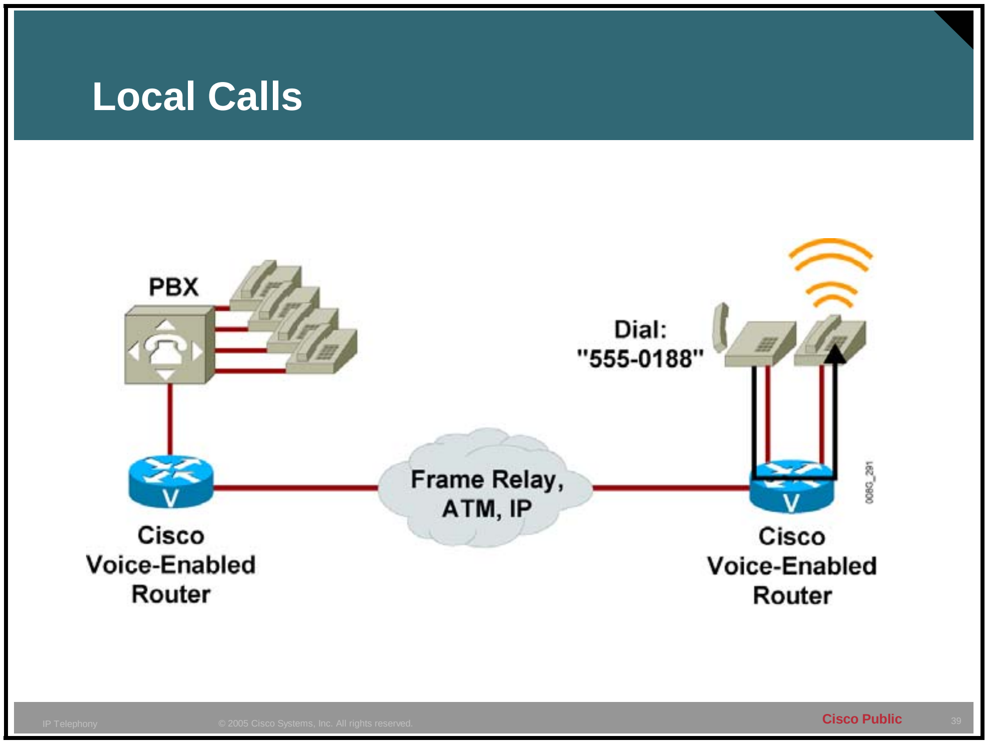
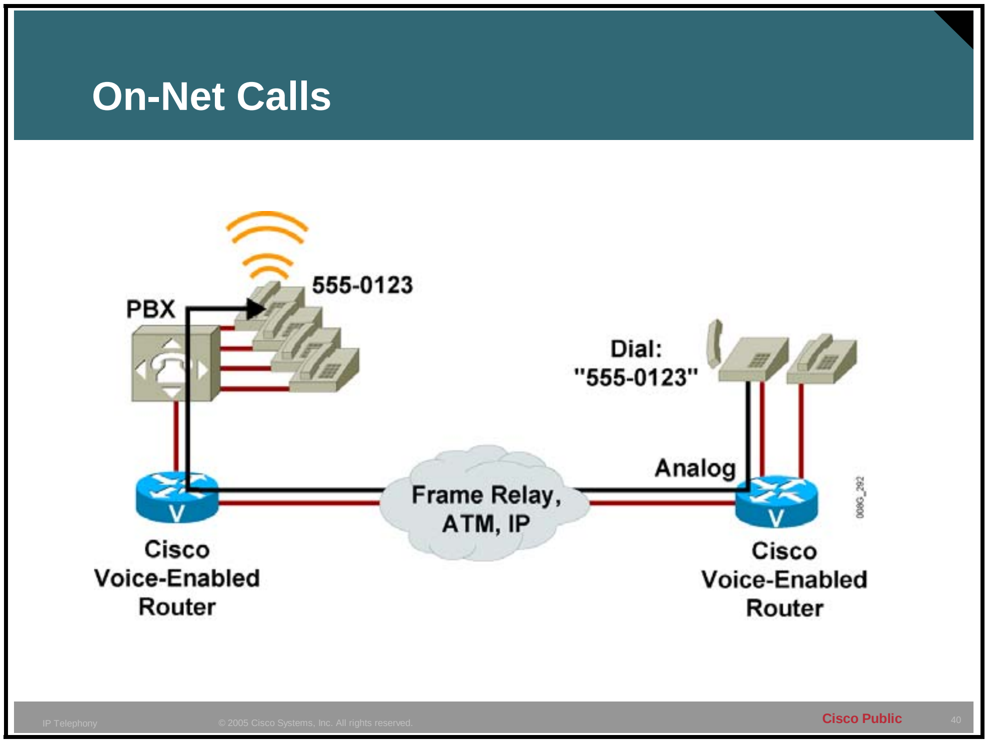
Configuring Voice Ports

Voice Port Applications

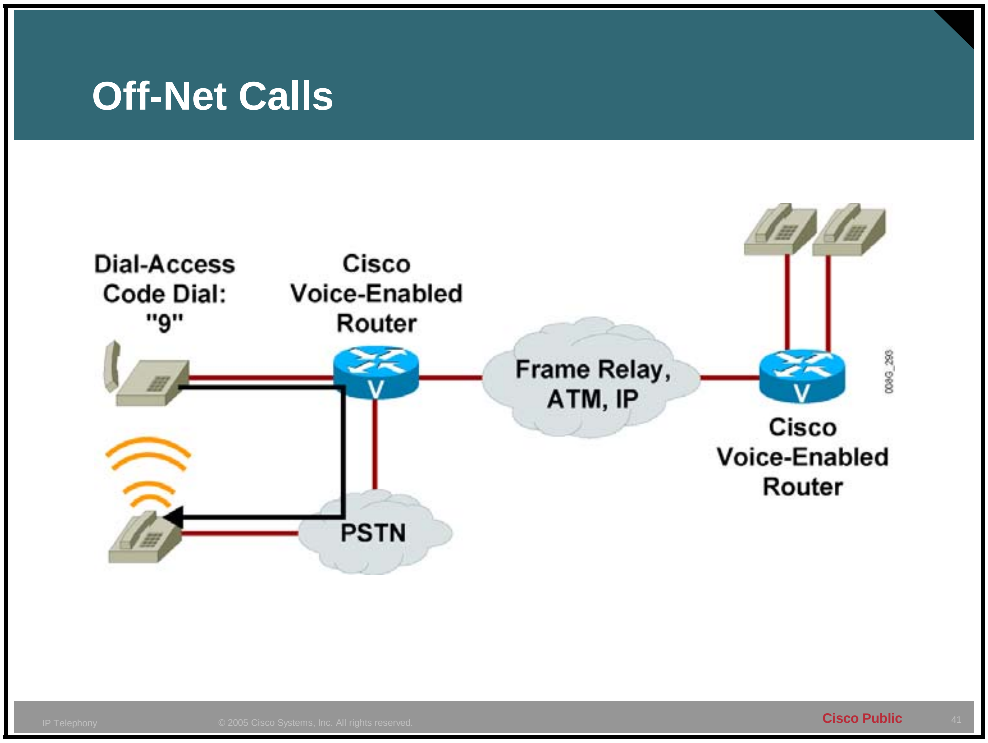
Different types of applications require specific types of ports. In many instances, the type of port is dependent on the voice device that is connected to the network. This topic identifies the different types of voice port applications within the network.



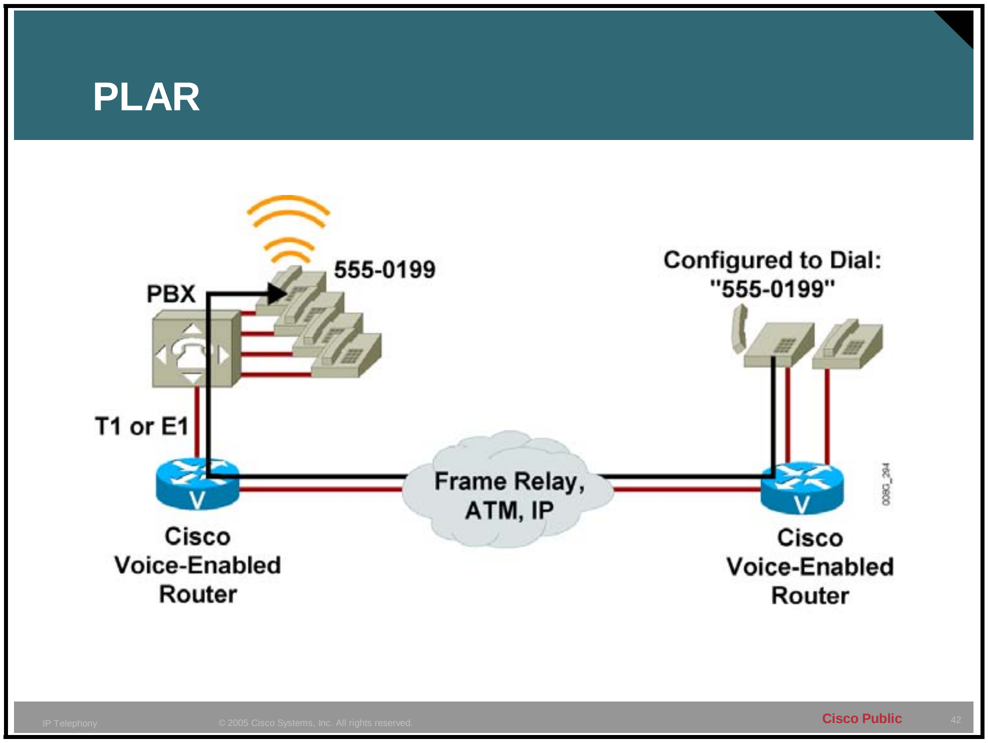
Local calls occur between two telephones connected to one Cisco voice-enabled router. This type of call is handled entirely by the router and does not travel over an external network. Both telephones are directly connected to Foreign Exchange Station (FXS) ports on the router.



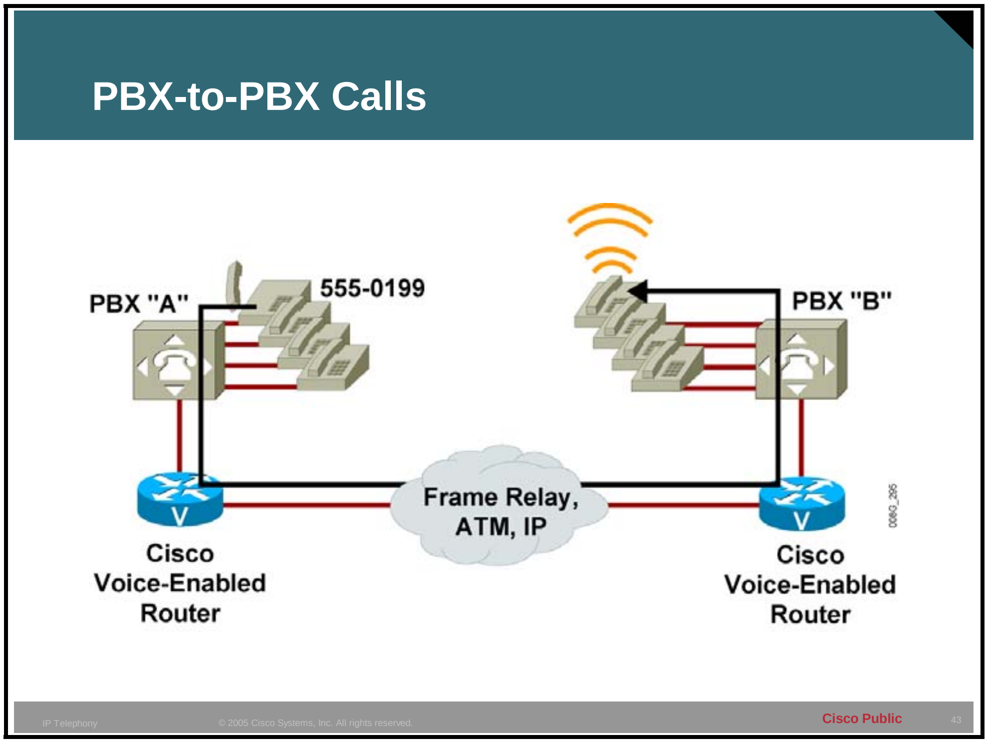
On-net calls occur between two telephones on the same data network. The calls can be routed through one or more Cisco voice-enabled routers, but the calls remain on the same data network. The edge telephones attach to the network through direct connections and FXS ports, or through a PBX, which typically connects to the network via a T1 connection. IP Phones that connect to the network via switches place on-net calls either independently or through Cisco CallManager. The connection across the data network can be a LAN connection, as in a campus environment, or a WAN connection, as in an enterprise environment.



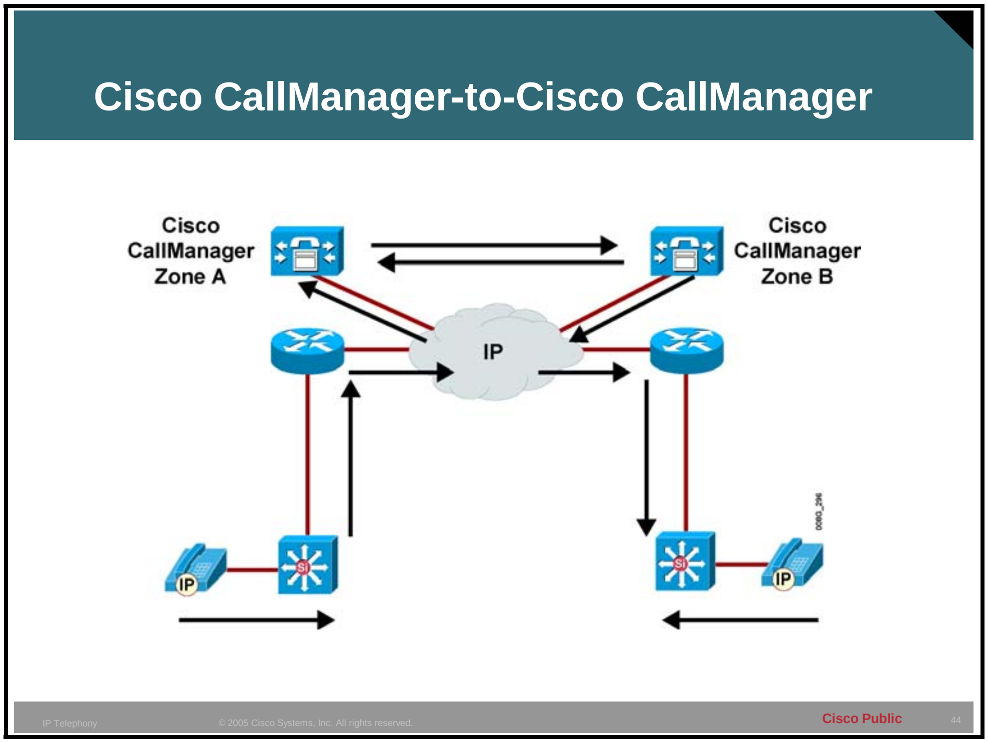
Off-net calls occur when, to gain access to the public switched telephone network (PSTN), the user dials an access code, such as “9,” from a telephone that is directly connected to a Cisco voice-enabled router or PBX. The connection to the PSTN is a single analog connection via a Foreign Exchange Office (FXO) port or a digital T1 or E1 connection.



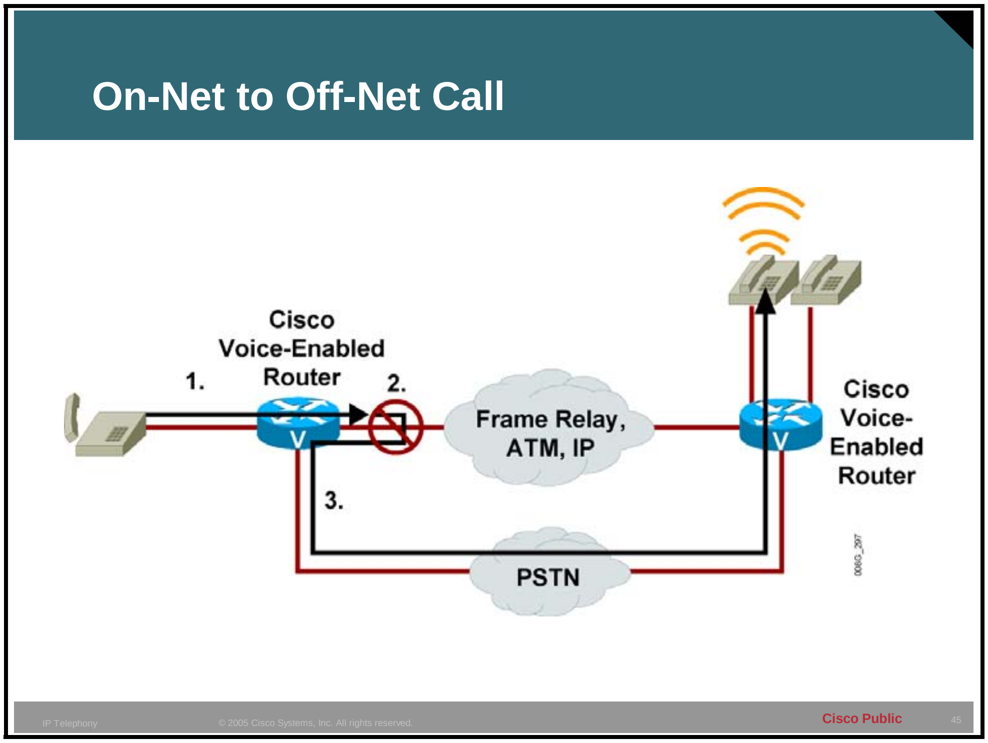
Private line, automatic ringdown (PLAR) calls automatically connect a telephone to a second telephone when the first telephone goes off hook. When this connection occurs, the user does not get a dial tone because the voice-enabled port that the telephone is connected to is preconfigured with a specific number to dial. A PLAR connection can work between any types of signaling, including recEive and transMit (ear and mouth [E&M]), FXO, FXS, or any combination of analog and digital interfaces.



A PBX-to-PBX call originates at a PBX at one site and terminates at a PBX at another site while using the network as the transport between the two locations. Many business environments connect sites with private tie trunks. When migrating to a converged voice and data network, this same tie-trunk connection can be emulated across the IP network. Modern PBX connections to the network are typically digital T1 or E1 with channel associated signaling (CAS) or PRI signaling, although PBX connections can also be analog.



As part of an overall migration strategy, a business may replace PBXs with a Cisco CallManager infrastructure. This infrastructure includes IP telephones that plug directly into the IP network. Cisco CallManager performs the same call-routing functions formerly provided by the PBX. When an IP Phone uses Cisco CallManager to place a call, Cisco CallManager—based on its configuration—assesses if the call is destined for another IP Phone under its control, or if the call must be routed through a remote Cisco CallManager for call completion. Although the call stays on the IP network, it may be sent between zones. Every Cisco CallManager is part of a zone. A zone is a collection of devices that are under a common administration, usually a Cisco CallManager or gatekeeper.



When planning a resilient call-routing strategy, it may be necessary to reroute calls through a secondary path should the primary path fail. On-net to off-net calls originate on an internal network and are routed to an external network, usually to the PSTN. On-net to off-net call-switching functionality may be necessary when a network link is down, or if a network becomes overloaded and unable to handle all calls presented.

Example: Voice Port Applications

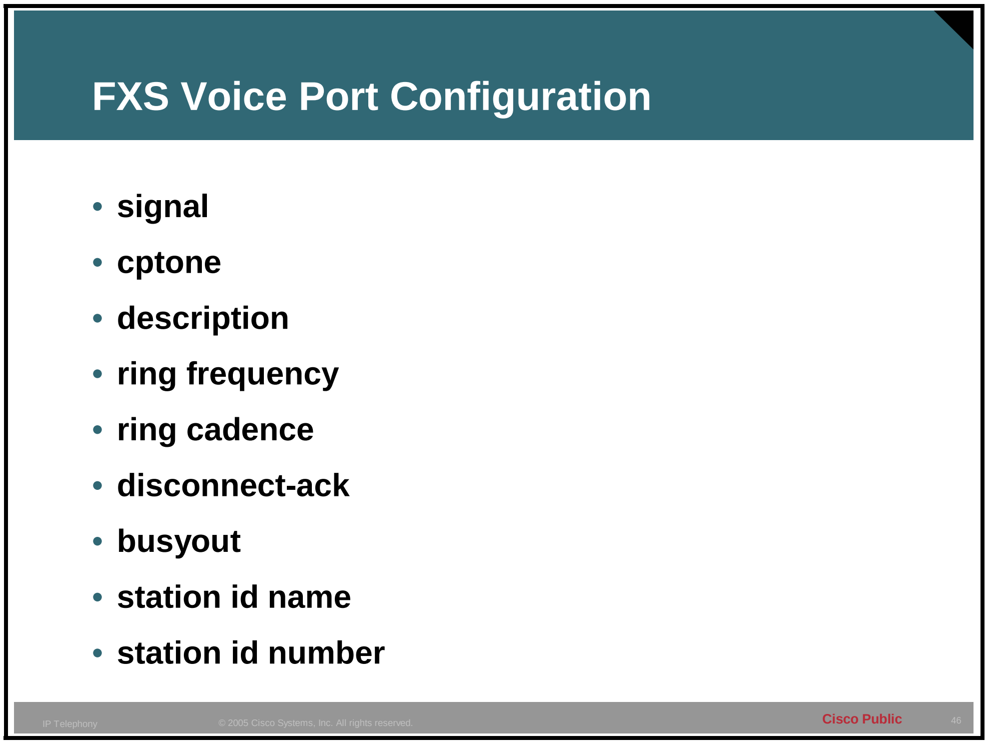
The table lists application examples for each type of call.

Voice Port Call Types

Type of Call	Example
Local calls	One staff member calls another staff member at the same office. The call is switched between two ports on the same voice-enabled router.
On-net calls	One staff member calls another staff member at a remote office. The call is sent from the local voice-enabled router, across the IP network, and terminated on the remote office voice-enabled router.
Off-net calls	A staff member calls a client who is located in the same city. The call is sent from the local voice-enabled router that acts as a gateway to the PSTN. The call is then sent to the PSTN for call termination.
PLAR calls	A client picks up a customer service telephone located in the lobby of the office and is automatically connected to a customer service representative without dialing any digits. The call is automatically dialed, based on the PLAR configuration of the voice port. In this case, as soon as the handset goes off hook, the voice-enabled router generates the prespecified digits to place the call.
PBX-to-PBX calls	One staff member calls another staff member at a remote office. The call is sent from the local PBX, through a voice-enabled router, across the IP network, through the remote voice-enabled router, and terminated on the remote office PBX.
Cisco CallManager-to-Cisco CallManager calls	One staff member calls another staff member at a remote office using IP Phones. The call setup is handled by the Cisco CallManagers at both locations. After the call is set up, the IP Phones generate IP packets carrying voice between sites.
On-net to off-net calls	One staff member calls another staff member at a remote office while the IP network is congested. When the originating voice-enabled router determines that it cannot terminate the call across the IP network, it sends the call to the PSTN with the appropriate dialed digits to terminate the call at the remote office via the PSTN network.

FXS Ports

FXS ports connect analog edge devices. This topic identifies the parameters that are configurable on the FXS port.



In North America, the FXS port connection functions with default settings most of the time. The same cannot be said for other countries and continents. Remember, FXS ports look like switches to the edge devices that are connected to them. Therefore, the configuration of the FXS port should emulate the switch configuration of the local PSTN.

Example: Configuring FXS Ports

For example, consider the scenario of an international company with offices in the United States and England. The PSTN of each country provides signaling that is standard for that country. In the United States, the PSTN provides a dial tone that is different from the tone in England. When the telephone rings to signal an incoming call, the ring is different in the United States. Another instance when the default configuration might be changed is when the connection is a trunk to a PBX or key system. In that case, the FXS port must be configured to match the settings of that device.

Configuration Parameters

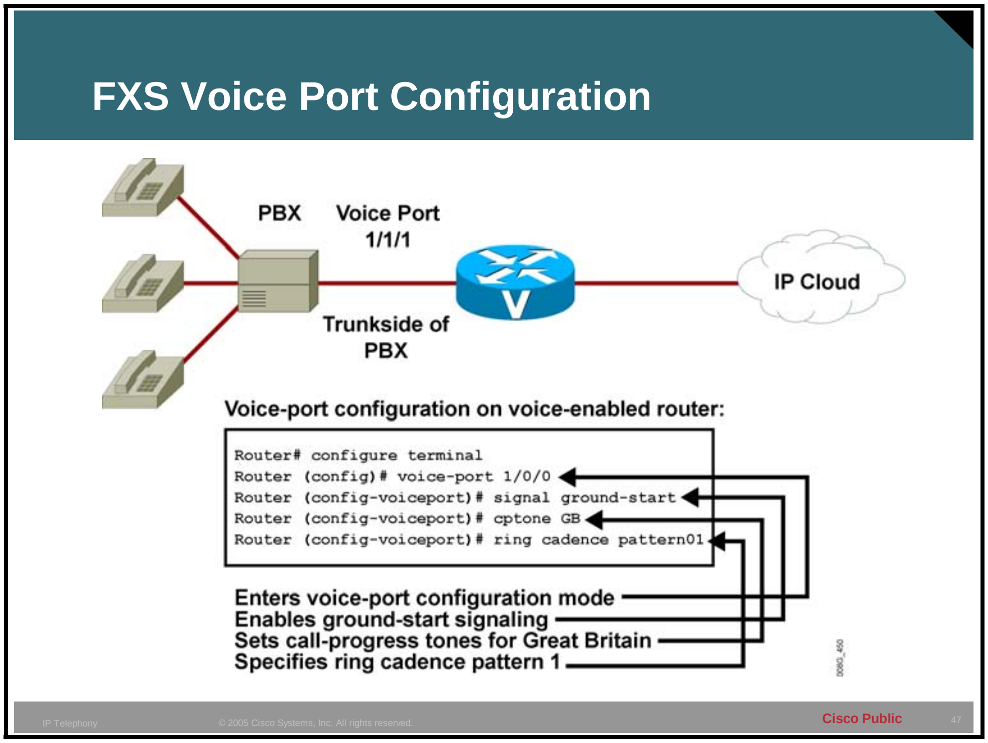
FXS port configuration allows you to set parameters based on the requirements of the connection if default settings need to be altered or the parameters need to be set for fine-tuning. You can set the following configuration parameters:

- **signal:** Sets the signaling type for the FXS port. In most cases, the default signaling of loop start works well. If the connected device is a PBX or a key system, the preferred signaling is ground start. Modern PBXs and key systems do not normally use FXS ports as connections to the network, but older systems may still have these interfaces. When connecting the FXS port to a PBX or key system, you must check the configuration of the voice system and set the FXS port to match the system setting.

- **cptone:** Configures the appropriate call-progress tone for the local region. The call-progress tone setting determines the dial tone, busy tone, and ringback tone to the originating party.
- **description:** Configures a description for the voice port. You must use the description setting to describe the voice port in **show** command output. It is always useful to provide some information about the usage of a port. The description could specify the type of equipment that is connected to the FXS port.
- **ring frequency:** Configures a specific ring frequency (in Hz) for an FXS voice port. You must select the ring frequency that matches the connected equipment. If set incorrectly, the attached telephone might not ring or might buzz. In addition, the ring frequency is usually country-dependent, and you should take into account the appropriate ring frequency for your area before you configure this command.
- **ring cadence:** Configures the ring cadence for an FXS port. The ring cadence defines how ringing voltage is sent to signal a call. The normal ring cadence in North America is 2 seconds of ringing followed by 4 seconds of silence. In England, normal ring cadence is a short ring followed by a longer ring. When configured, the **cptone** setting automatically sets the ring cadence to match that country. You can manually set the ring cadence if you want to override the default country value. You may have to shut down and reactivate the voice port before the configured value takes effect.
- **disconnect-ack:** Configures an FXS voice port to remove line power if the equipment on an FXS loop-start trunk disconnects first. This removal of line power is not something the user hears, but instead is a method for electrical devices to signal that one side has ended the call.
- **busyout:** Configures the ability to busy out an analog port.
- **station id name:** Provides the station name associated with the voice port. This parameter is passed as a calling name to the remote end if the call is originated from this voice port. If no Caller ID is received on an FXO voice port, this parameter will be used as the calling name. Maximum string length is limited to 15.
- **station id number:** Provides the station number that is to be used as the calling number associated with the voice port. This parameter is optional and, if provided, will be used as the calling number if the call is originated from this voice port. If not specified, the calling number will be used from a reverse dial-peer search. If no Caller ID is received on an FXO voice port, this parameter will be used as the calling number. Maximum string length is 15.

Example: FXS Port Configuration

The example shows how the British office is configured to enable ground-start signaling on a Cisco 2600 or 3600 series router on FXS voice port 1/0/0. The call-progress tones are set for Great Britain, and the ring cadence is set for pattern 1.


FXS Voice Port Configuration
PBX
Voice Port 1/1/1
Trunkside of PBX
IP Cloud
Voice-port configuration on voice-enabled router:
Router# configure terminal
Router (config)# voice-port 1/0/0
Router (config-voiceport)# signal ground-start
Router (config-voiceport)# cptone GB
Router (config-voiceport)# ring cadence pattern01
Enters voice-port configuration mode
Enables ground-start signaling
Sets call-progress tones for Great Britain
Specifies ring cadence pattern 1
IP Telephony
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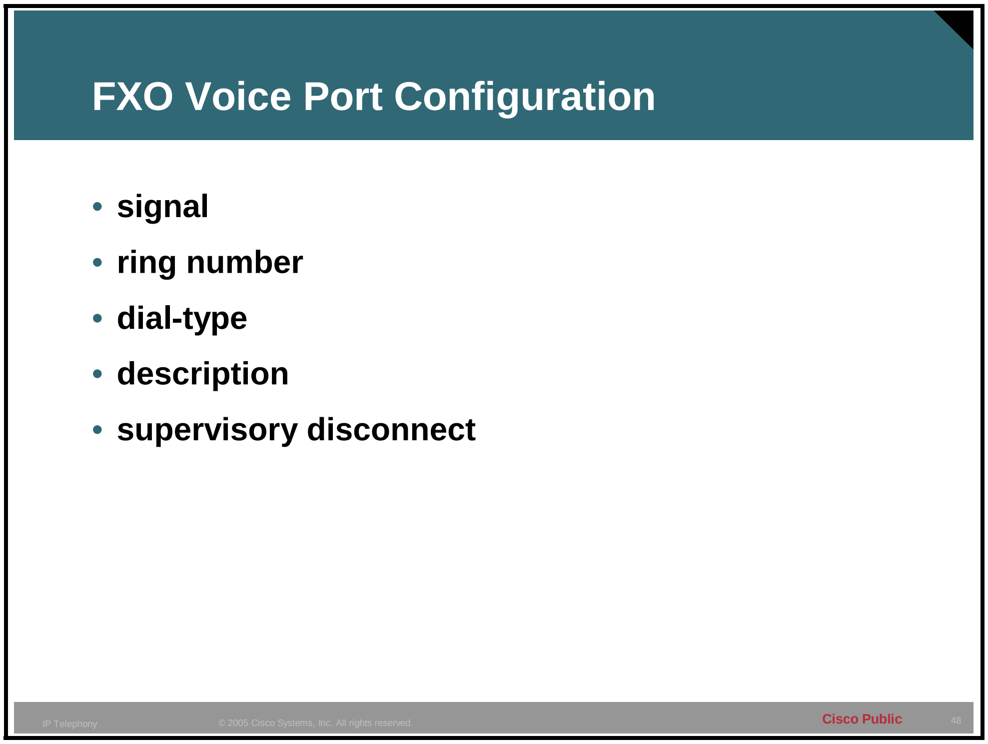
FXO Ports

FXO ports act like telephones and connect to central office (CO) switches or to a station port on a PBX. This topic identifies the configuration parameters that are specific to FXO ports.



Configuration Parameters

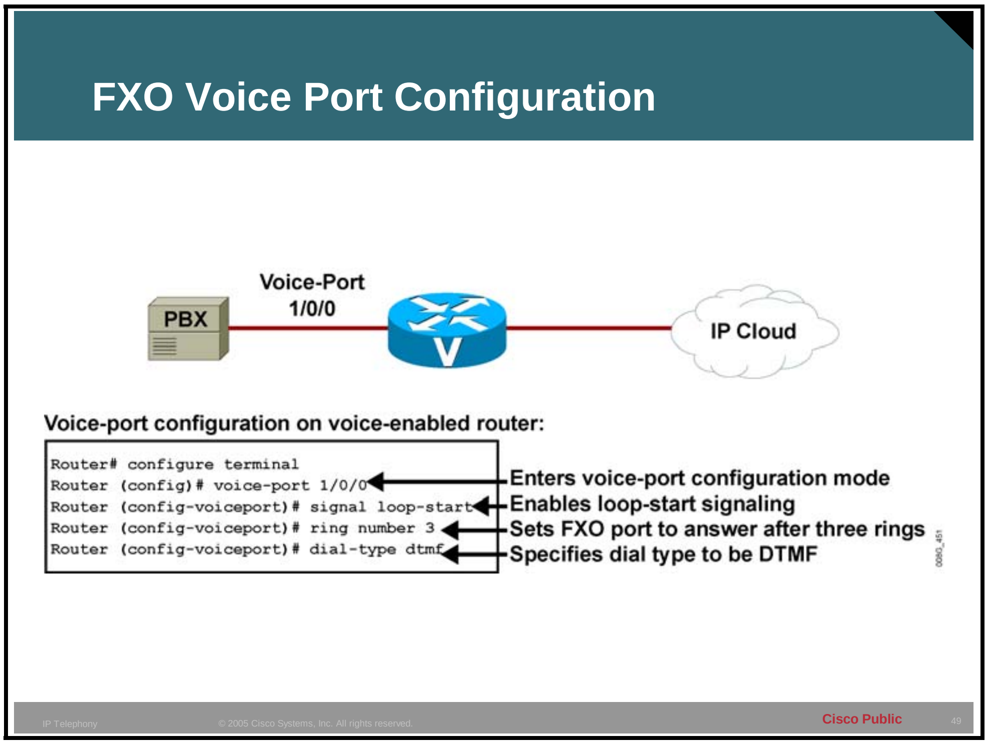
In most instances, the FXO port connection functions with default settings. FXO port configuration allows you to set parameters based on the requirements of the connection where default settings need to be altered or parameters set for fine-tuning. You can set the following configuration parameters:

- **signal:** Sets the signaling type for the FXO port. If the FXO port is connected to the PSTN, the default settings are adequate. If the FXO port is connected to a PBX, the signal setting must match the PBX.
- **ring number:** Configures the number of rings before an FXO port answers a call. This is useful when you have other equipment available on the line to answer incoming calls. The FXO port answers if the equipment that is online does not answer the incoming call within the configured number of rings.
- **dial-type:** Configures the appropriate dial type for outbound dialing. Older PBXs or key sets may not support dual-tone multifrequency (DTMF) dialing. If you are connecting an FXO port to this type of device, you may need to set the dial type for pulse dialing.
- **description:** Configures a description for the voice port. Use the description setting to describe the voice port in **show** command output.

- **supervisory disconnect:** Configures supervisory disconnect signaling on the FXO port. Supervisory disconnect signaling is a power denial from the switch that lasts at least 350 ms. When this condition is detected, the system interprets this as a disconnect indication from the switch and clears the call. You should disable supervisory disconnect on the voice port if there is no supervisory disconnect available from the switch. Typically, supervisory disconnect is available when connecting to the PSTN and is enabled by default. When the connection extends out to a PBX, you should verify the documentation to ensure that supervisory disconnect is supported.

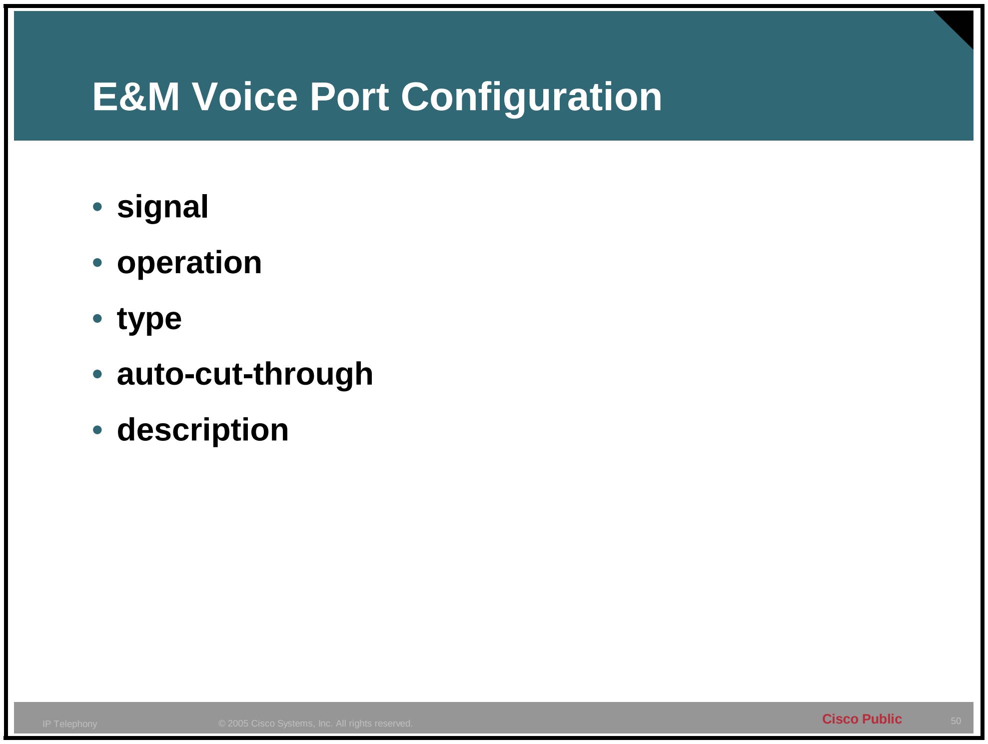
Example: FXO Port Configuration

The configuration in the figure enables loop-start signaling on a Cisco 2600 or 3600 series router on FXO voice port 1/0/0. The ring-number setting of "3" specifies that the FXO port does not answer the call until after the third ring, and the dial type is set to DTMF.



E&M Ports

E&M ports provide signaling that is used generally for switch-to-switch or switch-to-network trunk connections. This topic identifies the configuration parameters that are specific to the E&M port.



Configuration Parameters

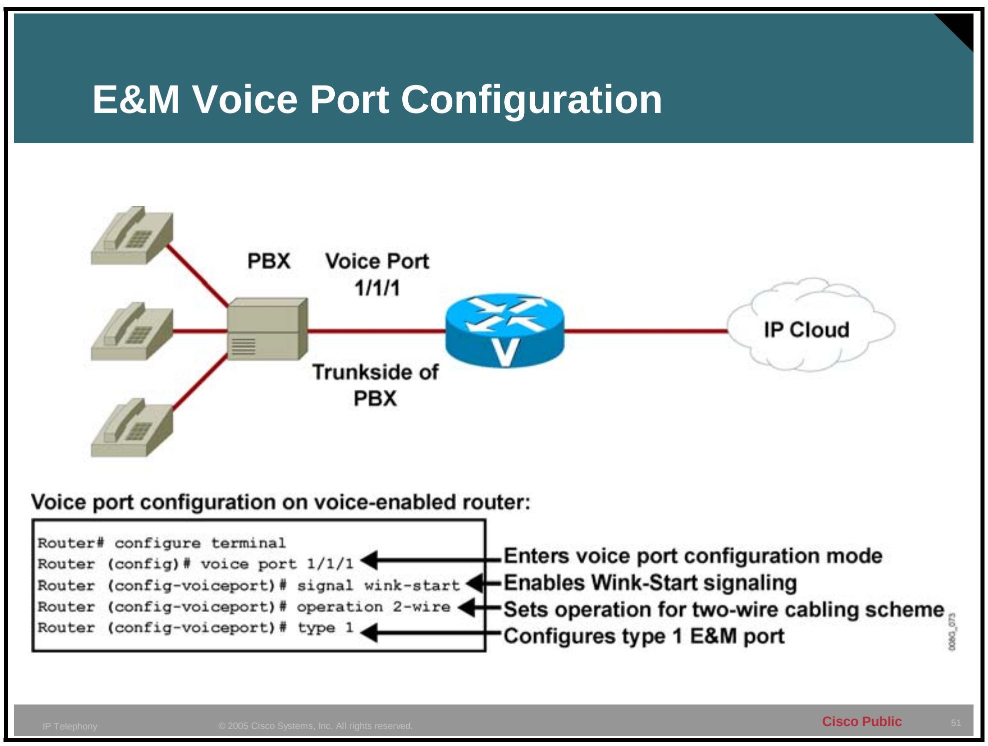
Although E&M ports have default parameters, you must usually configure these parameters to match the device that is connected to the E&M port. You can set the following configuration parameters:

- **signal:** Configures the signal type for E&M ports and defines the signaling that is used when notifying a port to send dialed digits. This setting must match that of the PBX to which the port is connected. You must shut down and reactivate the voice port before the configured value takes effect. With Wink-Start signaling, the router listens on the M-lead to determine when the PBX wants to place a call. When the router detects current on the M-lead, it waits for availability of digit registers and then provides a short *wink* on the E-lead to signal the PBX to start sending digits. With delay-start, the router provides current on the E-lead immediately upon seeing current on the M-lead. When current is stopped for the digit-sending duration, the E-lead stays high until digit registers are available. With immediate-start, the PBX simply waits a short time after raising the M-lead and then sends the digits without a signal from the router.
- **operation:** Configures the cabling scheme for E&M ports. The **operation** command affects the voice path only. The signaling path is independent of two-wire versus four-wire settings. If the wrong cable scheme is specified, the user may get voice traffic in one direction only. You must verify with the PBX configuration to ensure that the settings match. You must then shut down and reactivate the voice port for the new value to take effect.

- **type:** Configures the E&M interface type for a specific voice port. The type defines the electrical characteristics for the E- and M-leads. The E- and M-leads are monitored for on-hook and off-hook conditions. From a PBX perspective, when the PBX attempts to place a call, it goes high (off hook) on the M-lead. The switch monitors the M-lead and recognizes the request for service. If the switch attempts to pass a call to the PBX, the switch goes high on the E-lead. The PBX monitors the E-lead and recognizes the request for service by the switch. To ensure that the settings match, you must verify them with the PBX configuration.
- **auto-cut-through:** Configures the ability to enable call completion when a PBX does not provide an M-lead response. For example, when the router is placing a call to the PBX, even though they may have the same correct signaling configured, not all PBXs provide the wink with the same duration or voltage. The router may not understand the PBX wink. The **auto-cut-through** command allows the router to send digits to the PBX, even when the expected wink is not detected.
- **description:** Configures a description for the voice port. Use the **description** setting to describe the voice port in **show** command output.

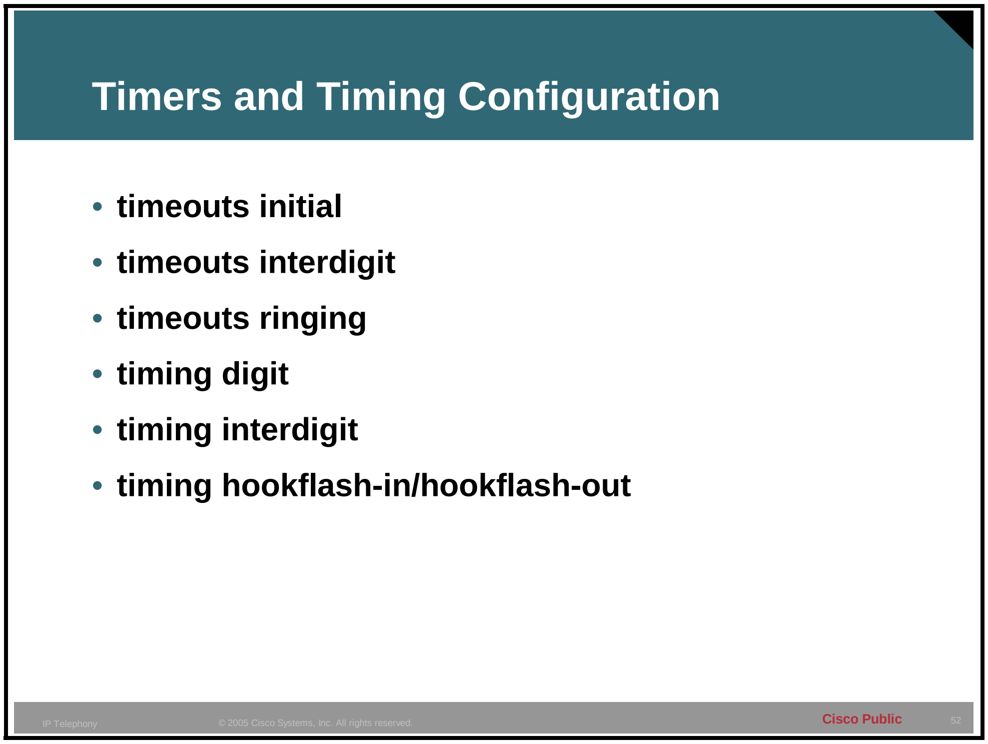
Example: E&M Port Configuration

The configuration in the figure enables Wink-Start signaling on a Cisco 2600 or 3600 series router on E&M voice port 1/1/1. The operation is set for the two-wire voice-cabling scheme and the type is set to 1.



Timers and Timing

This topic identifies the timing requirements and adjustments that are applicable to voice interfaces. Under normal use, these timers do not need adjusting. In instances where ports are connected to a device that does not properly respond to dialed digits or hookflash, or where the connected device provides automated dialing, these timers can be configured to allow more or less time for a specific function.



Configuration Parameters

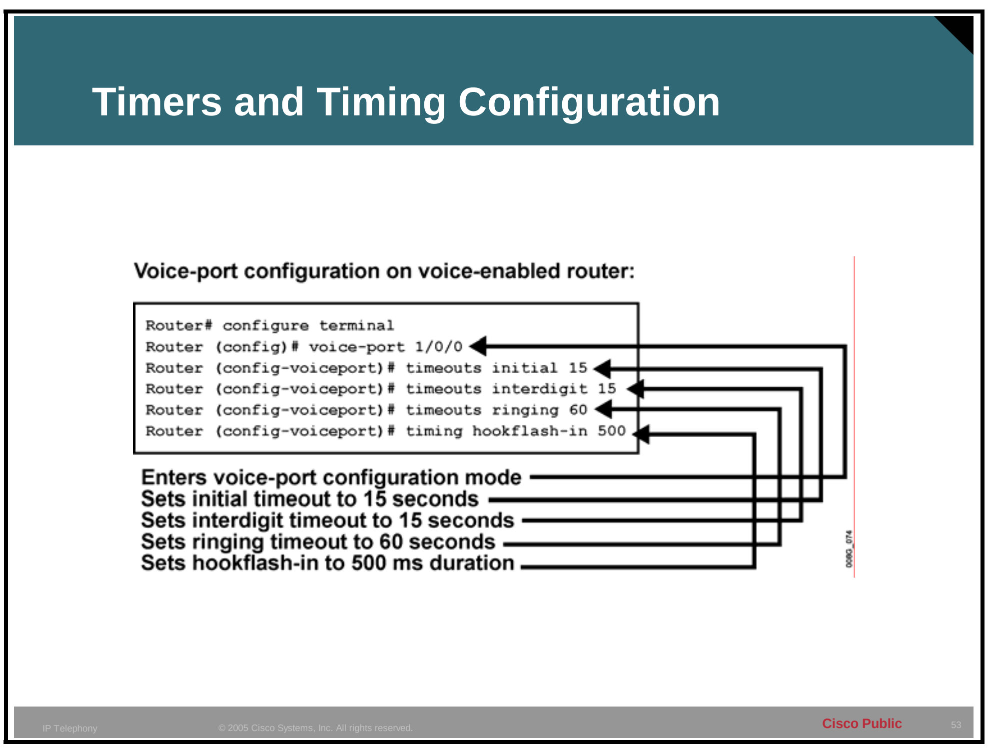
You can set a number of timers and timing parameters for fine-tuning the voice port. Following are voice port configuration parameters that you can set:

- **timeouts initial:** Configures the initial digit timeout value in seconds. This value controls how long the dial tone is presented before the first digit is expected. This timer typically does not need to be changed.
- **timeouts interdigit:** Configures the number of seconds for which the system will wait for the caller to input a subsequent digit of the dialed digits, after the caller has input the initial digit. If the digits are coming from an automated device, and the dial plan is a variable-length dial plan, you can shorten this timer so that the call proceeds without having to wait the full default of 10 seconds for the interdigit timer to expire.
- **timeouts ringing:** Configures the length of time that a caller can continue ringing a telephone when there is no answer. You can configure this setting to be less than the default of 180 seconds so that you do not tie up the voice port when it is evident that the call is not going to be answered.

- **timing digit:** Configures the DTMF digit-signal duration for a specified voice port. You can use this setting to fine-tune a connection to a device that may have trouble recognizing dialed digits. If a user or device dials too quickly, the digit may not be recognized. By changing the timing on the digit timer, you can provide for a shorter or longer DTMF duration.
- **timing interdigit:** Configures the DTMF interdigit duration for a specified voice port. You can change this setting to accommodate faster or slower dialing characteristics.
- **timing hookflash-in and hookflash-out:** Configures the maximum duration (in milliseconds) of a hookflash indication. Hookflash is an indication by a caller that the caller wishes to do something specific with the call, such as transfer the call or place the call on hold. For hookflash-in, if the hookflash lasts longer than the specified limit, the FXS interface processes the indication as on hook. If you set the value too low, the hookflash may be interpreted as a hang up; if you set the value too high, the handset has to be left hung up for a longer period to clear the call. For hookflash-out, the setting specifies the duration (in milliseconds) of the hookflash indication that the gateway generates outbound. You can configure this to match the requirements of the connected device.

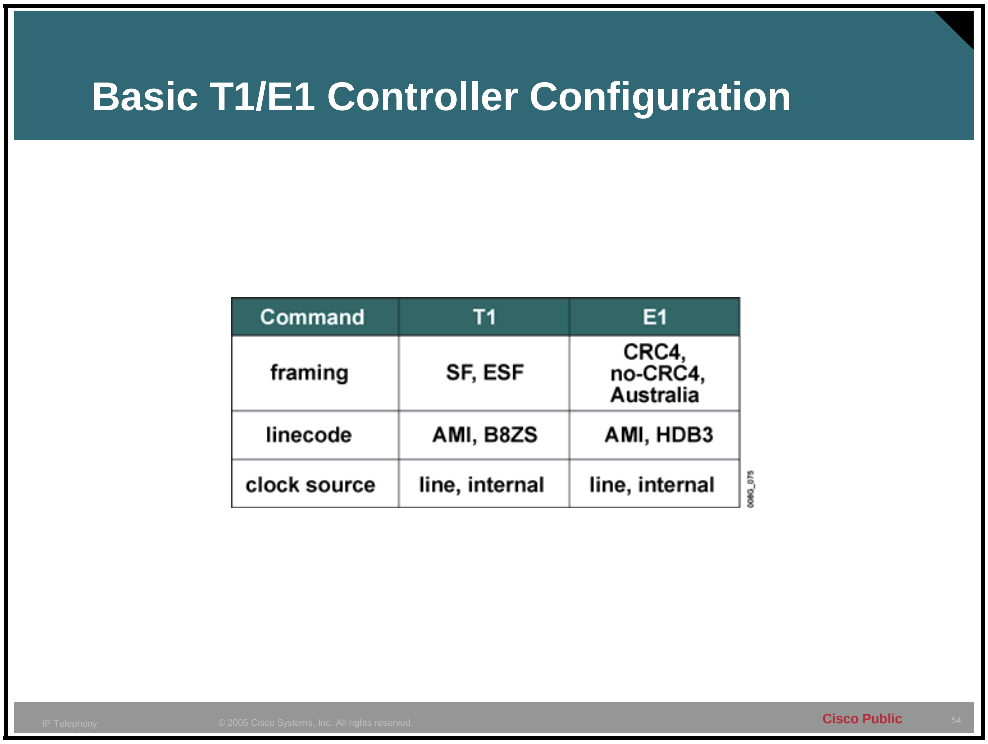
Example: Timers Configuration

The installation in the figure is for a home for the elderly, where users may need more time to dial digits than in other residences. Also, the requirement is to allow the telephone to ring, unanswered, for only one minute. The configuration in the figure enables several timing parameters on a Cisco voice-enabled router voice port 1/0/0. The initial timeout is lengthened to 15 seconds, the interdigit timeout is lengthened to 15 seconds, the ringing timeout is set to 60 seconds, and the hookflash-in timer is set to 500 ms.



Digital Voice Ports

This topic identifies the configuration parameters that are specific to T1 and E1 digital voice ports.



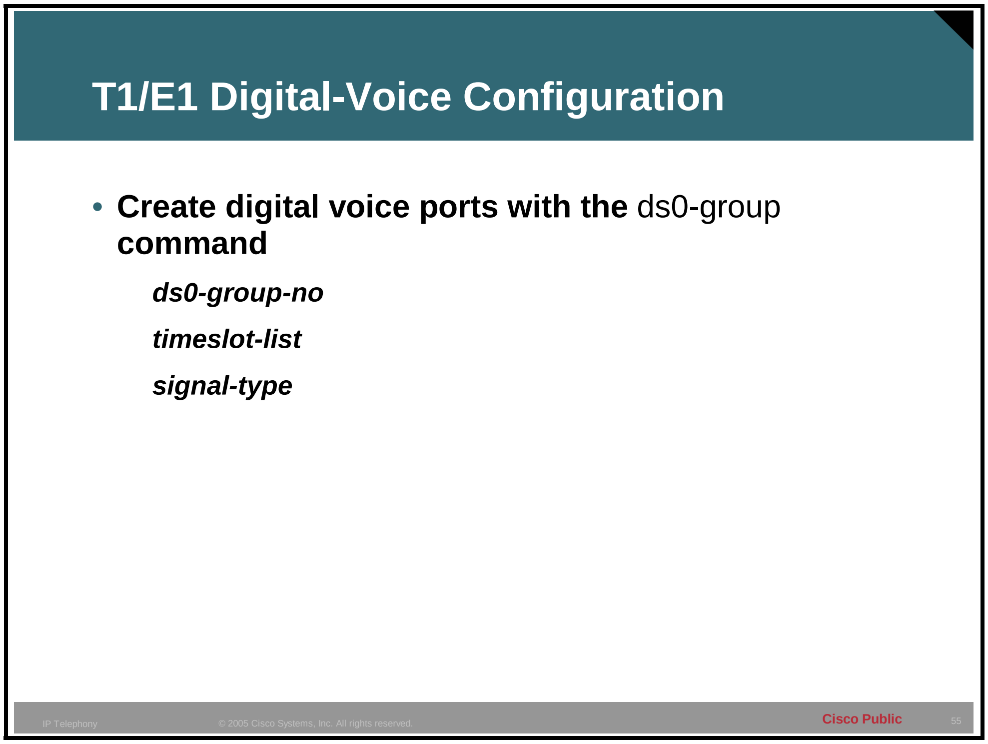
Command	T1	E1
framing	SF, ESF	CRC4, no-CRC4, Australia
linecode	AMI, B8ZS	AMI, HDB3
clock source	line, internal	line, internal

Configuration Parameters

When you purchase a T1 or E1 connection, make sure that your service provider gives you the appropriate settings. Before you configure a T1 or E1 controller to support digital voice ports, you must enter the following basic configuration parameters to bring up the interface.

- **framing:** Selects the frame type for a T1 or E1 data line. The framing configuration differs between T1 and E1.
  - **Options for T1:** Super Frame (SF) or Extended Superframe (ESF)
  - **Options for E1:** 4-bit cyclic redundancy check (CRC4), no-CRC4, or Australia
  - **Default for T1:** SF
  - **Default for E1:** CRC4
- **linecode:** Configures the line-encoding format for the DS1 link.
  - **Options for T1:** alternate mark inversion (AMI) or binary 8-zero substitution (B8ZS)
  - **Options for E1:** AMI or high density binary 3 (HDB3)
  - **Default for T1:** AMI
  - **Default for E1:** HDB3
- **clock source:** Configures clocking for individual T1 or E1 links.
  - **Options:** line or internal

— **Default:** line



You must create a digital voice port in the T1 or E1 controller to make the digital voice port available for specific voice port configuration parameters. You must also assign timeslots and signaling to the logical voice port through configuration. The first step is to create the T1 or E1 digital voice port with the **ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type** *signal-type* command.

The **ds0-group** command automatically creates a logical voice port that is numbered as *slot/port*:*ds0-group-no*.

The *ds0-group-no* parameter identifies the DS0 group (number from 0 to 23 for T1 and from 0 to 30 for E1). This group number is used as part of the logical voice port numbering scheme.

The **timeslots** command allows the user to specify which timeslots are part of the DS0 group. The *timeslot-list* parameter is a single timeslot number, a single range of numbers, or multiple ranges of numbers separated by commas.

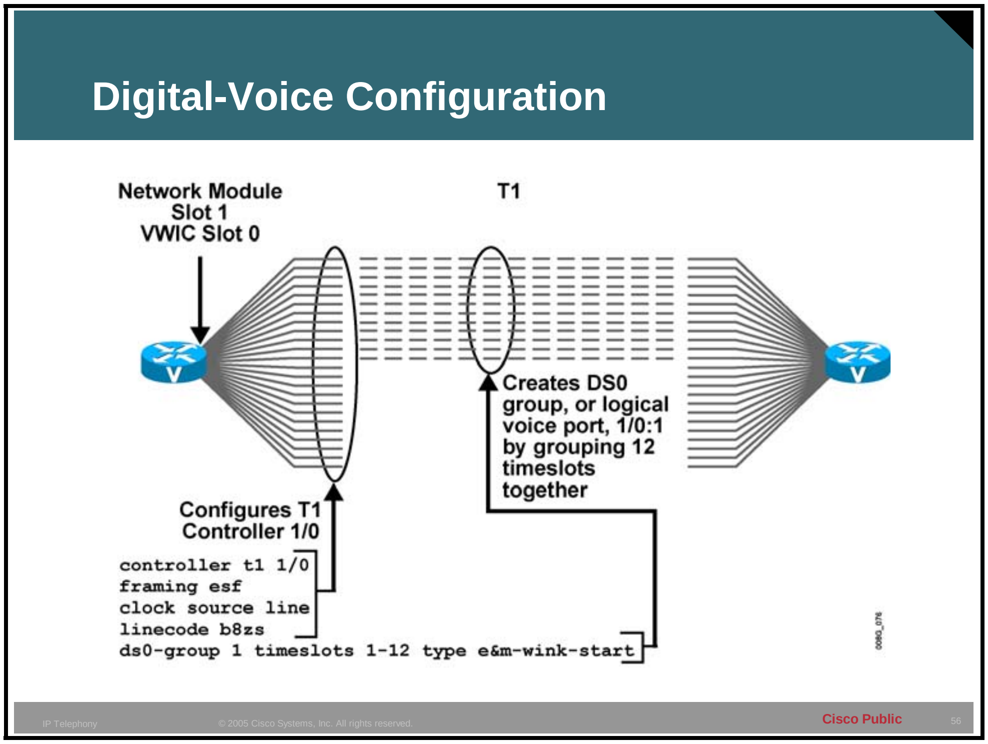
The **type** command defines the emulated analog signaling method that the router uses to connect to the PBX or PSTN. The type depends on whether the interface is T1 or E1.

After you specify a **ds0-group** command, the system creates a logical voice port. You must then enter the voice-port configuration mode to configure port-specific parameters. To enter voice-port configuration mode on a Cisco 2600 or 3600 series platform, use the **voice-port** *slot/port*:*ds0-group-no* command.

To delete a DS0 group, you must first shut down the logical voice port. When the port is in shutdown state, you can remove the DS0 group from the T1 or E1 controller with the **no ds0-group** *ds0-group-no* command.

Example: T1 Configuration

This example configures the T1 controller for ESF, B8ZS line code, and timeslots 1 through 12 with E&M Wink-Start signaling. The resulting logical voice port is **1/0:1**, where **1/0** is the module and slot number and **:1** is the *ds0-group-no* value that was assigned during configuration. You can configure the remaining timeslots for other signaling types or leave them unused.



ISDN

This topic identifies ISDN configurations for voice ports.

ISDN Configuration

- **Global configuration**
  isdn switch-type
- **T1/E1 controller configuration**
  pri-group
- **D channel configuration**
  isdn incoming-voice configuration
- **QSIG configuration**
  QSIG signaling

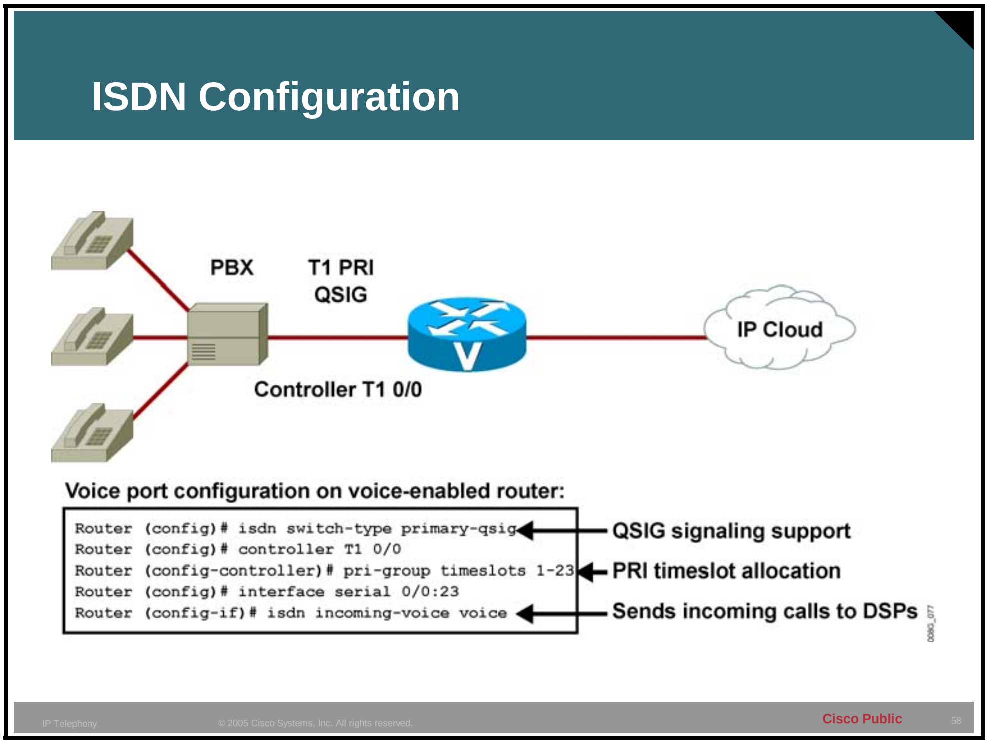
Configuration Parameters

Cisco voice-capable devices provide support for both PRI and BRI voice connections. Many PBX vendors support either T1/E1 PRI or BRI connections. In Europe, where ISDN is more popular, many PBX vendors support BRI connections. When designing how the PBX passes voice to the network, you must ensure that the router supports the correct connection. The first step in configuring ISDN capabilities for T1 or E1 PRI is to configure the T1 or E1 controller basics. After the clock source, framing, and line code are configured, ISDN voice functionality requires the following configuration commands:

- **isdn switch-type:** Configures the ISDN switch type. You can enter this parameter in global configuration mode or at the interface level. If you configure both, the interface switch type takes precedence over the global switch type. This parameter must match the provider ISDN switch. This setting is required for both BRI and PRI connections.
- **pri-group:** Configures timeslots for the ISDN PRI group. T1 allows for timeslots 1 to 23, with timeslot 24 allocated to the D channel. E1 allows for timeslots 1 to 31, with timeslot 16 allocated to the D channel. You can configure the PRI group to include all available timeslots, or you can configure a select group of timeslots for the PRI group.
- **isdn incoming-voice voice:** Configures the interface to send all incoming calls to the digital signal processor (DSP) card for processing.
- **QSIG signaling:** Configures the use of Q Signaling (QSIG) signaling on the D channel. You typically use this setting when connecting via ISDN to a PBX. The command to enable QSIG signaling is **isdn switch-type primary-qsig** for PRI and **isdn switch-type basic-qsig** for BRI connections.

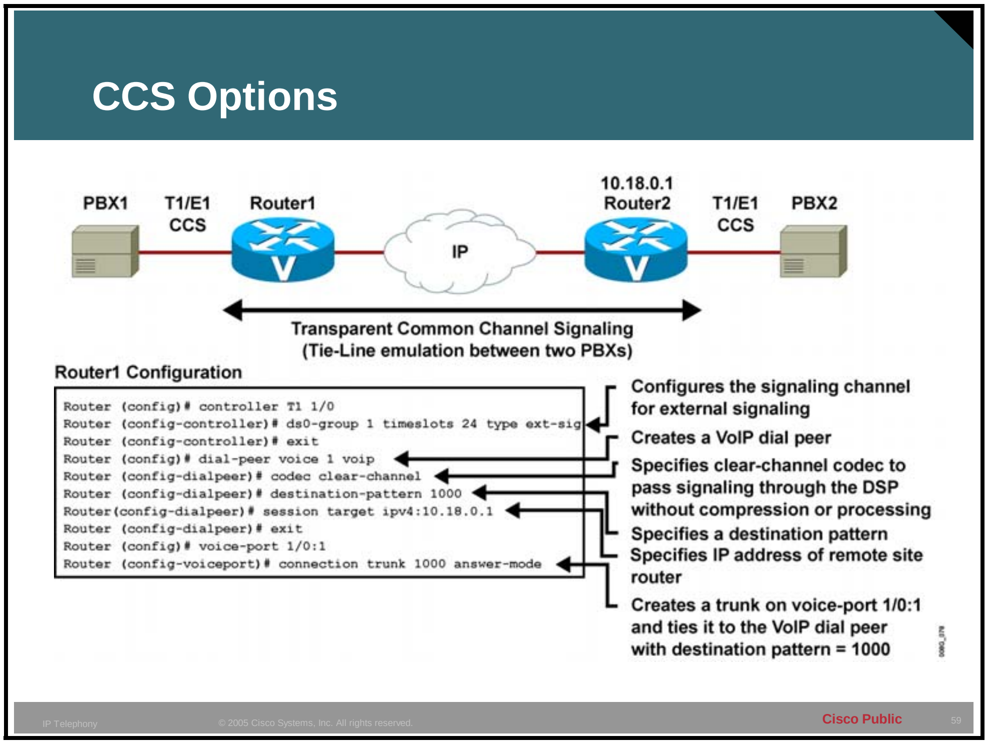
Example: ISDN QSIG Configuration

This example shows the configuration for a PBX connection to the Cisco voice-enabled router. The connection is configured for QSIG signaling across all 23 timeslots.



CCS Options

This topic describes how to pass proprietary signaling between two PBXs through the use of Transparent Common Channel Signaling (T-CCS).



In many cases, PBXs support proprietary signaling that is used to signal supplementary services only, such as making a light on the telephone blink when voice mail is waiting. Because the router does not understand this proprietary signaling, the signaling must be carried transparently across the network without interpretation. T-CCS allows the connection of two PBXs with digital interfaces that use a proprietary or unsupported common channel signaling (CCS) protocol. T1 and E1 traffic is transported transparently through the data network, and the T-CCS feature preserves proprietary signaling. From the PBX standpoint, this type of communication is accomplished through a point-to-point connection. Calls from the PBXs are not routed, but they do follow a preconfigured route to the destination.

T-CCS Configuration Process

The configuration for T-CCS in a Voice over IP (VoIP) environment calls for the following three-step process:

Step 1 Define the DS0 group.

Configure the command **ds0-group** *ds0-group-no* **timeslots** *timeslot-list* **type ext-sig** in the T1 or E1 controller configuration mode. The **timeslots** command specifies the D channel that carries call signaling. The **type ext-sig** command specifies that the signaling is coming from an external source.

Step 2 Create the dial peer.

- Configure a VoIP dial peer that points to the IP address of the remote voice-enabled router that connects to the remote PBX.

- Configure the dial peer for clear-channel codec that signals the DSP to pass the signaling without interpretation.
- The destination pattern specified in this dial peer is used to create a trunk in Step 3. The number entered here must match the number entered in the **trunk** command.
- The session target specifies the IP address of the remote voice-enabled router.
- Configure the dial peer to point to the IP address of the remote site voice-enabled router using the **session target** command.

Step 3 Create the voice port trunk.

Configure the **connection trunk** *digits* **answer-mode** command at the logical voice port to create a trunk from that port through the VoIP dial peer and across the IP network to the remote router. The *digits* parameter must match the destination pattern in the VoIP dial peer created in Step 2. The **answer-mode** parameter specifies that the router should not attempt to initiate a trunk connection but should wait for an incoming call before establishing the trunk.

The process for passing the signal transparently through the IP network is as follows:

Step 1 PBX1 sends proprietary signaling across the signaling channel to router 1.

Step 2 The logical voice port that corresponds to the signaling channel is configured for trunking, so the router looks for the dial peer that matches the **trunk** *digits* parameter.

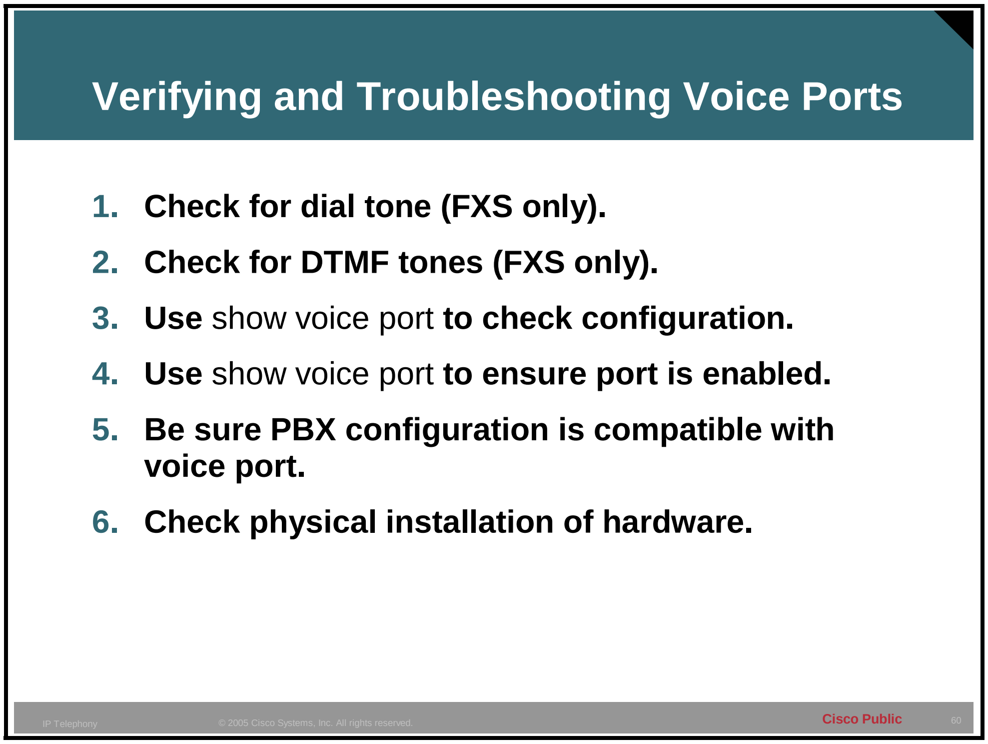
Step 3 The VoIP dial peer is configured for clear-channel codec and points to the IP address of the remote router (router 2) connecting the remote PBX (PBX2).

Step 4 The remote router has a plain old telephone service (POTS) dial peer configured that points to the logical voice port that is associated with the signaling channel of PBX2. The signal arrives at PBX2 in its native form.

This process shows the T-CCS signaling part of the configuration only. Additional DS0 group and dial-peer configuration is necessary for transport of the voice channels.

Monitoring and Troubleshooting

This topic describes the **show** and **test** commands that are used to monitor and troubleshoot voice ports.



You can perform the following steps to verify voice port configuration:

Step 1 Pick up the handset of an attached telephony device and check for a dial tone. If there is no dial tone, check the following:

- Is the plug firmly seated?
- Is the voice port enabled?
- Is the voice port recognized by the Cisco IOS?
- Is the router running the correct version of Cisco IOS in order to recognize the module?

Step 2 If you have a dial tone, check for DTMF voice band tones, such as touch-tone detection. If the dial tone stops when you dial a digit, the voice port is probably configured properly.

Step 3 Use the **show voice port** command to verify that the data configured is correct. If you have trouble connecting a call, and you suspect that the problem is associated with voice port configuration, you can try to resolve the problem by performing Steps 4 through 6.

Step 4 Use the **show voice port** command to make sure that the port is enabled. If the port is administratively down, use the **no shutdown** command. If the port was working previously and is not working now, it is possible that the port is in a hung state. Use the **shutdown**/**no shutdown** command sequence to reinitialize the port.

Step 5 If you have configured E&M interfaces, make sure that the values associated with your specific PBX setup are correct. Specifically, check for two-wire or four-wire Wink-Start, immediate-start, or delay-start signaling types, and the E&M interface type. These parameters need to match those set on the PBX for the interface to communicate properly.

Step 6 You must confirm that the voice network module (VNM) is correctly installed. With the device powered down, remove the VNM and reinsert it to verify the installation. If the device has other slots available, try inserting the VNM into another slot to isolate the problem. Similarly, you must move the voice interface card (VIC) to another VIC slot to determine if the problem is with the VIC card or with the module slot.

Commands to Verify Voice Ports

Command	Description
`show voice port`	Shows all voice port configurations in detail
`show voice port x/y/z`	Shows one voice port configuration in detail
`show voice port summary`	Shows all voice port configurations in brief
`show voice busyout`	Shows all ports configured as busyout
`show voice dsp`	Shows all DSP status
`show controller T1	E1`	Shows the operational status of the controller

There are six **show** commands for verifying the voice port and dial-peer configuration. These commands and their functions are shown in the figure.

Test Commands

Command	Description
test voice port detector {M lead | battery-reversal | ring | tip-ground | ring-ground | ring-trip} {on | off | disable}	Forces a detector into specific states for testing. For each signaling type (E&M, FXO, FXS), only the applicable keywords display.
test voice port inject-tone {local | network} {1000hz | 2000hz | 200hz | 3000hz | 300hz | 3200hz | 3400hz | 500hz | quiet | disable}	Injects a test tone into a voice port. A call must be established on the voice port under test. When you are finished testing, be sure to enter the disable command to end the test tone.
test voice port loopback {local | network | disable}	Performs loopback testing on a voice port. A call must be established on the voice port under test.
test voice port relay {E lead | loop | ring-ground | battery-reversal | power-denial | ring | tip-ground} {on | off | disable}	Tests relay-related functions on a voice port.
test voice port switch {fax | disable}	Forces a voice port into fax or voice mode for testing. If the voice port does not detect fax data, the voice port remains in fax mode for 30 seconds and then reverts automatically to voice mode.
csim xxxx	Simulates a call to destination xxxx.

The **test** commands provide the ability to analyze and troubleshoot voice ports on the Cisco 2600 and 3600 series routers. There are five **test** commands to force voice ports into specific states to test the voice port configuration.

When you finish the loopback testing, be sure to enter the **disable** command to end the forced loopback.

After you enter the **test voice port switch fax** command, you can use the **show voice call** command to check whether the voice port is able to operate in fax mode.

The **csim** command simulates a call to any end station for testing purposes. It is most useful when testing dial plans.

Note	Refer to the *Voice Port Testing Enhancements in Cisco 2600 and 3600 Series Routers* document for further information.

ISDN Commands

Command	Description
`show isdn active`	Shows ISDN active calls
`show isdn history`	Shows ISDN call history
`show isdn status`	Shows ISDN line status
`show isdn timers`	Shows ISDN timer values
`debug isdn events`	Displays ISDN events
`debug isdn q921`	Displays ISDN Q.921 packet history
`debug isdn q931`	Displays ISDN Q.931 packet history

The ISDN **show** and **debug** commands in the figure are useful for viewing and troubleshooting ISDN connections.