Digital J1 Voice Interface Card

Feature History

Release	Modification
12.2(8)T	This feature was introduced on the Cisco 2600 and Cisco 3600 series.

This document describes the Digital J1 Voice Interface Card feature in Cisco IOS Release 12.2(8)T. It includes the following sections:

Feature Overview

The J1 interface card provides the proper interface for directly connecting Cisco multiservice access routers to Private Branch Exchanges (PBXs) throughout Japan that use a J1 interface (2.048 Mbps TDM interface). This interface card supports 30 voice channels per port.

It provides the software and hardware features required to connect to over 80percent of the PBXs within Japan that use digital interfaces. This new J1 voice interface card (VIC) provides a TTC JJ-20.11 compliant interface between high-density voice network modules (NM-HDV) and a Japanese PBX.

The digital J1 card provides a single-port line interface in a VIC form factor. It is specifically designed to conform to the TTC JJ-20.10-12 standards that define the interface between a PBX and time-division multiplexer (TDM).

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Figure 1 shows the earlier solution offered to customers in Japan. A J1/T1 adapter box installed between the PBX and router provides the translation between J1 using coded mark inversion (CMI) line coding at a bit rate of 2.048 Mbps and a T1 line using either alternate mark inversion (AMI) or B8ZS line coding at a bit rate of 1.544 Mbps. Note that with this solution, only 24 channels are supported, instead of the full 30 channels of the J1 interface.

Figure 1 Solution without J1 Interface Card

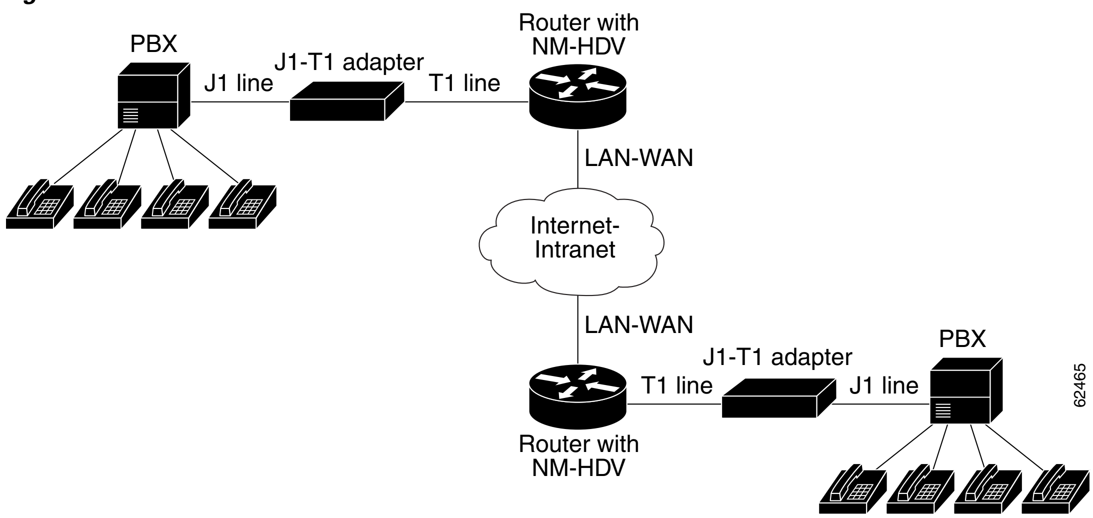
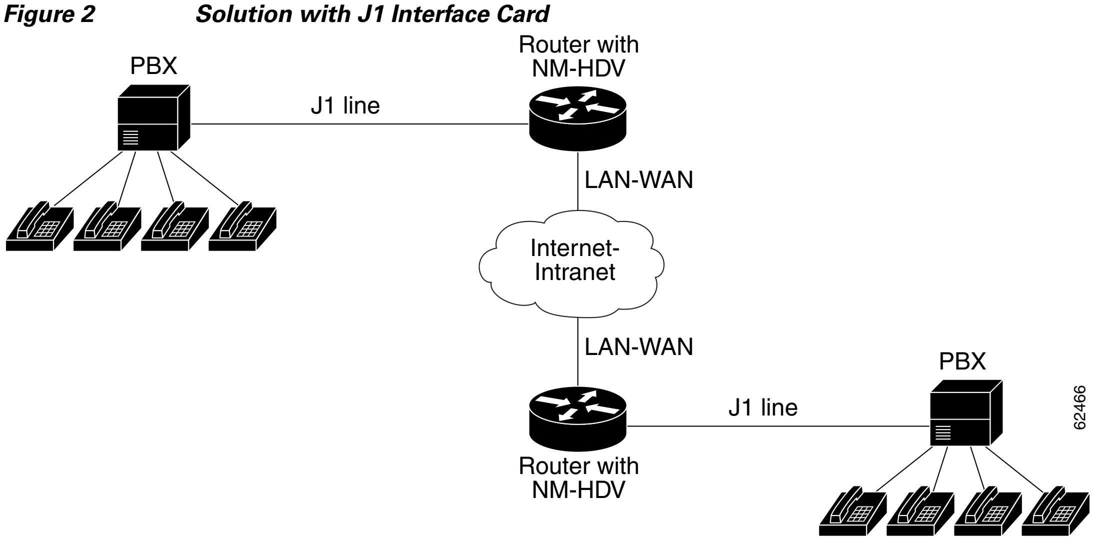


Figure 2 shows the solution using the J1 interface card. The interface is now between J1 and the VIC's time division multiplex access (TDMA) bus. Note that now all 30 channels of the J1 interface are supported.

Figure 2 Solution with J1 Interface Card



Benefits

- Support for Media Gateway Control Protocol (MGCP), H.248, H.323 (versions 1, 2, and 3), Session Initiation Protocol (SIP) and Cisco Call Manager (with Cisco IP phones) in association with VoIP, VoFR, and VoATM.
- Provides Alarm Indication Signal (AIS) alarm signaling per TTC JJ-20.11.
- Delivers the same performance as the existing 30 channel E1 NM-HDV.
- Allows one to enable and disable individual DS0's or channels.

Restrictions

- Voice only applications.
- Separate clock output not supported.
- Alarm relay output not supported.
- Per channel loopback not supported.
- Voice ports on the J1 interface cannot be configured using network management software. They can only be configured manually.

Related Documents

- *Installing and Configuring 1-Port J1 Voice Interface Cards*
- *Cisco IOS Security Configuration Guide*, Release 12.2
- *Cisco IOS Configuration Fundamentals Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that support specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to quickly determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions at http://www.cisco.com/register.

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Supported Standards, MIBs, and RFCs

Standards

- General specification TTC JJ-20.10
- TTC interface specification TTC JJ-20.11
- TTC Signaling specification TTC JJ-20.12 (E&M wink start, wink immediate, and DTMF only).

MIBs

None

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

RFCs

None

Prerequisites

- Cisco IOS Release 12.2(8)T or later release.

Configuration Tasks

See the following sections for configuration tasks for this feature. Each task in the list is identified as either required or optional:

- Configuring the J1 Controller (required)
- Configuring Channel-Associated Signaling (optional)
- Configuring the Clock Source (optional)
- Configuring Loopback (optional)
- Configuring Transparent Common Channel Signaling for a Clear-Channel Codec (optional)
- Verifying Configuration (optional)

Configuring the J1 Controller

Use the following procedure to configure the J1 controller.

	Command	Purpose
Step 1	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 2	`Router(config)#` **`controller j1`** *`slot/port`*	Selects the J1 controller to configure. *slot/port*—Backplane slot number and port number on the controller.

Configuring Channel-Associated Signaling

Configure the DS0 groups on the J1 controller for voice applications. The J1 controller supports the E&M wink start and E&M immediate channel associated signaling (CAS) protocols for the voice ports.

The following parameters have default values for the J1 interface:

- The companding type is ulaw.
- The CP tone is set to JP.

	Command	Purpose
Step 1	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 2	`Router(config)#` **`controller j1`** *`slot/port`*	Selects the J1 controller to configure and enters controller configuration mode. This example configures a J1 controller in slot 1 and port 0.

	Command	Purpose
Step 3	Router(config-controller)# **ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type** *signaling-type* **service** *service-type*	Command defines the j1 1/0 for use by compressed voice calls and the signaling method the router uses to connect to the PBX. **Note** This step shows the basic syntax and signaling types available with the **ds0-group** command. For the complete syntax, see the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2. The keywords and arguments are as follows: • *ds0-group-no*—Specifies the DS0 group number. • **timeslots** *timeslot-list*—Specifies the DS0 time slot range of values from 1 to 31 for J1 interfaces. Time slot 16 is reserved for signaling. • **type** *signaling-type*—(optional) Specifies the signaling type to be applied to the selected group. The options are as follows: – **e&m-delay-dial**—Specifies that the originating endpoint sends an off-hook signal and then and waits for an off-hook signal followed by an on-hook signal from the destination. – **e&m-immediate-start**—Specifies no specific off-hook and on-hook signaling. – **e&m-wink-start—**Specifies that the originating endpoint sends an off-hook signal and waits for a wink signal from the destination. – **none**—Specifies null signaling for external call control.
Step 4	`Router(config-controller)# exit`	Exits to global configuration mode. Return to Step 2 if your router has more than one J1 controller that you need to configure.

Configuring the Clock Source

Use the following procedure to configure the clock source for a J1 controller.

	Command	Purpose
Step 1	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 2	`Router(config)#` **`controller j1 1/0`**	Selects the J1 controller to configure and enters controller configuration mode. This example configures a J1 controller in slot 1 and port 0.
Step 3	`Router(config-controller)#` **`clock source`** `{`**`line`** `|`**`internal`**`}`	Specifies the clock source, either internal or line. • **line**—The controller recovers external clock from the line and provides the recovered clock to the internal (system) clock generator. The line value is the default clock source. • **internal**—The controller synchronizes itself to the internal (system) clock.
Step 4	`Router(config-controller)#` **`exit`**	Exits to global configuration mode. Return to Step 2 if your router has more than one J1 controller that you need to configure.

Configuring Loopback

Use the following procedure to configure the loopback for testing a J1 controller.

	Command	Purpose
Step 1	`Router#` **`configure terminal`**	Enters global configuration mode. You have entered global configuration mode when the prompt changes to `Router(config)#`.
Step 2	`Router(config)#` **`controller j1 1/0`**	Selects the J1 controller to configure and enters controller configuration mode. This example configures a J1 controller in slot 1 and port 0.
Step 3	`Router(config-controller)#` **`loopback`** `{`**`local`** `|` **`line`** `|` **`isolation`**`}`	Sets the loopback method for testing the J1 interface. • **local**—Places the interface into local loopback mode. • **line**—Places the interface into external loopback mode at the line level • **isolation**—Both local and line loopback.
Step 4	`Router(config-controller)#` **`exit`**	Exits to global configuration mode.

Configuring Transparent Common Channel Signaling for a Clear-Channel Codec

Use the following procedure to configure transparent common channel signaling (T-CCS).

	Command	Purpose
Step 1	`Router#` **`configure terminal`**	Enters global configuration mode.
Step 2	`Router(config)#` **`controller j1`** *`slot/port`*	Selects the J1 controller to configure. *slot/port*—Backplane slot number and port number on the controller.
Step 3	Router(config-controller)# **ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type** *ext-sig*	This command defines the j1 0 for use by compressed voice calls and the signaling method the router uses to connect to the PBX. **Note** This step shows the basic syntax and signaling types available with the **ds0-group** command. For the complete syntax, see the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2. The keywords and arguments are as follows: • *ds0-group-no*—Specifies the DS0 group number. • **timeslots** *timeslot-list*—Specifies the DS0 time slot range of values from 1 to 31 for J1 interfaces. Time slot 16 is reserved for signaling. • **type** *ext-sig*—(optional) The signaling method selection for type depends on the connection that you are making:The external signaling interface specifies that the signaling traffic comes from an outside source.
Step 4	`Router(config-controller)#` **`no shutdown`**	Activates the controller.
Step 5	`Router(config-controller)#` **`exit`**	Exits controller configuration mode.
Step 6	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enters dial-peer configuration mode and define a local dial peer that connects to the plain old telephone service (POTS) network. *The value of number* is one or more digits identifying the dial peer. Valid entries are from 1 through 2147483647. The **pots** keyword indicates a peer using a basic telephone service.

	Command	Purpose
Step 7	Router(config-dialpeer)# **destination-pattern** *string* [**T**]	Configures the dial peer's destination pattern so that the system can reconcile dialed digits with a telephone number. *The value of string* is a series of digits that specify the E.164 or private dialing plan phone number. Valid entries are the digits 0 through 9 and the letters A through D. The plus symbol (+) is not valid. You can enter the following special characters: • 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. • Use the comma (,) only in prefixes, the comma 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.
Step 8	Router(config-dialpeer)# **port** *slot/port:ds0-group-no*	Associates the dial peer with a specific logical interface. *The value of slot* is the router location where the voice module is installed. Valid entries are from 0 through 3. *The value of port* indicates the voice interface card location. Valid entries are 0 or 1. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s.
Step 9	Router(config-dialpeer)# **exit**	Exit dial-peer configuration mode to complete the POTS dial-peer configuration.
Step 10	Router(config)# **dial-peer voice** *number* **voip**	Enters dial-peer configuration mode and defines a remote VoIP dial peer. *The value of number* is one or more digits identifying the dial peer. Valid entries are from 1 through 2147483647. The **voip** keyword indicates a VoIP peer using voice encapsulation on the IP network.
Step 11	Router(config-dialpeer)# **codec clear-channel**	Set codec option to **clear-channel** to use the clear channel codec.

	Command	Purpose
Step 12	`Router(config-dialpeer)#` **`vad`**	(optional) This setting is enabled by default. It activates voice activity detection (VAD) which allows the system to reduce unnecessary voice transmissions caused by unfiltered background noise.
Step 13	`Router(config-dialpeer)#` **`destination-pattern`** *`string`* `[`**`T`**`]`	Configures the dial peer's destination pattern so that the system can reconcile dialed digits with a telephone number. *The value of string* is a series of digits that specify the E.164 or private dialing plan phone number. Valid entries are the digits 0 through 9 and the letters A through D. The plus symbol (+) is not valid. You can enter the following special characters: • 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. • Use the comma (,) only in prefixes, the comma 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.
Step 14	`Router(config-dialpeer)#` **`session target`** `{`**`ipv4:`***`destination-address`* `|` **`dns:`**`[`**`$s$.`** `|` **`$d$.`** `|` **`$e$.`** `|` **`$u$.`**`]` *`host-name`*`}`	Configure the IP session target for the dial peer. **The ipv4:***destination-address* parameter indicates IP address of the dial peer. **The dns:***host-name parameter* 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. 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 *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
Step 15	`Router(config-dialpeer)#` **`exit`**	Exit dial peer configuration mode for the VoIP dial-peer configuration.

Verifying Configuration

To verify that J1 controller is configured correctly, enter the **show running-config** privileged EXEC command to display the command settings for the router, as shown in the "Configuration Examples" section.

Troubleshooting Tips

Diagnostics and Fault Isolation

Three digital loopback modes are possible for diagnostics and fault isolation.

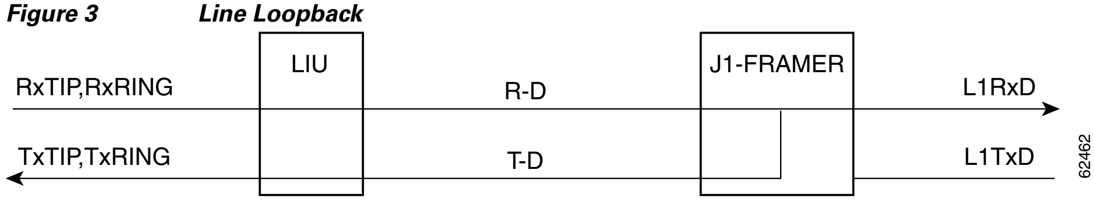
Loopback Modes

The J1 Framer has three loopback modes that are initiated through software control; line loopback, local loopback, and isolation loopback. Line loopback loops the received signal (R-D) from the PBX to the transmit going back to the PBX. Local loopback loops the transmitted signal (T-D) from the host to the receive going back to the host. Isolation loopback routes PBX and TDM generated traffic back to their respective sources. (Tx=transmit interface; Rx=receive interface;
Tip / Ring leads carry audio between the signaling unit and the trunking circuit).

- Line Loopback: To place the controller into line loopback, use the following command in controller configuration mode.

Command	Purpose
loopback line	Line loopback loops the receiver inputs to the transmitter outputs. The receive path is not affected by the activation of this loopback.

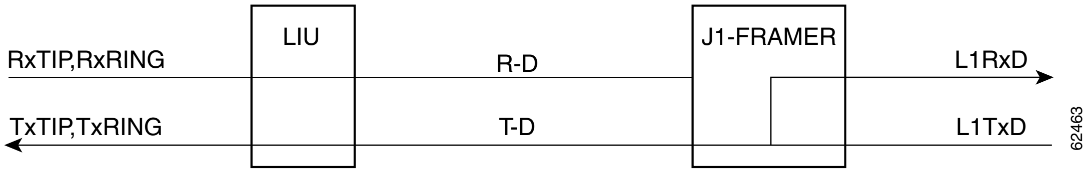
Figure 3 Line Loopback



- Local Loopback: To place the controller into local loopback, use the following command in controller configuration mode. Use the **no** form of this command to turn off the loopback. The command should only be used for testing purposes.

Command	Purpose
loopback local	Local loopback loops the transmit line encoder outputs to the receive line encoder inputs. The transmit path is not affected by the activation of this loopback.

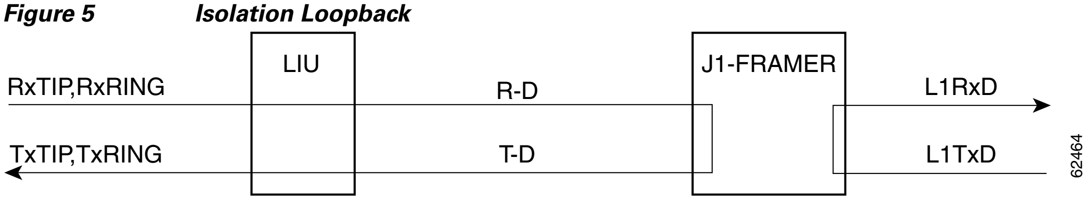
Figure 4 Local Loopback



- Isolation Loopback: To place the controller into line loopback, use the following command in controller configuration mode.

Command	Purpose
loopback isolation	Both line and local loopback are turned on.

Figure 5 Isolation Loopback



Monitoring and Maintaining the J1 Controller

To monitor and maintain the J1 controller use the following privileged EXEC command.

Command	Purpose
`Router#` **`show controllers j1`** *slot/port*	Displays statistics for the J1 link.
`Router#` **`show dial-peer voice`**	Displays configuration information for dial peers.

Configuration Examples

The following displays the screen output using the **show running-config** command. Then it is broken down into specific examples:

- Controller (J1) Example
- Channel-Associated Signaling Example
- Clock Source Example
- Loopback Example
- Transparent Common Channel Signaling for a Clear-Channel Codec Example

```
Router#show run
Building configuration...

Current configuration :2023 bytes
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname kmm-3660-1
!
boot system tftp /tftpboot/kmenon/c3660-is-mz 223.255.254.254
enable password lab
!
voice-card 1
!
voice-card 3
!
voice-card 4
!
ip subnet-zero
!
!
!
!
!
voice service pots
!
!
!
!
!
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
controller J1 1/0
 clock source line
!
controller E1 3/0
!
controller E1 3/1
!
controller T1 4/0
 framing esf
 linecode b8zs
```

```
 channel-group 0 timeslots 24
!
controller T1 4/1
 framing esf
 linecode b8zs
 channel-group 0 timeslots 24
!
!
!
!
interface Multilink1
 ip address 30.30.30.1 255.255.255.0
 keepalive 1
 no cdp enable
 ppp multilink
 no ppp multilink fragmentation
 multilink-group 1
!
interface FastEthernet0/0
 ip address 1.7.29.1 255.255.0.0
 no ip mroute-cache
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 1.8.0.1 255.255.0.0
 no ip mroute-cache
 duplex auto
 speed auto
!
interface Serial4/0:0
 no ip address
 encapsulation ppp
 no fair-queue
 no cdp enable
 ppp multilink
 multilink-group 1
!
interface Serial4/1:0
 no ip address
 encapsulation ppp
 no fair-queue
 no cdp enable
 ppp multilink
 multilink-group 1
!
ip default-gateway 1.7.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.1.1.1
ip route 1.9.0.1 255.255.255.255 30.30.30.2
ip route 223.255.254.254 255.255.255.255 1.7.0.1
no ip http server
ip pim bidir-enable
!
!
!
snmp-server engineID local 00000009020000044D0EF520
snmp-server packetsize 4096
!
call rsvp-sync
!
no mgcp timer receive-rtcp
!
mgcp profile default
```

```
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
 destination-pattern 88
!
dial-peer voice 20 voip
 destination-pattern 3050
 session target ipv4:1.8.0.2
 codec clear-channel
!
dial-peer voice 77 pots
 destination-pattern 77
!
dial-peer voice 100 voip
 incoming called-number 100
 destination-pattern 100
 session target ipv4:1.8.0.2
 no vad
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
!
end
```

Controller (J1) Example

The following example shows the Cisco IOS interface card in slot 4, port 0 of a Cisco 3660 configured as a J1 controller:

```
controller J1 4/0
```

Channel-Associated Signaling Example

The following example shows the DS0 groups on the J1 controller.

```
controller J1 4/0
 clock source line
 ds0-group 1 timeslots 1-15,17-31 type e&m-wink-start
```

Clock Source Example

The following example shows the J1 controller clock source is configured to line, where the controller recovers external clock from the line and provides the recovered clock to the internal (system) clock generator.

```
controller J1 3/0
 clock source line
```

Loopback Example

The following example shows the loopback method for testing the J1 controller is set at the line level.

```
controller J1 3/0
 clock source line
 loopback line
```

Transparent Common Channel Signaling for a Clear-Channel Codec Example

The following example shows the codec option set to clear-channel.

```
dial-peer voice 20 voip
 destination-pattern 3050
 session target ipv4:1.8.0.2
 codec clear-channel
```

Command Reference

The following new and modified commands are pertinent to this feature. To see the command pages for these commands and other commands used with this feature, go to the *Cisco IOS Master Commands List*, Release 12.4, at http://www.cisco.com/univercd/cc/td/doc/product/software/ios124/124mindx/124index.htm.

New Commands

- **show controllers j1**

Modified Commands

- **clock source (controller j1)**
- **controller (j1)**
- **ds0-group (controller j1)**
- **loopback (controller j1)**
- **microcode reload controller (j1)**

Glossary

AIS—alarm indication s22ignal. An all-ones signal transmitted in lieu of the normal signal to maintain transmission continuity and to indicate to the receiving terminal that there is a transmission fault that is located either at, or upstream from, the transmitting terminal.

AMI—alternate mark inversion. Line-code type used on T1 and E1 circuits.

CAS—channel associated signaling. The transmission of signaling information within the voice channel. CAS signaling often is referred to as robbed-bit signaling because user bandwidth is being robbed by the network for other purposes.

CCS—common channel signaling. Signaling system used in telephone networks that separates signaling information from user data. A specified channel is exclusively designated to carry signaling information for all other channels in the system.

CMI—coded mark inversion. ITU-T line coding technique specified for STS-3c transmissions.

codec—In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.

E&M—recEive and transMit (or ear and mouth). Trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco's analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M is also available on E1 and T1 digital interfaces.

FPGA—field programmable gate array.

J1 framer—A functional block within the VIC FPGA which works in tandem with the LIUs to perform the J1 framing, monitoring and loopback functions.

LIU—line interface unit.

MGCP—Media Gateway Control Protocol. A merging of the IPDC and SGCP protocols.

OOF—Out Of Frame. A designation for a condition defined as either the network or the DTE equipment sensing an error in framing bits.

NM-HDV—High-Density Voice network modules.

SIP—session initiation protocol. Protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.

TDM—time division multiplex. Technique in which information from multiple channels can be allocated bandwidth on a single wire based on preassigned time slots. Bandwidth is allocated to each channel regardless of whether the station has data to transmit.

TDMA—time division multiplex access. Type of multiplexing where two or more channels of information are transmitted over the same link by allocating a different time interval ("slot" or "slice") for the transmission of each channel, that is, the channels take turns to use the link. Some kind of periodic synchronizing signal or distinguishing identifier usually is required so that the receiver can tell which channel is which.

VIC—voice interface card. Connects the system to either the PSTN or to a PBX.

VoATM—Voice over ATM. Voice over ATM enables a router to carry voice traffic (for example, telephone calls and faxes) over an ATM network. When sending voice traffic over ATM, the voice traffic is encapsulated using a special AAL5 encapsulation for multiplexed voice.

VoFR—Voice over Frame Relay. Voice over Frame Relay enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network using FRF.12 encapsulation.

VoIP—Voice over IP. The ability to carry normal telephony-style voice over an IP-based internet with POTS-like functionality, reliability, and voice quality.