Step 10 Router(config-controller)# pri-group timeslots <i>timeslot-list</i></p>	<p>Enter a single timeslot number and a single range of values. For E1, the allowable values are from 1 to 31.</p>
<p>Step 11 Router(config-controller)# no shutdown</p>	<p>Activate the controller.</p>
<p>Step 12 Router(config-controller)# exit</p>	<p>Exit controller configuration mode.</p>

E1 Timing, Signaling, Framing, and Line Encoding

With the introduction of the digital E1 packet voice trunk network modules for the Cisco 2600 and 3600 series routers, you must set timing, signaling, framing, and line encoding. Digital E1 packet voice trunk network modules can connect to either a PBX (or similar telephony device) or to a central office (CO) to provide PSTN connectivity.

The differences that set E1 digital configuration apart from analog configuration are as follows:

- **Timing**—Analog interfaces do not require specific timing configuration. Digital E1 interfaces require not only that you set timing but that you consider the source of the timers.
- **Framing**—Analog interfaces do not require specific framing configuration. Digital E1 interfaces require that you configure for cyclic redundancy checking 4 (CRC-4) framing. Set the framing format to match that of the PBX or CO that connects to the digital E1 packet voice trunk network module.
- **Line Encoding**—Analog interfaces do not require specific line encoding configuration. Digital E1 interfaces require that you configure for high density bipolar 3 (HDB3) encoding (similar to alternative mark inversion, or AMI). Set the line encoding to match that of the PBX or CO that connects to the digital E1 packet voice trunk network module.

Timing

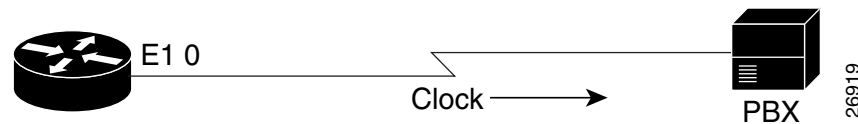
This section describes the five basic timing scenarios that can occur when a digital E1 packet voice trunk network module is connected to a PBX, CO, or both. In all of the examples below, the PSTN (or CO) and the PBX are interchangeable for purposes of providing or receiving clocking.

The digital E1 module has an onboard Phase-Lock Loop (PLL) chip that can either provide a clock source to both E1s or receive clocking that can drive the second E1 in the same digital E1 packet voice trunk network module. All timing commands are E1 controller configuration commands.

Single E1 Port Provides Clocking

In this scenario, the digital E1 module is the clock source for the connected device. The PLL generates the clock internally and drives the clocking on the E1 line.

Figure 1 Single E1 Port Providing Clock



The following configuration sets up this clocking method:

```
controller E1 1/0
framing crc4
linecoding hdb3
clock source internal
pri-group timeslots 1-31
```

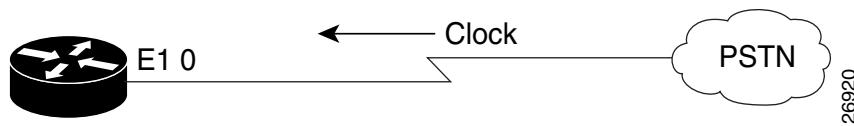


Note

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 provides a higher Stratum level.

Single E1 Port Receiving Clock from the Line

In this scenario, the digital E1 module receives clocking from the connected device (CO or PBX). The PLL clocking is driven by the clock reference on the receive (Rx) side of the E1 connection.

Figure 2 Single E1 Receiving Clock from Line

The following configuration sets up this clocking method:

```
controller E1 1/0
framing crc4
linecoding b8zs
clock source line
pri-group timeslots 1-31
```

Dual E1s Receiving Clocking from the Line

In this scenario, the digital E1 has two reference clocks, one from the PBX and another from the CO. Because the PLL can only derive clocking from one source, this case is more complex than the two preceding examples.

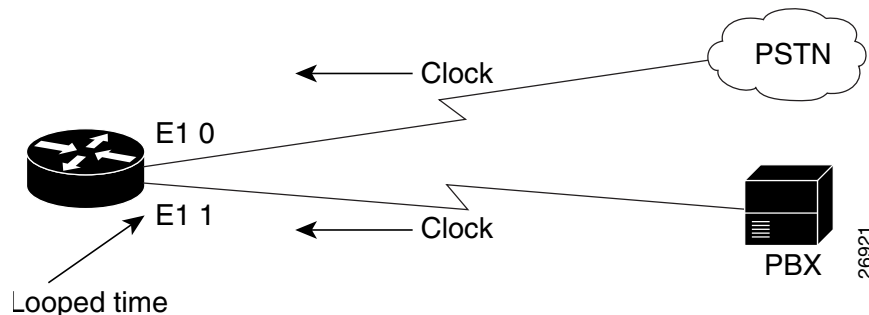
Before looking at the details, consider two important concepts that pertain to the clocking method:

- **Looped-Time Clocking**—The E1 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” the E1 so that the connected device has a viable clock and does not see slips. PBXs are not designed to accept slips on an 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 E1 port.
- **Slips**—These messages indicate that the E1 port is receiving clock information that is out of phase (out of synch). Because the router has only a single PLL, it can experience controlled slips while it receives clocking from two different time sources.

The router can usually handle controlled slips because its single PLL architecture anticipates them.



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

Figure 3 Dual E1s Receiving Line Clocking

In this scenario, the PLL derives clocking from the CO and puts the E1 port connected to the PBX into looped-time mode. This is usually the best method because the CO provides an excellent clock source (and usually requires that it provide that source) and a PBX usually must receive clocking from the other E1 port.

The following configuration sets up this clocking method:

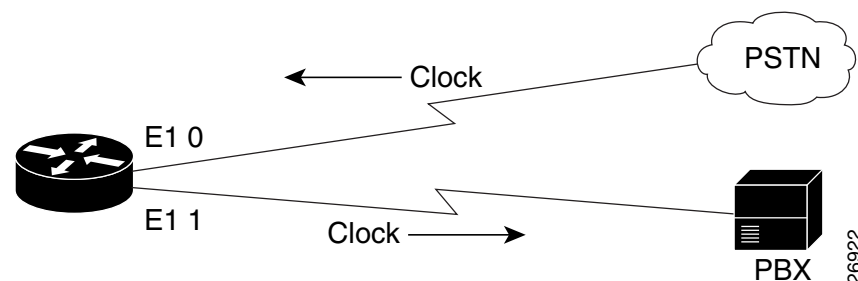
```
controller E1 1/0 << description - connected to the CO
framing crc4
linecoding hdb3
clock source line primary
pri-group timeslots 1-31
!
controller E1 1/1 << description - connected to the PBX
framing crc4
linecoding hdb3
clock source line
pri-group timeslots 1-31
```

The **clock source line primary** command tells the router to use this E1 port to drive the PLL. All other E1 ports configured as **clock source line** are then put into an implicit loop-timed mode. If the primary E1 port fails or goes down, the other E1 instead receives the clock that drives the PLL. In this configuration, E1 port 1/1 might see controlled slips, but these should not force it down. This method prevents the PBX from seeing slips.

Dual E1s—One Receives Clocking and One Provides Clocking

In this scenario, the digital E1 module receives clocking for the PLL from E1 0 and uses this clock as a reference to clock E1 1. If E1 0 fails, the PLL internally generates the clock reference to drive E1 1.

Figure 4 Dual E1s—One Receiving and One Providing Clocking



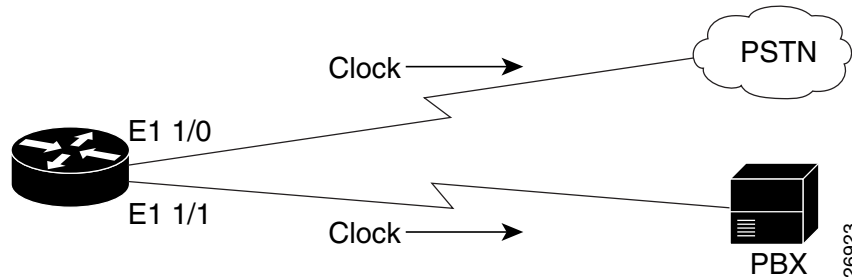
The following configuration sets up this clocking method:

```
controller E1 1/0
framing crc4
linecoding hdb3
clock source line
pri-group timeslots 1-31
!
controller E1 1/1
framing crc4
linecoding hdb3
clock source internal
pri-group timeslots 1-31
```

Dual E1s—Both Clocks from Router

In this scenario, the router generates the clock for the PLL and therefore for both E1s.

Figure 5 Dual E1s—Both Clocks from Router



The following configuration sets up this clocking method:

```
controller E1 1/0
framing crc4
linecoding hdb3
clock source internal
pri-group timeslots 1-31
!
controller E1 1/1
framing esf
linecoding b8zs
clock source internal
pri-group timeslots 1-31
```

Signaling

There are three types of signaling that you should consider for digital E1:

- **Channel-associated signaling (CAS)**—CAS means that instead of having a specific timeslot (such as an ISDN D channel in PRI) designated to provide signaling only, signaling bits (on-hook and off-hook) are within the sixth, twelfth, eighteenth, and twenty-fourth frames of each timeslot. CAS is often called robbed-bit signaling (RBS) because it takes bits from bearer channels and uses them for signaling. CAS must be specified on both ends of the E1 link and is enabled by default on digital E1 packet voice trunk port adapters.



Note Digital E1 packet voice trunk port adapters support E1 CAS. The digital E1 port adapter can support E&M wink-start, immediate-start, and delay-start signaling, as well as FXS and FXO ground-start and loop-start signaling.

- **E&M signaling**—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 because it provides better Answer Supervision (knowledge that the connected device is ready to answer the call). However, not all COs and PBXs can handle wink-start signaling. The E&M connection between the router and switch (CO or PBX) must use matching E&M signaling types otherwise calls cannot be connected properly. E&M signaling is defined with the **ds0-group** controller configuration command, as in the following example:

```
controller E1 1/0
ds0-group 1 timeslots 1-24 type e&m-wink-start
```



Note Currently, wink-start signaling provides only the functionality of feature-group B and not that of feature-group D, which will be supported in later releases.

- **FXO and FXS signaling**—While most digital E1 connections used for switch-to-switch (or switch-to-router) trunks are E&M connections, a digital E1 port adapter can also support FXS and FXO connections, which people normally use to provide emulated-OPX (Off-Premises eXtension) from a PBX to remote stations. As a general rule, FXO ports connect to FXS ports. 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. The FXO or FXS connection between the router and switch (CO or PBX) must use matching signaling, or calls cannot connect properly. FXS and FXO signaling are defined with the **ds0-group** controller configuration command, as in the following example:

```
controller E1 1/0
ds0-group 1 timeslots 1-24 type fxo-ground-start
```

or

```
controller E1 1/0
ds0-group 1 timeslots 1-24 type fxs-loop-start
```



Note While some switches (CO or PBX) can specify both an inbound and outbound signaling method, Cisco VoIP gateway routers can only specify one signaling type for both inbound and outbound calls. The switch inbound and outbound signaling types must match, or calls might only work in one direction.

Framing

Digital E1 packet voice trunk port adapters support two types of framing for E1 CAS: Extended Superframe (ESF) and Super Frame (SF), also called D4 framing. The framing type of the router and switch (CO or PBX) must match. The **framing** controller configuration command defines E1 framing, as in the following example:

```
controller E1 1/0
framing esf
```

or

```
controller E1 1/0
framing sf
```

Line Encoding

Digital E1 packet voice trunk port adapters support two types of line encoding for E1 CAS: B8ZS (bipolar-8 zero substitution) and AMI (alternate mark inversion). The line encoding of the router and switch (CO or PBX) must match. The **linecoding** controller configuration command defines E1 framing, as in the following example:

```
controller E1 1/0
linecoding b8zs
```

```

or
controller E1 1/0
linecoding ami

```

Verifying Voice Card and Controller Settings

To verify the configuration of voice card and controller settings, perform the following steps:

- Step 1** Enter the **show running-config** command to display the current voice-card setting. If no codec complexity is shown, the default of medium complexity is set. The following example shows an excerpt from the command output:

```

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

voice-card 1
  codec complexity high
.
.
.

```

- Step 2** Enter the privileged EXEC **show controllers E1** command to display the status of E1 controllers and display information about clock sources and other settings for the E1 ports:

```

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

```

Configuring Serial Interfaces

The way you set up serial and LAN interfaces depends on your application. To configure VoIP, you must at least set up IP addresses for serial interfaces. When a user dials enough digits to match a configured destination pattern, the telephone number is mapped to an IP host through the dial-plan mapper. The IP host has a direct connection to either the destination telephone number or a PBX that completes the call to the configured destination pattern.

This document does not explain all possible serial interface configuration options, nor does it show LAN interface configuration. For complete information, see the Cisco IOS Release 12.1 *Cisco IOS Interface Configuration Guide* and the *Cisco IOS Interface Command Reference*.

The “Configuration Example” section on page 37 shows a sample configuration. For more information about setting up voice networks, see *Cisco IOS Multiservice Applications Configuration Guide* for Cisco IOS Release 12.1.

**Note**

For information about monitoring serial interfaces in order to trigger a busyout condition on a voice port when an interface is down, see “Configuring Voice Ports” on page 17.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# interface serial <i>slot/port:channel-group</i>	Enter interface configuration mode for a serial interface that you specify by slot and port. The <i>channel-group</i> portion of the command is only required for channelized E1 interfaces. (For setting up channelized E1 interfaces, see <i>Dial Services Configuration Guide</i> for Cisco IOS Release 12.1.)
Step 3	Router(config-if)# ip address <i>ip-address mask</i>	Assign the IP address and subnet mask to the interface.
Step 4	Router(config-if)# isdn switch-type <i>primary-qsig</i>	Assign a switch type PRI or BRI interface, using <i>primary-qsig</i> for E1.
Step 5	Router(config-if)# isdn protocol-emulate [<i>user network</i>]	Configure the router’s PRI interface to serve as either the primary QSIG slave or the QSIG master.
Step 6	Router(config-if)# isdn incoming-voice [<i>data [56 64] modem [56 64]</i>]	Route incoming calls to the modem and treat them as analog data, bypass the modem, or treat them as data.
Step 7	Router(config-if)# fair-queue [<i>congestive-discard-threshold [dynamic-queues reservable-queues]</i>]	Initiate a fair-queue for congestion control.
Step 8	Router(config-if)# exit	Exit the interface.


Verifying Serial Interface Configuration

To verify serial interface configuration, enter the **show interfaces serial** privileged EXEC command, which displays the status of all serial interfaces or of a specific serial interface, as shown in the following example. You can use this command to check the encapsulation, IP addressing, and other settings:

```
Router# show interface serial10/0:0
Serial10/0:0 is up, line protocol is up
  Hardware is QUICC Serial
  Internet address is 1.156.1.1/24
  MTU 1500 bytes, BW 1536 Kbit, DLY 20000 usec,
    reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation HDLC, loopback not set
  Keepalive not set
  Last input 00:00:00, output 00:00:00, output hang never
  Last clearing of "show interface" counters never
  Input queue: 0/75/0 (size/max/drops); Total output drops: 0
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0 (size/max total/threshold/drops)
    Conversations 0/1/256 (active/max active/max total)
    Reserved Conversations 0/0 (allocated/max allocated)
  5 minute input rate 1000 bits/sec, 1 packets/sec
  5 minute output rate 1000 bits/sec, 1 packets/sec
    637 packets input, 64736 bytes, 0 no buffer
    Received 181 broadcasts, 0 runts, 5 giants, 0 throttles
    3617 input errors, 1506 CRC, 1646 frame, 0 overrun, 0 ignored, 0 abort
    682 packets output, 67213 bytes, 0 underruns
    0 output errors, 0 collisions, 1070 interface resets
    0 output buffer failures, 0 output buffers swapped out
    13 carrier transitions
  Timeslot(s) Used:1-24, Transmitter delay is 0 flags
```

Configuring Voice Ports

Follow these steps to set up voice ports to support the local and remote stations. Not all possible commands are shown here. To learn more, see *Cisco IOS Multiservice Applications Configuration Guide* and *Cisco IOS Multiservice Applications Command Reference* for Cisco IOS Release 12.1.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# voice-port <i>slot/port:pri-group-number</i>	<p>Enter voice-port configuration mode.</p> <p><i>slot</i> is the router location where the voice module is installed. Valid entries are from 0 to 3.</p> <p><i>port</i> indicates the voice interface card location. Valid entries are 0 or 1.</p> <p>There is only one voice port per controller for QSIG.</p> <p> Note This voice-port command syntax does not apply to analog voice network modules and voice interface cards. The latter are specified using <i>slot/subunit/port</i>, designating the router slot for the voice network module, the location of the voice interface card in the network module, and the port on the voice interface card.</p>
Step 3	Router(config-voice-port)# busyout monitor interface <i>interface number</i>	<p>(Optional) This command allows you to specify a LAN or WAN interface that will be monitored, and, when it is down, trigger a busyout (offhook) state on the voice port. This allows rerouting of calls. Busyout state for QSIG voice port implies that both the voice port and the signaling lines are down. You can enter the command repeatedly to specify as many interfaces, virtual interfaces, and subinterfaces as are required for a voice port.</p> <p>For example, if you enter the command three times so that three interfaces are monitored, the voice port only goes into busyout state when all three interfaces are down. When any one of the interfaces is operational, the busyout state is removed.</p>
Step 4	Router(config-voice-port)# comfort-noise	<p>(Optional) The comfort noise parameter is enabled by default. It 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 silence can be unsettling to callers.</p>

	Command	Purpose
Step 5	Router(config-voice-port)# echo-cancel enable	(Optional) This setting is enabled by default. Echo cancellation adds to the quality of voice transmissions by adjusting the echo that occurs on the interface due to impedance mismatches. Some echo is reassuring; echo over 25 milliseconds can cause problems.
Step 6	Router(config-voice-port)# echo-cancel coverage {16 24 32 8}	(Optional) This command adjusts the echo canceller by the specified number of milliseconds; the default is 16.
Step 7		Repeat Steps 2 through 6 for each DS0 group you create.
Step 8	Router(config-voice-port)# exit	Exit voice-port configuration mode.
Step 9	Router# compand type [a-law u-law]	This command converts between analog and digital signals in PCM format. Specifying u-law is the North American mu-law ITU-T PCM encoding standard. Specifying a-law is the European a-law ITU-T PCM encoding standard.
Step 10	Router# cp-tone	This command specifies a regional analog voice interface-related tone, ring, and cadence setting. You can also display the settings for different countries by using the question mark. For details about these commands, see the <i>Cisco IOS Multiservice Applications Command Reference</i> for Cisco IOS Release 12.1.

Verifying Voice Ports


To verify the voice-port configuration, enter the privileged EXEC **show voice port slot/port:ds0-group** command. The following sample output from the command shows explanatory information. Important command output is shown in bold.


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

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
```

Configuring Voice Dial Peers

Follow these steps to set up voice dial peers to support the local and remote stations. Not all possible commands are shown here. To learn more, see *Cisco IOS Multiservice Applications Configuration Guide* and *Cisco IOS Multiservice Applications Command Reference* for Cisco IOS Release 12.1.

	Command	Purpose
Step 1	Router# configure terminal	Enter global configuration mode.
Step 2	Router(config)# dial-peer voice <i>number</i> pots	Enter dial-peer configuration mode and define a local dial peer that will connect to the plain old telephone service (POTS) network. <i>number</i> is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. pots indicates a peer using basic telephone service.
Step 3	Router(config-dialpeer)# destination-pattern <i>string</i> [T]	Configure the dial peer's destination pattern so that the system can reconcile dialed digits with a telephone number. <i>string</i> is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9 and the letters A through D. The plus symbol (+) is not valid. The following special characters can be entered: <ul style="list-style-type: none"> • The star character (*) that appears on standard touch-tone dial pads can be in any dial string but not as a leading character (for example, *650). • The period (.) acts as a wildcard character. • The comma (,) can be used only in prefixes and inserts a one-second pause. When the timer (T) character is included at the end of the destination pattern, the system collects dialed digits as they are entered—until the interdigit timer expires (10 seconds, by default)—or the user dials the termination of end-of-dialing key (default is #).  Note The timer character must be a capital T.

Command	Purpose
Step 4 Router(config-dialpeer)# prefix <i>string</i>	<p>(Optional) Include a dial-out prefix that the system enters automatically instead of people dialing it.</p> <p><i>string</i> is a value from 0 to 9, and you can use a comma (,) to indicate a pause.</p> <hr/> <p> Note There are other digit manipulation commands available to handle such situations as prefixes for special services, ignoring some digits, and dialing in to remote PBXs as though they are local.</p>
Step 5 Router(config-dialpeer)# port <i>slot/port:ds0-group-number</i>	<p>This command associates the dial peer with a specific logical interface.</p> <p><i>slot</i> is the router location where the voice module is installed. Valid entries are from 0 to 3.</p> <p><i>port</i> indicates the voice interface card location. Valid entries are 0 or 1.</p> <p>Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital E1 card.</p>
Step 6 Router(config)# dial-peer voice <i>number</i> voip	<p>Enter dial-peer configuration mode and define a remote VoIP dial peer.</p> <p><i>number</i> is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647.</p> <p>voip indicates a VoIP peer using voice encapsulation on the IP network.</p>

Command	Purpose
<p>Step 7</p> <pre>Router(config-dialpeer)# codec {g711alaw g711ulaw g723ar53 g723ar63 g723r53 g723r63 g726r16 g726r24 g726r32 g728 g729r8 [pre-ietf] g729br8} [bytes]</pre>	<p>The voice-card configuration codec complexity command sets the codec options that are available when you enter this command. See Step 3 of the “Configuring Voice Card and E1 Controller Settings” section on page 5.</p> <p>If you do not set codec complexity, g729r8 with IETF bit-ordering is used.</p> <p>If you set codec complexity to high, the following options are available:</p> <ul style="list-style-type: none"> • g711alaw—G.711 A Law 64,000 bps • g711ulaw—G.711 u Law 64,000 bps • g723ar53—G.723.1 Annex A 5,300 bps • g723ar63—G.723.1 Annex A 6,300 bps • g723r53—G.723.1 5,300 bps • g723r63—G.723.1 6,300 bps • g726r16—G.726 16,000 bps • g726r24—G.726 24,000 bps • g726r32—G.726 32,000 bps • g728—G.728 16,000 bps • g729r8—G.729 8,000 bps (default) • g729br8—G.729 Annex B 8,000 bps <p>If you set codec complexity to medium, the following options are valid:</p> <ul style="list-style-type: none"> • g711alaw—G.711 A Law 64,000 bps • g711ulaw—G.711 u Law 64,000 bps • g726r16—G.726 16,000 bps • g726r24—G.726 24,000 bps • g726r32—G.726 32,000 bps • g729r8—G.729 Annex A 8,000 bps • g729br8—G.729 Annex B with Annex A 8,000 bps <p>The optional <i>bytes</i> parameter sets the number of voice data bytes per frame. Acceptable values are from 10 to 240 in increments of 10 (for example, 10, 20, 30, and so on). Any other value is rounded down (for example, from 236 to 230).</p> <p>If you specify g729r8, then the IETF (Internet Engineering Task Force) bit-ordering is used. For interoperability with a Cisco 2600, 3600, or AS5300 router running a Cisco IOS release prior to Release 12.0(5)T or 12.0(4)XH, you <i>must</i> specify the additional keyword pre-ietf after g729r8.</p>

	Command	Purpose
Step 8	Router(config-dialpeer)# vad	(Optional) This setting is enabled by default. It activates voice activity detection (VAD). VAD allows the system to reduce unnecessary voice transmissions caused by unfiltered background noise.
Step 9	Router(config-dialpeer)# dtmf-relay [cisco-rtp] [h245-signal] [h245-alphanumeric]	<p>(Optional) Dual-tone multifrequency (DTMF) describes the tone that sounds in response to a keypress on a touch-tone telephone. DTMF tones are compressed at one end of a call and decompressed at the other end.</p> <p>If a low-bandwidth codec, such as a G.729 or G.723, is used, the tones can sound distorted. The dtmf-relay command transports DTMF tones generated after call establishment out-of-band by using a method that transmits with greater fidelity than is possible in-band for most low-bandwidth codecs. Without DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated telephone menu systems, such as voicemail and interactive voice response (IVR) systems.</p> <p>A signaling method is supplied only if the remote end supports it, and the options are: Cisco proprietary (cisco-rtp), standard H.323 (h245-alphanumeric), and H.323 standard with signal duration (h245-signal).</p>
Step 10	Router(config-dialpeer)# fax-rate { 2400 4800 7200 9600 12000 14400 disable voice }	(Optional) Specify the transmission speed of a fax to be sent to this dial peer. The disable keyword turns off fax transmission capability, and the voice keyword specifies the highest possible fax speed supported by the voice rate.
Step 11	Router(config-dialpeer)# session target { ipv4:destination-address dns:[\$\$\$. \$d\$. \$e\$. \$u\$.] <i>host-name</i> }	<p>Configure the IP session target for the dial peer.</p> <p>ipv4:destination-address indicates IP address of the dial peer.</p> <p>dns:host-name indicates that the domain name server will resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device.</p> <p>There are also wildcards available for defining domain names with the keyword by using source, destination, and dialed information in the host name. For complete command syntax information, see <i>Cisco IOS Multiservice Applications Command Reference</i> for Cisco IOS Release 12.1.</p>
Step 12	Router(config-dialpeer)# forward-digit [<i>all</i> <i>default</i> <i>extra</i> <i>..</i>]	Configure the interface to forward digits for voice calls.

	Command	Purpose
Step 13	Router(config-dialpeer)# huntstop	(Optional) Disable hunting by the interface for dial peers.
Step 14	Router(config-dialpeer)# exit	Exit interface configuration.

Verifying Voice Dial Peers

Enter the privileged EXEC **show dial-peer voice** command. The following text is sample output from the command for a POTS dial peer. Important command output is shown in bold.

```
Router# show dial-peer voice 1
VoiceEncapPeer1
  tag = 1, dest-pat = \Q+14085551000',
  answer-address = \Q',
  group = 0, Admin state is up, Operation state is down
  Permission is Both,
  type = pots, prefix = \Q',
  session-target = \Q', voice-port =
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Accepted Calls = 0, Refused Calls = 0
  Last Disconnect Cause is "10"
  Last Disconnect Text is ""
  Last Setup Time = 0
```

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

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

Monitoring and Maintaining E1 Digital Packet Voice Configuration

This section presents some useful commands for understanding, maintaining, and troubleshooting your configuration. Table 1 lists the **debug** and **show** commands.

Table 1 *Debug and Show Commands for Maintaining and Troubleshooting Your Configuration*

Command	Purpose
Router# show dialplan number <i>number</i>	Shows which dial-peer is matched by a called number.
Router# show call active voice	Shows statistics for currently active voice calls.
Router# show call active fax	Shows statistics for currently active fax calls.
Router# show call history voice	Shows statistics on previous voice calls.
Router# show call history fax	Shows statistics on previous fax calls.
Router# show voice port	Shows the status of voice ports. See “Verifying Voice Ports” on page 18.
Router# show controller E1 <i>slot/port</i>	Shows the status of the E1 controller. See “Verifying Voice Card and Controller Settings” on page 14.
Router# show isdn status	Shows the status of an individual ISDN line.
Router# debug ccapi inout	Debugs the E1
Router# debug isdn q931	Debugs calls as they are set up and torn down on ISDN network connections (Layer 3) between the local router (user side) and the network.
Router# debug vpm all	Debugs the E1 signaling.
Router# debug vtsp all	Debugs the digits received and sent.
Router# debug voip ccapi inout	Debugs the call setup process.

Show Commands

This section illustrates some of the privileged EXEC show commands that are useful for analyzing your system. Note that important information appears in bold.

The **show dialplan number** command provides information about the dial peer associated with a specified dial-plan number. Notice that the dial peer is operational and that IP Precedence has been configured to the preferred setting of 5.

**Note**

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

```
Router# show dialplan number 75435
Macro Exp.: ##75435
VoiceOverIpPeer70000
  information type = voice,
  tag = 70000, destination-pattern = `##7....',
  answer-address = `', preference=0,
  group = 70000, Admin state is up, Operation state is up,
  incoming called-number = `', connections/maximum = 0/unlimited,
  DTMF Relay = disabled,
  application associated:
  type = voip, session-target = `ipv4:171.68.253.18',
  technology prefix:
  settlement: disabled
  ip precedence = 5, UDP checksum = disabled,
  session-protocol = cisco, req-qos = best-effort,
  acc-qos = best-effort,
  fax-rate = 14400, payload size = 20 bytes
  codec = g729r8, payload size = 20 bytes,
  Expect factor = 10, Icpif = 30,signaling-type = cas,
  VAD = disabled, Poor QOV Trap = disabled,
  Connect Time = 0, Charged Units = 0,
  Successful Calls = 3, Failed Calls = 0,
  Accepted Calls = 3, Refused Calls = 0,
  Last Disconnect Cause is "10 ",
  Last Disconnect Text is "normal call clearing.",
  Last Setup Time = 344813.
Matched: ##75435 Digits: 3
Target: ipv4:171.68.253.18
```

The **show call active voice** command displays information about a current call:

```
Router# show call active voice

GENERIC:
SetupTime=94523746 ms
Index=448
PeerAddress=##73072
PeerSubAddress=
PeerId=70000
PeerIfIndex=37
LogicalIfIndex=0
ConnectTime=94524043
DisconnectTime=94546241
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=6251
TransmitBytes=125020
ReceivePackets=3300
ReceiveBytes=66000
VOIP:
ConnectionId[0x142E62FB 0x5C6705AF 0x0 0x385722B0]
RemoteIPAddress=171.68.235.18
RemoteUDPPort=16580
RoundTripDelay=29 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPlayout=63690
GapFillWithSilence=0 ms
GapFillWithPrediction=180 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=30 ms
ReceiveDelay=40 ms
LostPackets=0 ms
EarlyPackets=1 ms
LatePackets=18 ms
VAD = disabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas
```

The **show call history voice** command shows statistics about previous calls:

```
Router# show call history voice

GENERIC:
SetupTime=94893250 ms
Index=450
PeerAddress=##52258
PeerSubAddress=
PeerId=50000
PeerIfIndex=35
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=94893780
DisconectTime=95015500
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=32258
TransmitBytes=645160
ReceivePackets=20061
ReceiveBytes=401220
VOIP:
ConnectionId[0x142E62FB 0x5C6705B3 0x0 0x388F851C]
RemoteIPAddress=171.68.235.18
RemoteUDPPort=16552
RoundTripDelay=23 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
SessionProtocol=cisco
SessionTarget=ipv4:171.68.235.18
OnTimeRvPayout=398000
GapFillWithSilence=0 ms
GapFillWithPrediction=1440 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPayoutDelay=97 ms
LoWaterPayoutDelay=30 ms
ReceiveDelay=49 ms
LostPackets=1 ms
EarlyPackets=1 ms
LatePackets=132 ms
VAD = disabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas
```

The **show isdn status** command shows the status of ISDN calls:

```
Router# show isdn status

Global ISDN Switchtype = primary-qsig
ISDN Serial1/0/15 interface
***** Network side configuration *****
dsl 0, interface ISDN Switchtype = primary-qsig
**** Master side configuration ****
Layer 1 Status
ACTIVE
Layer 2 Status
TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
Layer 3 Status
24 Active Layer 3 Call(s)
Activated dsl 0 CCBs = 24
CCBcallid=E3C, sapi=0, ces=0, B-chan=1, calltype=VOICE
CCBcallid=E3D, sapi=0, ces=0, B-chan=2, calltype=VOICE
CCBcallid=E3E, sapi=0, ces=0, B-chan=3, calltype=VOICE
CCBcallid=E3F, sapi=0, ces=0, B-chan=4, calltype=VOICE
CCBcallid=E40, sapi=0, ces=0, B-chan=5, calltype=VOICE
CCBcallid=E47, sapi=0, ces=0, B-chan=6, calltype=VOICE
CCBcallid=E48, sapi=0, ces=0, B-chan=7, calltype=VOICE
CCBcallid=E49, sapi=0, ces=0, B-chan=8, calltype=VOICE
CCBcallid=E50, sapi=0, ces=0, B-chan=9, calltype=VOICE
CCBcallid=E51, sapi=0, ces=0, B-chan=10, calltype=VOICE
CCBcallid=E52, sapi=0, ces=0, B-chan=11, calltype=VOICE
CCBcallid=E53, sapi=0, ces=0, B-chan=12, calltype=VOICE
CCBcallid=E54, sapi=0, ces=0, B-chan=13, calltype=VOICE
CCBcallid=E5B, sapi=0, ces=0, B-chan=14, calltype=VOICE
CCBcallid=E5C, sapi=0, ces=0, B-chan=15, calltype=VOICE
CCBcallid=E5D, sapi=0, ces=0, B-chan=17, calltype=VOICE
CCBcallid=E5E, sapi=0, ces=0, B-chan=18, calltype=VOICE
CCBcallid=E5F, sapi=0, ces=0, B-chan=19, calltype=VOICE
CCBcallid=E60, sapi=0, ces=0, B-chan=20, calltype=VOICE
CCBcallid=E61, sapi=0, ces=0, B-chan=21, calltype=VOICE
CCBcallid=E62, sapi=0, ces=0, B-chan=22, calltype=VOICE
CCBcallid=E63, sapi=0, ces=0, B-chan=23, calltype=VOICE
CCBcallid=E64, sapi=0, ces=0, B-chan=24, calltype=VOICE
CCBcallid=E6B, sapi=0, ces=0, B-chan=25, calltype=VOICE
The Free Channel Mask 0xFE000000
Total Allocated ISDN CCBs = 24
```

The **show dial-peer voice summary** command displays information about dial-peers that are active:

```
Router# show dial-peer voice summary

dial-peer hunt 0
TAG TYPE ADMIN OPER PREFIX DEST-PATTERN PREF SESS-TARGET PORT
1 pots up up 3 0 1/015
100 voip down down 1 0 ipv41.2.79.7
200 voip down down 1 0 ipv41.2.79.31
300 vofr up up 1 0 Serial0/0 990
400 voip down down 1 0 ipv45.5.5.2
```

The **show voice call summary** command displays a summary of all dial-peers that are active:

Router# **show voice call summary**

PORT	CODEC	VAD	VTSP	STATE	VPM	STATE
1/015.1	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.2	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.3	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.4	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.5	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.6	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.7	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.8	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.9	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.10	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.11	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.12	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.13	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.14	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.15	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.17	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.18	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.19	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.20	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.21	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.22	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.23	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.24	g729r8	y	S_CONNECT		S_TSP_CONNECT	
1/015.25	g729r8	y	S_CONNECT		S_TSP_CONNECT	

The **show voice dsp** command displays current status of all DSP voice channels:

Router# **show voice dsp**

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

The **show voice trace** command displays a trace of all active voice transitions:

```
Router# show voice trace

1/015 1 State Transitions (state, event) -> (state, event) ...
(S_NULL, E_TSP_INFO_IND) -> (S_SETUP_INDICATED, E_TSP_INFO_IND) ->
(S_SETUP_INDICATED, E_TSP_INFO_IND) -> (S_SETUP_INDICATED, E_CC_PROCEEDING) ->
(S_SETUP_INDICATED, E_CC_ALERT) -> (S_ALERTING, E_CC_BRIDGE) ->
(S_ALERTING, E_CC_CONNECT) -> (S_CONNECT, E_CC_CAPS_IND) ->
(S_CONNECT, E_CC_CAPS_ACK) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_DSP_DTMF_DIGIT_BEGIN) ->
(S_CONNECT, E_DSP_DTMF_DIGIT) -> (S_CONNECT, E_TIMER) ->
```

The **show adapi** command displays information about the call distribution application programming interface (CDAPI):

```
Router# show cdapi

Registered CDAPI Applications/Stacks
=====

Application TSP CDAPI Application Voice
      Application Type(s)  Voice Facility Signaling
      Application Level    Tunnel
      Application Mode      Enbloc

Signaling Stack ISDN
      Interface Se1/015

CDAPI Message Buffers
=====

Used Msg Buffers 0, Free Msg Buffers 6400
Used Raw Buffers 0, Free Raw Buffers 3200
Used Large-Raw Buffers 0, Free Large-Raw Buffers 320
2600-1#
2600-1#
2600-1#s vo call 1/015.1
1/015 1 vtsp level 0 state = S_CONNECT

callid 0x0EDE B01 state S_TSP_CONNECT cldd 1 cllg 3456546347
2600-1# ***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms) -383401219, Rx Delay Est(ms) 61
Rx Delay Lo Water Mark(ms) 61, Rx Delay Hi Water Mark(ms) 90
***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms) 0, Interpolate Conceal(ms) 0
Silence Conceal(ms) 0, Retroact Mem Update(ms) 0
Buf Overflow Discard(ms) 20, Talkspurt Endpoint Detect Err 0
***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts 286, Rx Signal Pkts 0, Rx Comfort Pkts 0
Rx Dur(ms) 24870, Rx Vox Dur(ms) 8510, Rx Fax Dur(ms) 0
Rx Non-seq Pkts 0, Rx Bad Hdr Pkts 0
Rx Early Pkts 0, Rx Late Pkts 0
***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts 826, Tx Sig Pkts 0, Tx Comfort Pkts 0
Tx Dur(ms) 24870, Tx Vox Dur(ms) 24790, Tx Fax Dur(ms) 0
```

```

***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header) 0, Tx Pkt Drops(HPI SAM Overflow) 0
***DSP LEVELS***
TDM Bus Levels(dBm0) Rx -12.5 from PBX/Phone, Tx -13.2 to PBX/Phone
TDM ACOM Levels(dBm0) +0.0, TDM ERL Level(dBm0) +23.5
TDM Bgd Levels(dBm0) -12.1, with activity being voice

```

Debug Commands

This section illustrates some of the EXEC mode debug commands that are useful when analyzing and troubleshooting your system. Note that important information appears in bold.

The **debug isdn q931** command displays information about call setup and teardown of ISDN network connections (Layer 3) between the local router (user side) and the network.

The **debug voip ccapi inout** EXEC command traces the execution path through the call control API, which serves as the interface between the call-session application and the underlying network-specific software.

During the capabilities exchange shown in the command output, both sides agree on what compression to use, and the **debug voip ccapi inout** output helps you determine what each side is negotiating.

This command shows how a call flows through the system. By using this debug level, you can see the call setup and teardown operations performed on both the telephony and network call legs:

```

Router# debug isdn q931
Router# debug voip ccapi inout

001041 ISDN Se1/015 RX <- SETUP pd = 8 callref = 0x1EC5 << the originating call
001041      Sending Complete
001041      Bearer Capability i = 0x8090A3
001041      Channel ID i = 0xA98381
001041      Calling Party Number i = 0x91, '0987654321'
001041      Calling Party SubAddr i = 0x80, 'P123'
001041      Called Party Number i = 0x91, '2312'
001041      Called Party SubAddr i = 0x80, 'P321'
001041      High Layer Compat i = 0x9181
001041      Locking Shift to Codeset 5
001041      Codeset 5 IE 0x31 i = 0x80
001041      Codeset 5 IE 0x32 i = 0x80
0010180388626431 vtsp_tsp_call_setup_ind (sdb=0x81A57008, tdm_info=0x0,
.
.
.
0029107374182399 ISDN BR1/0 TX -> SETUP pd = 8 callref = 0x0001 << terminating call
0029105245511244      Bearer Capability i = 0x8090A3
0029103079215104      Channel ID i = 0xA98381
0029103079215104      Calling Party Number i = 0x91, '0987654321'
0029103079215104      Calling Party SubAddr i = 0x80, 'P123'
0029103079215104      Called Party Number i = 0x91, '312'
0029103079215104      Called Party SubAddr i = 0x80, 'P321'
0029103079215104      Sending Complete
0029103079215104      High Layer Compat i = 0x9181
0029103079215104      Locking Shift to Codeset 5
0029105245510852      Codeset 5 IE 0x31 i = 0x80
0029103079215104      Codeset 5 IE 0x32 i = 0x80
002925 ISDN BR1/0 RX <- RELEASE_COMP pd = 8 callref = 0x8001
002925      Cause i = 0x8096 - Number changed
002925      Facility i = 0x91A4053132333435
002925      User-User i = 0x08, 'USER', 0x20, 'INFORMATION'
.
.

```

```

.
003234359738368      Channel ID i = 0xA98381
003234359738368      Calling Party Number i = 0x91, '0987654321'
003234359738368      Calling Party SubAddr i = 0x80, 'P123'
003234359738368      Called Party Number i = 0x91, '312'
003234359738368      Called Party SubAddr i = 0x80, 'P321'
003234359738368      Sending Complete
003234359738368      High Layer Compat i = 0x9181
003234359738368      Locking Shift to Codeset 5
003236526034116      Codeset 5 IE 0x31 i = 0x80
003234359738368      Codeset 5 IE 0x32 i = 0x80
003209 ISDN BR1/0 RX <- CALL_PROC pd = 8 callref = 0x8003
003209      Channel ID i = 0xA98381
003224 ISDN BR1/0 RX <- PROGRESS pd = 8 callref = 0x8003
003224      Progress Ind i = 0x8181 - Call not end-to-end ISDN, may have
in-band
info

```

Table 2 explains the codec negotiation values that appear—in hexadecimal format—during the capabilities exchange portion of the command output.

Table 2 Codec Negotiation Values in debug voip ccapi inout

Negotiation Value in Decimal	Meaning
1	U-law PCM (g711ulaw)
2	A-law PCM (g711alaw)
3	32k ADPCM (g726r32)
4	24k ADPCM (g726r24)
5	16k ADPCM (g726r16)
6	CS-ACELP - pre-IETF (g729r8 pre-ietf)
7	Medium complexity CS-ACELP - pre-IETF (g729ar8 pre-ietf)
8	CS-ACELP with VAD (g729br8)
9	Medium complexity CS-ACELP with VAD (G.729abr8)
10	16K LD-CELP (g728)
11	G.723.1 High Rate - 6300 bps (g723r63)
12	G.723.1 High Rate with VAD - 6300 bps (g723ar63)
13	G.723.1 Low Rate - 5300 bps (g723r53)
14	G.723.1 Low Rate with VAD - 5300 bps (g723ar53)
19	CS-ACELP - IETF standard (g729r8)
20	Medium complexity CS-ACELP - IETF standard (g729ar8)

Reference Information

The information in this section helps you interpret the output from **debug** and **show** commands.

Table 3 shows Q.931 call disconnection causes. In the examples that follow, the disconnects are caused by normal call clearing.

Table 3 Q.931 Call Disconnection Causes

Call Disconnection Cause Value	Meaning and Number
CC_CAUSE_UANUM = 0x1	/* unassigned number. (1) */
CC_CAUSE_NO_ROUTE = 0x3	/* no route to destination. (3) */
CC_CAUSE_NORM = 0x10	/* normal call clearing. (16) */
CC_CAUSE_BUSY = 0x11	/* user busy. (17) */
CC_CAUSE_NORS = 0x12	/* no user response. (18) */
CC_CAUSE_NOAN = 0x13	/* no user answer. (19) */
CC_CAUSE_REJECT = 0x15	/* call rejected. (21) */
CC_CAUSE_INVALID_NUMBER = 0x1C	/* invalid number. (28) */
CC_CAUSE_UNSP = 0x1F	/* normal, unspecified. (31) */
CC_CAUSE_NO_CIRCUIT = 0x22	/* no circuit. (34) */
CC_CAUSE_NO_REQ_CIRCUIT = 0x2C	/* no requested circuit. (44) */
CC_CAUSE_NO_RESOURCE = 0x2F	/* no resource. (47) */
CC_CAUSE_NOSV = 0x3F	/* service or option not available, Unspecified. (63) */
CC_CAUSE_UNINITIALIZED = 0	/* un-initialized (0) */
CC_CAUSE_UANUM = 1	/* unassigned num */
CC_CAUSE_NO_ROUTE_TO_TRANSIT_NETWORK = 2	
CC_CAUSE_NO_ROUTE = 3	/* no rt to dest */
CC_CAUSE_SEND_INFO_TONE = 4	
CC_CAUSE_MISDIALLED_TRUNK_PREFIX = 5	
CC_CAUSE_CHANNEL_UNACCEPTABLE = 6	
CC_CAUSE_CALL_AWARDED = 7	
CC_CAUSE_PREEMPTION = 8	
CC_CAUSE_PREEMPTION_RESERVED = 9	
CC_CAUSE_NORM = 16	
CC_CAUSE_BUSY = 17	/* user busy */
CC_CAUSE_NORS = 18	/* no user response*/
CC_CAUSE_NOAN = 19	/* no user answer. */
CC_CAUSE_SUBSCRIBER_ABSENT = 20	
CC_CAUSE_REJECT = 21	/* call rejected. */
CC_CAUSE_NUMBER_CHANGED = 22	
CC_CAUSE_NON_SELECTED_USER_CLEARING = 26	
CC_CAUSE_DESTINATION_OUT_OF_ORDER = 27	
CC_CAUSE_INVALID_NUMBER = 28	

Table 3 Q.931 Call Disconnection Causes (continued)

Call Disconnection Cause Value	Meaning and Number
CC_CAUSE_FACILITY_REJECTED = 29	
CC_CAUSE_RESPONSE_TO_STATUS_ENQUIRY = 30	
CC_CAUSE_UNSP = 31	/* unspecified. */
CC_CAUSE_NO_CIRCUIT = 34	/* no circuit. */
CC_CAUSE_REQUESTED_VPCI_VCI_NOT_AVAILABLE = 35	
CC_CAUSE_VPCI_VCI_ASSIGNMENT_FAILURE = 36	
CC_CAUSE_CELL_RATE_NOT_AVAILABLE = 37	
CC_CAUSE_NETWORK_OUT_OF_ORDER = 38	
CC_CAUSE_PERM_FRAME_MODE_OUT_OF_SERVICE = 39	
CC_CAUSE_PERM_FRAME_MODE_OPERATIONAL = 40	
CC_CAUSE_TEMPORARY_FAILURE = 41	
CC_CAUSE_SWITCH_CONGESTION = 42	
CC_CAUSE_ACCESS_INFO_DISCARDED = 43	
CC_CAUSE_NO_REQ_CIRCUIT = 44	
CC_CAUSE_NO_VPCI_VCI_AVAILABLE = 45	
CC_CAUSE_PRECEDENCE_CALL_BLOCKED = 46	
CC_CAUSE_NO_RESOURCE = 47	/* no resource. */
CC_CAUSE_QOS_UNAVAILABLE = 49	
CC_CAUSE_FACILITY_NOT_SUBSCRIBED = 50	
CC_CAUSE_CUG_OUTGOING_CALLS_BARRED = 53	
CC_CAUSE_CUG_INCOMING_CALLS_BARRED = 55	
CC_CAUSE_BEARER_CAPABILITY_NOT_AUTHORIZED = 57	
CC_CAUSE_BEARER_CAPABILITY_NOT_AVAILABLE = 58	
CC_CAUSE_INCONSISTENCY_IN_INFO_AND_CLASS = 62	
CC_CAUSE_NOSV = 63	/* service or option * not available * unspecified. */
CC_CAUSE_BEARER_CAPABILITY_NOT_IMPLEMENTED = 65	
CC_CAUSE_CHAN_TYPE_NOT_IMPLEMENTED = 66	
CC_CAUSE_FACILITY_NOT_IMPLEMENTED = 69	
CC_CAUSE_RESTRICTED_DIGITAL_INFO_BC_ONLY = 70	
CC_CAUSE_SERVICE_NOT_IMPLEMENTED = 79	
CC_CAUSE_INVALID_CALL_REF_VALUE = 81	
CC_CAUSE_CHANNEL_DOES_NOT_EXIST = 82	
CC_CAUSE_CALL_EXISTS_CALL_ID_IN_USE = 83	
CC_CAUSE_CALL_ID_IN_USE = 84	
CC_CAUSE_NO_CALL_SUSPENDED = 85	

Table 3 Q.931 Call Disconnection Causes (continued)

Call Disconnection Cause Value	Meaning and Number
CC_CAUSE_CALL_CLEARED = 86	
CC_CAUSE_USER_NOT_IN_CUG = 87	
CC_CAUSE_INCOMPATIBLE_DESTINATION = 88	
CC_CAUSE_NON_EXISTENT_CUG = 90	
CC_CAUSE_INVALID_TRANSIT_NETWORK = 91	
CC_CAUSE_AAL_PARAMS_NOT_SUPPORTED = 93	
CC_CAUSE_INVALID_MESSAGE = 95	
CC_CAUSE_MANDATORY_IE_MISSING = 96	
CC_CAUSE_MESSAGE_TYPE_NOT_IMPLEMENTED = 97	
CC_CAUSE_MESSAGE_TYPE_NOT_COMPATIBLE = 98	
CC_CAUSE_IE_NOT_IMPLEMENTED = 99	
CC_CAUSE_INVALID_IE_CONTENTS = 100	
CC_CAUSE_MESSAGE_IN_INCOMP_CALL_STATE = 101	
CC_CAUSE_RECOVERY_ON_TIMER_EXPIRY = 102	
CC_CAUSE_NON_IMPLEMENTED_PARAM_PASSED_ON = 103	
CC_CAUSE_UNRECOGNIZED_PARAM_MSG_DISCARDED = 110	
CC_CAUSE_PROTOCOL_ERROR = 111	
CC_CAUSE_INTERWORKING = 127	

Table 4 Tone Types and Their Meanings

Tone Type	Meaning
CC_TONE_RINGBACK	0x1—Ring Tone
CC_TONE_FAX	0x2—Fax Tone
CC_TONE_BUSY	0x4—Busy Tone
CC_TONE_DIALTONE	0x8—Dial Tone
CC_TONE_OOS	0x10—Out-of-Service Tone
CC_TONE_ADDR_ACK	0x20—Address Acknowledgement Tone
CC_TONE_DISCONNECT	0x40—Disconnect Tone
CC_TONE_OFF_HOOK_NOTICE	0x80—Tone indicating the telephone was left off hook
CC_TONE_OFF_HOOK_ALERT	0x100 /* A more urgent version of CC_TONE_OFF_HOOK_NOTICE*/
CC_TONE_CUSTOM	0x200—Custom Tone - used when specifying a custom tone
CC_TONE_NULL	0x0—Null Tone

These are codec capabilities bits that can appear in command output:

- CC_CAP_CODEEC_G711U 0x1
- CC_CAP_CODEEC_G711A 0x2
- CC_CAP_CODEEC_G723ar63 0x2000
- CC_CAP_CODEEC_G723ar53 0x4000
- CC_CAP_CODEEC_G723r63 0x100
- CC_CAP_CODEEC_G723r53 0x200
- CC_CAP_CODEEC_G726r16 0x10
- CC_CAP_CODEEC_G729 0x4
- CC_CAP_CODEEC_G729 0x8000
- CC_CAP_CODEEC_G729a 0x8
- CC_CAP_CODEEC_G729b 0x800
- CC_CAP_CODEEC_G729ab 0x1000

These are fax capabilities bits that can appear in command output. The numbers following “FAX_” refer to the fax speed (for example, “144” means 14,400 bps):

- CC_CAP_FAX_NONE 0x1
- CC_CAP_FAX_VOICE 0x2
- CC_CAP_FAX_144 0x4
- CC_CAP_FAX_96 0x8
- CC_CAP_FAX_72 0x10
- CC_CAP_FAX_48 0x20
- CC_CAP_FAX_24 0x40
- CC_CAP_FAX_120 0x80

These are the VAD on and off capability bits:

- CC_CAP_VAD_OFF 0x1
- CC_CAP_VAD_ON 0x2

Configuration Example

This section displays the configuration example of a router running a digital E1 packet voice trunk network module interface:

```
Router#show running-config
```

```
Building configuration...
```

```
Current configuration
!
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2600-1
!
!
!
memory-size iomem 10
voice-card 1
!
ip subnet-zero
no ip domain-lookup
!
frame-relay switching
isdn switch-type primary-qsig
isdn voice-call-failure 0
voice hunt user-busy
!
!
!
controller E1 1/0
  pri-group timeslots 1-31
!
controller E1 1/1
  shutdown
!
!
!
interface Ethernet0/0
  ip address 1.2.79.1 255.255.0.0
  no ip directed-broadcast
  no cdp enable
!
interface Serial0/0
  no ip address
  no ip directed-broadcast
  encapsulation frame-relay
  no ip mroute-cache
  load-interval 30
  clockrate 800000
  frame-relay traffic-shaping
  frame-relay class voice-vc
  frame-relay interface-dlci 990
  vofr data 4 call-control 5
  frame-relay intf-type dce
!
interface Ethernet0/1
  no ip address
  no ip directed-broadcast
```

```

shutdown
no cdp enable
!
interface Serial0/1
 ip address 5.5.5.1 255.0.0.0
 no ip directed-broadcast
 encapsulation frame-relay
 no ip mroute-cache
 clockrate 800000
 frame-relay traffic-shaping
 frame-relay class voice-data
 frame-relay interface-dlci 991
 frame-relay ip rtp header-compression
 frame-relay intf-type dce
!
interface Serial1/015
 no ip address
 no ip directed-broadcast
 ip mroute-cache
 no logging event link-status
 isdn switch-type primary-qsig
 isdn overlap-receiving
 isdn protocol-emulate network
 isdn incoming-voice voice
 no isdn T309-enable
 isdn bchan-number-order ascending
 fair-queue 64 256 0
 no cdp enable
!
router rip
 network 172.28.0.0
!
router igrp 1
 redistribute connected
 network 1.0.0.0
!
ip default-gateway 1.2.0.1
ip classless
ip route 223.255.254.254 255.255.255.255 1.2.0.1
no ip http server
!
!
map-class frame-relay voice-vc
 no frame-relay adaptive-shaping
 frame-relay cir 512000
 frame-relay bc 512000
 frame-relay fair-queue
 frame-relay voice bandwidth 512000
 frame-relay fragment 100
!
map-class frame-relay voice-data
 no frame-relay adaptive-shaping
 frame-relay cir 512000
 frame-relay bc 1000
 frame-relay fair-queue
 frame-relay fragment 200
 frame-relay ip rtp priority 2000 16383 500
dialer-list 1 protocol ip permit
dialer-list 1 protocol ipx permit
no cdp run
!
voice-port 1/015
 compand-type a-law
!

```

```
dial-peer voice 1 pots
 destination-pattern 3
 direct-inward-dial
 port 1/015
 forward-digits all
!
dial-peer voice 100 voip
 shutdown
 destination-pattern 1
 session target ipv41.2.79.7
!
dial-peer voice 200 voip
 shutdown
 destination-pattern 1
 session target ipv41.2.79.31
!
dial-peer voice 300 vofr
 destination-pattern 1
 session target Serial0/0 990
!
dial-peer voice 400 voip
 shutdown
 destination-pattern 1
 session target ipv45.5.5.2
!
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 password ard
 login
!
end
```

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command references.

- **pri-group**

pri-group

To specify a ISDN Primary Rate interface (PRI) on a channelized T1 or E1 controller, enter the **pri-group** controller configuration command. To remove the ISDN-PRI configuration, enter the **no** form of this command.

pri-group timeslots *timeslot-range*

no pri-group

Syntax Description

<i>timeslot-range</i>	Specifies a single range of values. For T1, the allowable range is from 1 to 23. For E1, the allowable values are from 1 to 15.
-----------------------	---

Defaults

There is no ISDN-PRI group configured.

Command Modes

Controller configuration

Command History

Release	Modification
11.0	This command was introduced.
12.0(2)T	The command was introduced for the Cisco MC3810 multiservice access concentrator.
12.0(7)XK	The command was introduced for the Cisco 2600 and 3600 series.
12.1(2)T	The command was implemented in Cisco IOS Release 12.1(2)T.

Usage Guidelines

The **pri-group** command applies to the configuration of Voice over Frame Relay and Voice over ATM on the Cisco MC3810 multiservice access concentrator and the Cisco 2600 and 3600 series routers.

Before you enter the **pri-group** command, you must specify an ISDN-PRI switch type and an E1 or T1 controller.



Note

Only one PRI group can be configured on a controller.

Examples

The following example configures ISDN-PRI on all timeslots of controller E1 on a Cisco 2600 series router:

```
Router(config-controller)# pri-group timeslots 1-7, 16

controller E1 4/0
!
controller E1 4/1
  pri-group timeslots 1-7,16
!
```

Related Commands

Command	Description
isdn switch-type	To configure the Cisco 2600 series router PRI interface to support QSIG signaling, enter this command.

Glossary

AAL—ATM Adaptation Layer. Service-dependent sublayer of the data link layer. The AAL accepts data from different applications and presents it to the ATM layer in the form of 48-byte ATM payload segments. AALs consist of two sublayers: convergence sublayer (CS) and segmentation and reassembly (SAR). AALs differ on the basis of the source-destination timing used, whether they use constant bit rate (CBR) or variable bit rate (VBR), and whether they are used for connection-oriented or connectionless mode data transfer. At present, the four types of AAL recommended by the ITU-T are AAL1, AAL2, AAL3/4, and AAL5.

AAL1—ATM Adaptation Layer 1. One of four AALs recommended by the ITU-T. AAL1 is used for connection-oriented, delay-sensitive services requiring constant bit rates, such as uncompressed video and other isochronous traffic.

AMI—alternate mark inversion. Line-code type used on T1 and E1 circuits. In AMI, zeros are represented by 01 during each bit cell, and ones are represented by 11 or 00, alternately, during each bit cell. AMI requires that the sending device maintain ones density. Ones density is not maintained independent of the data stream. Sometimes called *binary coded alternate mark inversion*.

ATM—Asynchronous Transfer Mode. International standard for cell relay in which multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells. Fixed-length cells allow cell processing to occur in hardware, thereby reducing transit delays. ATM is designed to take advantage of high-speed transmission media such as E3, SONET, and T3.

B8ZS—binary 8-zero substitution. Line-code type, used on T1 and E1 circuits, in which a special code is substituted whenever 8 consecutive zeros are sent over the link. This code is then interpreted at the remote end of the connection. This technique guarantees ones density independent of the data stream.

CAS—channel-associated signaling. Trunk signaling (for example, in a T1 line) in which control signals, such as those for synchronizing and bounding frames, are carried in the same channel along with voice and data signals.

CBR—constant bit rate. QoS class defined by the ATM Forum for ATM networks. CBR is used for connections that depend on precise clocking to ensure undistorted delivery.

CCS—common channel signaling. Trunk signaling (for example, using Primary Rate Interface) in which a control channel carries signaling for separate voice and data channels.

CES—circuit emulation service. Enables users to multiplex or concentrate multiple circuit emulation streams for voice and video with packet data on a single high-speed ATM link without a separate ATM access multiplexer.

CO—central office. Local telephone company office to which all local loops in a given area connect and in which circuit switching of subscriber lines occurs.

codec—Coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog.

DTMF—Dual-tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch tone).

Drop and Insert—(also called TDM Cross-Connect) Allows DS0 channels from one T1 or E1 facility to be digitally cross-connected to DS0 channels on another T1 or E1. Using this method, channel traffic is sent between a PBX and CO PSTN switch or other telephony device, so that some PBX channels are directed for long-distance service through the PSTN while the router compresses others for interoffice VoIP calls. In addition, Drop and Insert can cross-connect a telephony switch (from the CO or PSTN) to a channel bank for external analog connectivity.

DSP—digital signal processor, same as PVDM.

- E1**—European digital carrier facility used for transmitting data through the telephone hierarchy. The transmission rate for E1 is 2.048 megabits per second (Mbps).
- E&M**—rEceive and transMit, or Ear and Mouth. Type of signaling originally developed for analog two-state voltage telephony using the ear and mouth leads; in digital telephony, uses two bits.
- ESF**—Extended Superframe. Framing type used on T1 circuits that consists of 24 frames of 192 bits each, with the 193rd bit providing timing and other functions. ESF is an enhanced version of SF format.
- FXO**—Foreign Exchange Office. A voice interface emulating a PBX trunk line to a switch or telephone equipment to a PBX extension interface.
- FXS**—Foreign Exchange Station. A voice interface for connecting telephone equipment, emulates the extension interface of a PBX or the subscriber interface for a switch.
- IETF**—Internet Engineering Task Force
- ISDN**—Integrated Services Digital Network. Communication protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other source traffic.
- IVR**—interactive voice response. Term used to describe systems that provide information in the form of recorded messages over telephone lines in response to user input in the form of spoken words or more commonly DTMF signaling. Examples include banks that allow you to check your balance from any telephone and automated stock quote systems.
- packet**—Logical grouping of information that includes a header containing control information and (usually) user data. Packets are most often used to refer to network layer units of data.
- POTS**—plain old telephone service
- PVDM**—packet voice data module
- PSTN**—Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.
- QoS**—quality of service. Measure of performance for a transmission system that reflects its transmission quality and service availability.
- SF**—Super Frame. Common framing type used on T1 circuits. SF consists of 12 frames of 192 bits each, with the 193rd bit providing error checking and other functions. SF is superseded by ESF, but is still widely used. Also called D4 framing.
- SNMP**—Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security.
- T1**—Digital WAN carrier facility. T1 transmits DS 1-formatted data at 1.544 Mbps through the telephone switching network, using alternate mark inversion or B8ZS coding.
- T1 trunk**—Digital WAN carrier facility. See T1.
- TDM**—time-division multiplexing.
- Trunk**—Physical and logical connection between two switches across which network traffic travels. A backbone is composed of a number of trunks.
- UNI**—User-Network Interface. ATM Forum specification that defines an interoperability standard for the interface between ATM-based products (a router or an ATM switch) located in a private network and the ATM switches located within the public carrier networks. Also used to describe similar connections in Frame Relay networks.
- VAD**—voice activity detection.

