

After reading this chapter, you should be able to perform the following tasks:

- Configure analog and digital voice interfaces as new devices are introduced into the voice path
- Configure analog and digital voice ports for optimal voice quality

Voice Interface Configuration

Voice gateways bridge the gap between the VoIP world and the traditional telephony world (for example, a PBX, the PSTN, or an analog phone). Cisco voice gateways connect to traditional telephony devices via voice ports. This chapter introduces basic configuration of analog and digital voice ports, and demonstrates how to fine-tune voice ports with port-specific configurations.

Upon completing this chapter, you will be able to configure voice interfaces on Cisco voice-enabled equipment for connection to traditional, nonpacketized telephony equipment.

Configuring Voice Ports

Connecting voice devices to a network infrastructure requires an in-depth understanding of signaling and electrical characteristics that are specific to each type of interface. Improperly matched electrical components can cause echo and make a connection unusable. As another consideration, configuring devices for international implementation requires knowledge of country-specific settings. This section provides voice port configuration parameters for signaling and country-specific settings.

Before delving into the specific syntax of configuring these voice ports, this section begins by considering several examples of voice applications. The applications discussed help illustrate the function of the voice ports, whose configuration is addressed at the end of this section.

Voice Applications

Different types of applications require specific types of ports. In many instances, the type of port is dependent on the voice device connected to the network. Different types of voice applications include the following:

- Local calls
- On-net calls
- Off-net calls
- Private line, automatic ringdown (PLAR) calls

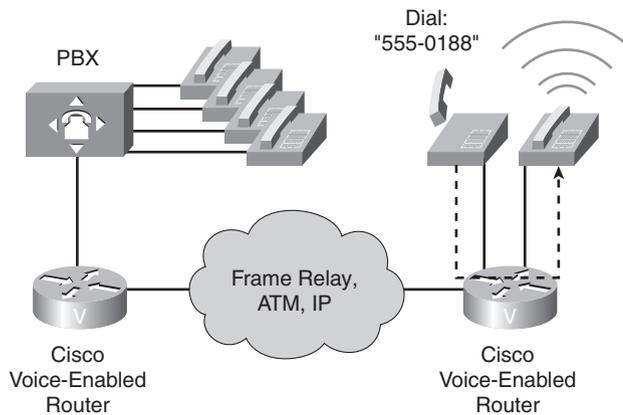
- PBX-to-PBX calls
- CallManager-to-CallManager calls
- On-net to off-net calls

The following sections describe each type and provide an illustration of each.

Local Calls

Local calls, as illustrated in Figure 3-1, 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.

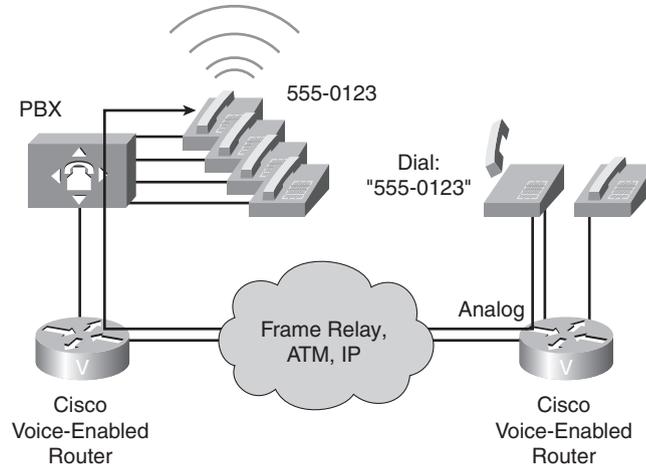
Figure 3-1 *Local Calls*



On-Net Calls

On-net calls occur between two telephones on the same data network, as shown in Figure 3-2. 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 through Cisco Unified 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.

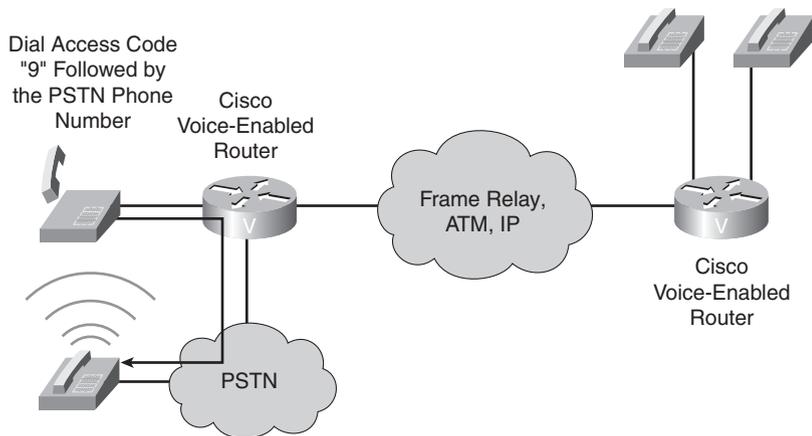
Figure 3-2 *On-Net Calls*



Off-Net Calls

Figure 3-3 shows an example of an off-net call. 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 typically a single analog connection via a Foreign Exchange Office (FXO) port or a digital T1 or E1 connection.

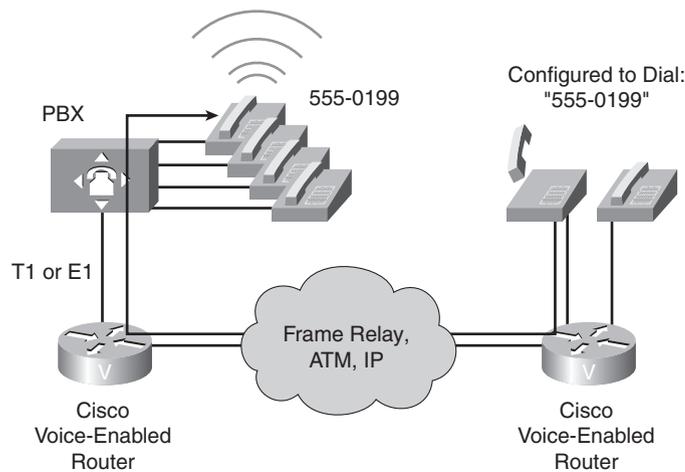
Figure 3-3 *Off-Net Calls*



PLAR Calls

PLAR calls automatically connect a telephone to a second telephone when the first telephone goes off hook, as depicted in Figure 3-4. 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. As an example, you might have encountered a PLAR connection at an airline ticket counter, where you pick up a handset and are immediately connected with an airline representative.

Figure 3-4 *PLAR Calls*

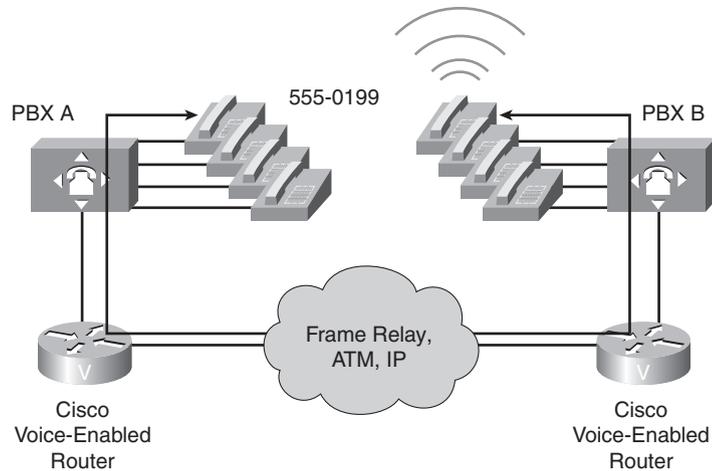


PBX-to-PBX Calls

PBX-to-PBX calls, as shown in Figure 3-5, originate at a PBX at one site and terminate 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.

CallManager-to-CallManager Calls

As part of an overall migration strategy, a business might replace PBXs with a Cisco Unified CallManager infrastructure. This infrastructure includes IP telephones that

Figure 3-5 *PBX-to-PBX Calls*

plug directly into the IP network. Cisco Unified CallManager performs the same call-routing functions formerly provided by the PBX. When an IP phone uses Cisco Unified CallManager to place a call, Cisco CallManager, based on its configuration, assesses whether the call is destined for another IP phone under its control or whether the call must be routed through a remote Cisco CallManager for call completion. Although the call stays on the IP network, it might 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 Unified CallManager or gatekeeper. Figure 3-6 provides an example of a CallManager-to-CallManager call.

On-Net to Off-Net Calls

When planning a resilient call-routing strategy, it might be necessary to reroute calls through a secondary path should the primary path fail. An on-net to off-net call, as illustrated in Figure 3-7, originates on an internal network and is routed to an external network, usually to the PSTN. On-net to off-net call-switching functionality might be necessary when a network link is down or if a network becomes overloaded and unable to handle all calls presented.

Figure 3-6 *CallManager-to-CallManager Calls*

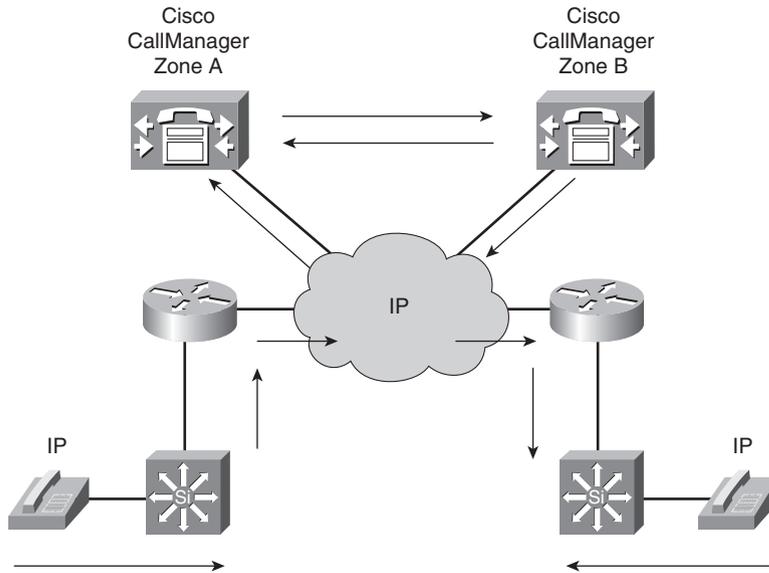
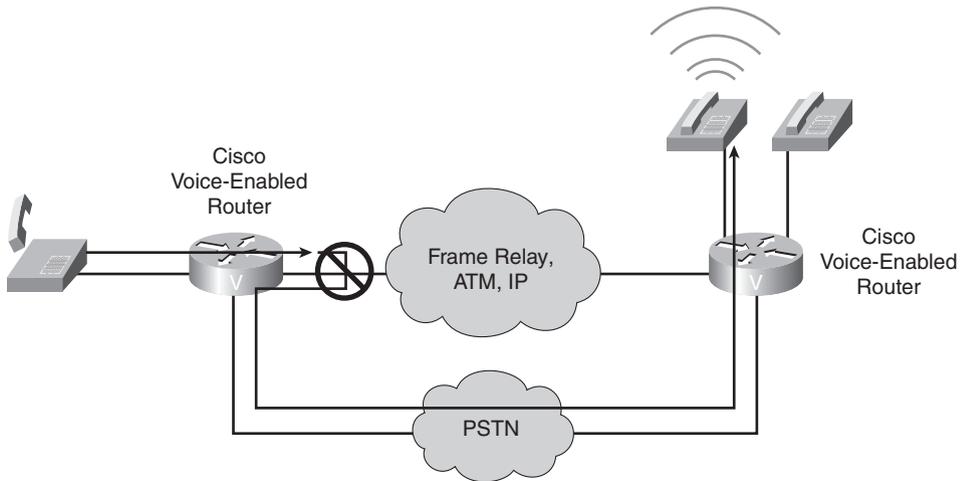


Figure 3-7 *On-Net to Off-Net Calls*



Summarizing Examples of Voice Port Applications

Table 3-1 lists application examples for each type of call.

Table 3-1 *Voice Port Call Types*

Type of Call	Example
Local call	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 call	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 call	A staff member calls a client who is located in the same city. The call is sent from the local voice-enabled router, which acts as a gateway, to the PSTN. The call is then sent to the PSTN for call termination.
PLAR call	A client picks up a customer service telephone located in the lobby of an 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 call	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.
CallManager-to-CallManager call	One staff member calls another staff member at a remote office using IP phones. The call setup is handled by the Cisco Unified 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 call	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, such as analog phones, fax machines, and modems. 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. For example, if the local PSTN uses loop-start signaling, the FXS port should use loop-start signaling.

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 dial tone in England. Also, 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.

FXS Configuration Parameters

FXS port configuration allows you to set parameters based on the requirements of the connection. You can alter the default settings and fine-tune the parameters for specific needs. For example, you might need to connect an older phone with a mechanical ringer to a router, and the FXS port's default ringing frequency does not work for the older phone's ringer. You could adjust the ringing frequency parameter to fix that problem. You can set the following configuration parameters for an FXS port:

- **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 might 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 can use the description setting to describe the voice port in **show** command output. It is almost 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.

NOTE

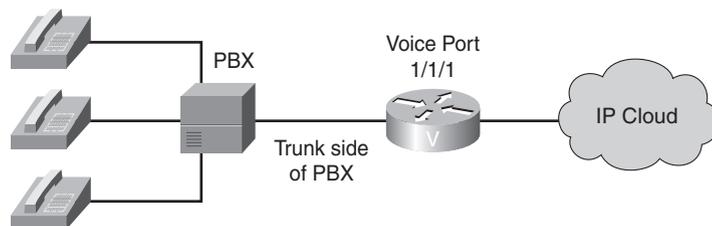
Typically, the ring frequency parameter is an issue only for electromechanical bell ringers. Back in the days of party lines (that is, where multiple residences shared a local loop connection going back to the CO), these mechanical ringers were tuned to ring at different frequencies. However, most modern piezoelectric ringers are unaffected by the ring frequency setting.

- **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. The United Kingdom uses a double ring of 0.4 seconds separated by 0.2 seconds of silence, followed by 2 seconds of silence. 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 might 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, perhaps for maintenance purposes.
- **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. 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. Maximum string length is 15.

Configuring FXS Ports

Figure 3-8 shows an FXS port of a router connecting into a PBX.

Figure 3-8 *FXS Voice Port Configuration*



The syntax given in Example 3-1 shows how the British office, in Figure 3-8, is configured to enable ground-start signaling on a router on FXS voice port 1/1/1. Notice the call-progress tones are set for Great Britain, and the ring cadence is set for pattern 1.

Example 3-1 FXS Voice Port Configuration on a Voice-Enabled Router

```
Router#configure terminal
Router(config)#voice-port 1/1/1
!Enters voice-port configuration mode
Router(config-voiceport)#signal ground-start
!Enables ground-start signaling
Router(config-voiceport)#cptone GB
!Sets call-progress tones for Great Britain
Router(config-voiceport)#ring cadence pattern01
!Specifies ring cadence pattern 1
```

FXO Ports

FXO ports act like telephones and connect to central office (CO) switches or to a station port on a PBX. This section describes FXO port configuration parameters and provides a sample FXO port configuration.

FXO 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. As with FXS ports, you can alter default settings and fine-tune parameters. You can set the following configuration parameters for an FXO port:

- **signal**—Sets the signaling type for the FXO port. If the FXO port is connected to the PSTN, the default setting of *loop start* is usually appropriate. 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 might not support dual-tone multifrequency (DTMF) dialing. If you are connecting an FXO port to this type of device, you might 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.

Configuring FXO Ports

Figure 3-9 shows a router's FXO port connecting into a PBX.

Figure 3-9 *FXO Voice Port Configuration*



The configuration in Example 3-2 enables loop-start signaling on the 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.

Example 3-2 *FXO Voice Port Configuration on a Voice-Enabled Router*

```
Router#configure terminal
Router(config)#voice-port 1/0/0
!Enters voice-port configuration mode
Router(config-voiceport)#signal loop-start
!Enables loop-start signaling
Router(config-voiceport)#ring number 3
!Sets FXO port to answer after three rings
Router(config-voiceport)#dial-type dtmf
!Specifies dial type of DTMF
```

E&M Ports

E&M ports provide signaling that is used generally for switch-to-switch or switch-to-network trunk connections. This section describes E&M configuration parameters and provides an example of an E&M port configuration.

E&M 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 which defines the signaling 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.

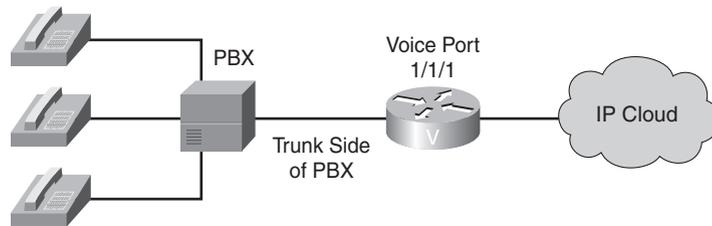
With wink-start signaling, the router listens on the M-lead to determine when the PBX wants to place a call. When the router detects current on the M-lead, it waits for availability of digit registers and then provides a short wink on the E-lead to signal the PBX to start sending digits.

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 operation settings. If the wrong cable scheme is specified, the user might get voice traffic in one direction only. You must check 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 check them against 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 might have the same correct signaling configured, not all PBXs provide the wink with the same duration or voltage. Therefore, the router might 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.

Configuring E&M Ports

Figure 3-10 shows a router's E&M port connecting to a PBX's E&M interface.

Figure 3-10 E&M Voice Port Configuration

The configuration in Example 3-3 enables wink-start signaling on the router, on E&M voice port 1/1/1. The operation is set for the two-wire voice-cabling scheme, meaning the voice path consists of two wires, and the type is set to 1.

Example 3-3 E&M Voice Port Configuration on a Voice-Enabled Router

```

Router#configure terminal
Router(config)#voice port 1/1/1
!Enters voice port configuration mode
Router(config-voiceport)#signal wink-start
!Enables Wink-Start signaling
Router(config-voiceport)#operation 2-wire
!Sets operation for two-wire cabling scheme
Router(config-voiceport)#type 1
!Configures type 1 E&M port

```

Timers and Timing

A variety of default timer settings are configured for various voice ports. 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 (that is, quickly transitioning from the off-hook state to the on-hook state and back to the off-hook state) or where the connected device provides automated dialing, these timers can be configured to allow more or less time for a specific function.

Timers and Timing 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 from its default of 10 seconds.

- **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 might have trouble recognizing dialed digits. If a user or device dials too quickly, the digit might 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.

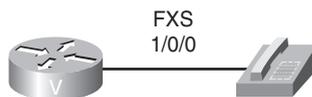
For **hookflash-in**, if the hookflash lasts longer than the specified limit, the FXS interface processes the indication as on hook. If you set the value too low, the hookflash might be interpreted as a hang up; if you set the value too high, the handset has to be left hung up for a longer period to 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.

Configuring Timers

The installation in Figure 3-11 serves as a solution in a home for the elderly, where users might need more time to dial digits than in other residences. Also, there is a requirement to allow the telephone to ring, unanswered, for only one minute. Figure 3-11 shows an analog phone connecting into FXS port 1/0/0 on a voice-enabled router.

Figure 3-11 *Timers and Timing Configuration*



The FXS port's configuration, as shown in Example 3-4, enables several timing parameters for this voice port. 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.

Example 3-4 *Timer Voice Port Configuration on a Voice-Enabled Router*

```
Router#configure terminal
Router(config)#voice-port 1/0/0
!Enters voice-port configuration mode
Router(config-voiceport)#timeouts initial 15
!Sets initial timeout to 15 seconds
Router(config-voiceport)#timeouts interdigit 15
!Sets interdigit timeouts to 15 seconds
Router(config-voiceport)#timeouts ringing 60
!Sets ringing timeout to 60 seconds
Router(config-voiceport)#timing hookflash-in 500
!Sets hookflash-in to 500 ms duration
```

Digital Voice Ports

While analog voice ports typically carry a single voice call on a single port, digital voice ports often carry multiple voice conversations over a single voice port. Therefore, digital voice ports might be more appropriate for environments with a high call volume. As a rule of thumb, if a design requires more than seven voice connections, the designer should consider a digital voice port, as opposed to multiple analog voice ports, due to the economies of scale offered by the higher-density digital voice ports.

Digital Voice 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, as follows:
 - **Options for T1**—Super Frame (SF) or Extended Superframe (ESF). SF is the default.
 - **Options for E1**—4-bit cyclic redundancy check (CRC4), no-CRC4, or Australia. CRC4 is the default.

- **linecode**—Configures the line-encoding format for the DS1 link. The linecode configuration differs for T1 and E1, as follows:
 - **Options for T1**—Alternate mark inversion (AMI) or binary 8-zero substitution (B8ZS). AMI is the default.
 - **Options for E1**—AMI or high-density binary 3 (HDB3). HDB3 is the default.
- **clock source**—Configures clocking for individual T1 or E1 links. Line and internal are the options for both T1 and E1. Line is the default.

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. 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 following list describes the command syntax:

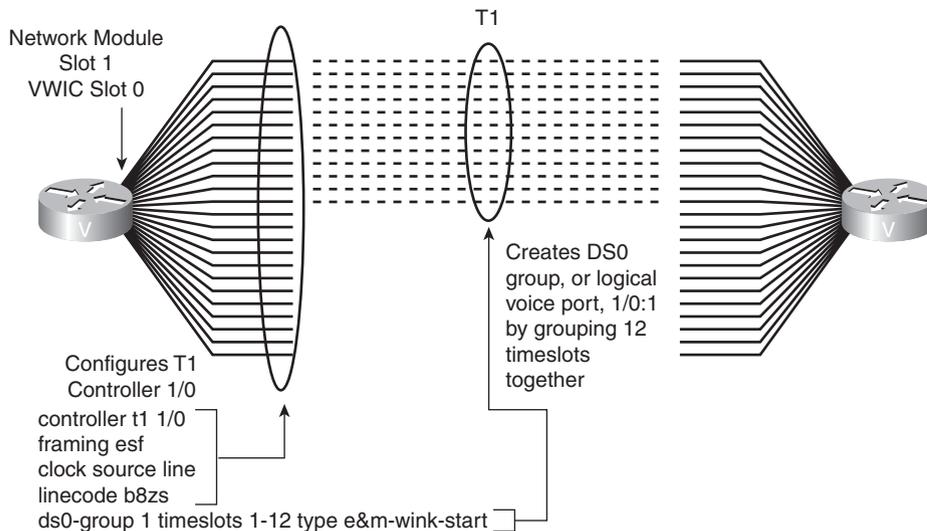
- The **ds0-group** command automatically creates a logical voice port that is numbered as *slot/port:ds0-group-no*.
- The *ds0-group-no* parameter identifies the DS0 group (number from 0 to 23 for T1 and from 0 to 30 for E1). This group number is used as part of the logical voice port numbering scheme.
- 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.
- The *signal-type* parameter is the signaling type being used by all channels in the DS0 group. For example, the signaling type could be configured as *e&m-wink-start*, which would cause each channel in the DS0 group to use E&M wink-start signaling.

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, 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.

Configuring a T1 Controller

Figure 3-12 illustrates how specific channels within a T1 can be defined as a DS0 group.

Figure 3-12 *Digital Voice Configuration*

The syntax for the topology shown is presented in Example 3-5. The example configures the T1 controller for ESF framing, B8ZS line coding, 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.

Example 3-5 *DS0 Group Configuration on a Voice-Enabled Router*

```
Router#configure terminal
Router(config)#controller t1 1/0
!Enters controller configuration mode
Router(config-controller)#framing esf
!Sets the framing to Extended Superframe
Router(config-controller)#clock source line
!Specifies that the controller receives clocking from the network
Router(config-controller)#linecode b8zs
!Sets the linecoding to Bipolar 8 Zero Substitution
Router(config-controller)#ds0-group 1 timeslots 1-12 type e&m-wink-start
!Defines a DS0 group, which contains the first 12 channels of the T1, and configures
the signaling for those channels as E&M Wink Start signaling
```

ISDN

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.

ISDN Configuration Parameters

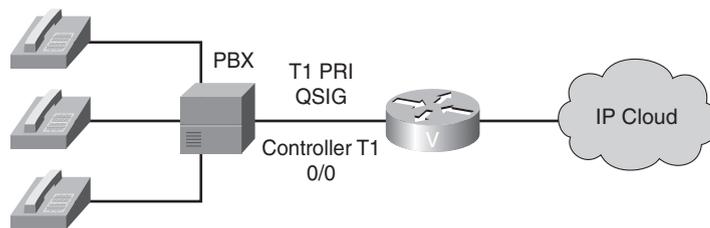
The first step in configuring ISDN capabilities for T1 or E1 PRI is to configure the T1 or E1 controller basics. After the clock source, framing, and line code are configured, ISDN voice functionality requires the following configuration commands:

- **isdn switch-type**—Configures the ISDN switch type. You can enter this parameter in global configuration mode or in interface configuration mode. 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.

Configuring ISDN

Figure 3-13 shows an ISDN port on a voice-enabled router connecting into an ISDN port on a PBX.

Figure 3-13 ISDN Configuration



Example 3-6 shows the configuration for this connection. The connection is configured for QSIG signaling across all 23 timeslots.

Example 3-6 ISDN Voice Port Configuration on a Voice-Enabled Router

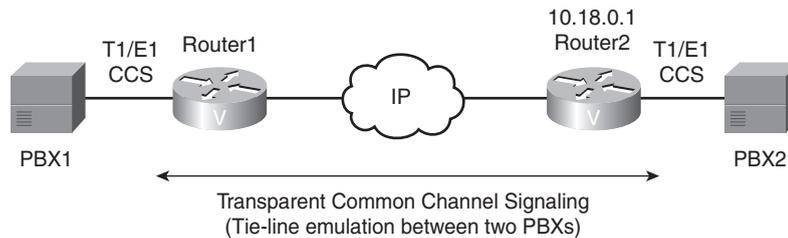
```

Router (config)#isdn switch-type primary-qsig
!QSIG signaling support
Router (config)#controller T1 0/0
Router (config-controller)#pri-group timeslots 1-23
!PRI timeslot allocation
Router (config)#interface serial 0/0:23
Router (config-if)#isdn incoming-voice voice
!Sends incoming calls to DSPs

```

Common Channel Signaling Options

Proprietary signaling can be passed between two PBXs through the use of Transparent Common Channel Signaling (T-CCS), an example of which is provided in Figure 3-14.

Figure 3-14 CCS Options

The syntax for the configuration shown is provided in Example 3-7. The **dial-peer** and **voice-port** commands are covered in Chapter 4, “Voice Dial Peer Configuration.” For now, consider the **ds0-group 1 timeslots 24 type ext-sig** command, entered in controller configuration mode. This command specifies that the 24th channel of the T1 is going to be carrying external signaling, which the router should not attempt to interpret. Also, notice the **codec clear-channel** command. This command tells the router to pass the signaling information through the DSP transparently, without any compression or processing.

Example 3-7 Transparent Common Channel Signaling Configuration

```

Router(config)#controller T1 1/0
Router(config-controller)#ds0-group 1 timeslots 24 type ext-sig
!Configures the signaling channel for external signaling
Router(config-controller)#exit
Router (config)#dial-peer voice 1 voip
!Creates a VoIP dial peer
Router(config-dialpeer)#codec clear-channel
!Specifies clear-channel CODEC to pass signaling through the DSP without
!compression or processing
Router(config-dialpeer)#destination-pattern 1000

```

continues

Example 3-7 *Transparent Common Channel Signaling Configuration (Continued)*

```
!Specifies a destination pattern
Router(config-dialpeer)#session target ipv4:10.18.0.1
!Specifies IP address of remote site router
Router(config-dialpeer)#exit
Router(config)#voice-port 1/0:1
Router(config-voiceport)#connection trunk 1000 answer-mode
!Creates a trunk on voice-port 1/0:1 and ties it to the VoIP dial peer with
!a destination pattern of 1000
```

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.

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

1. To 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.
2. Create the dial peer, as follows:
 - a Configure a VoIP dial peer that points to the IP address of the remote voice-enabled router that connects to the remote PBX.
 - b Configure the dial peer for clear-channel codec that signals the DSP to pass the signaling without interpretation.
 - c Specify the destination pattern for the dial peer, which is used to create the trunk in Step 3. The number entered here must match the number entered in the **trunk** command.
 - d Configure the session target to point to the IP address of the remote voice-enabled router.
 - e Configure the dial peer to point to the IP address of the remote site voice-enabled router using the **session target** command.
3. Create the voice port trunk.

NOTE Chapter 4 covers dial peers in depth.

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:

1. PBX1 sends proprietary signaling across the signaling channel to router 1.
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.
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).
4. The remote router has a plain old telephone service (POTS) dial peer configured that points to the logical voice port 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

After physically connecting analog or digital devices to a Cisco voice-enabled router, you might need to issue **show**, **test**, or **debug** commands to verify or troubleshoot your configuration. For example, the following list enumerates six steps to monitor and troubleshoot voice ports:

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?

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.
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.
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.
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.
6. You must confirm that the voice network module (VNM) (that is, the module in the router that contains the voice ports) 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.

For your reference, Table 3-2 lists six **show** commands for verifying the voice port configuration.

Table 3-2 *Commands to Verify Voice Ports*

Command	Description
show voice port	Shows all voice port configurations in detail
show voice port slot/ subunit/port	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 status of all DSPs
show controller T1 E1	Shows the operational status of a controller

For your further reference, Table 3-3 provides a series of commands used to test Cisco voice ports. The **test** commands provide the ability to analyze and troubleshoot voice ports on voice-enabled routers. As Table 3-3 shows, there are five **test** commands to force voice ports into specific states to test the voice port configuration. The **csim start dial-string** command simulates a call to any end station for testing purposes.

Table 3-3 test Commands

Command	Description
test voice port <i>port_or_DS0-group_identifier</i> detector { m-lead battery-reversal ring tip-ground ring-ground ring-trip } { on off disable }	Forces a detector into specific states for testing.
test voice port <i>port_or_DS0-group_identifier</i> 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 <i>port_or_DS0-group_identifier</i> loopback { local network disable }	Performs loopback testing on a voice port. A call must be established on the voice port under test. When you finish the loopback testing, be sure to enter the disable command to end the forced loopback.
test voice port <i>port_or_DS0-group_identifier</i> 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 <i>port_or_DS0-group_identifier</i> 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. 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.
csim start dial-string	Simulates a call to the specified dial string. It is most useful when testing dial plans.

Table 3-4 lists ISDN **show** and **debug** commands specific to the monitoring and troubleshooting of ISDN connections.

Table 3-4 *ISDN Commands*

Command	Description
show isdn active	Shows ISDN calls in progress
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 in real time
debug isdn q921	Displays ISDN Q.921 packets in real time
debug isdn q931	Displays ISDN Q.931 packets in real time

Tuning Voice Quality

User acceptance of the converged voice and data network depends on the quality of current calls compared to the quality through their original providers. As new devices are introduced in the voice path, it is important to understand how the electrical characteristics of interfaces impact voice quality. This section discusses these electrical characteristics and how to fine-tune them for improved voice quality.

Electrical Characteristics

Voice signal power in a long-distance connection must be tightly controlled. The delivered signal power must be high enough to be clearly understood, but not so strong that it leads to instabilities such as echo. In the traditional telephony network, telephone companies control the signal power levels at each analog device. Now that the IP network is carrying voice, it might be necessary to adjust signal power on a voice interface to fine-tune the voice quality.

Echo

Most initial voice signals enter the network through a two-wire local loop. Most switches connect to other switches through a four-wire connection. As voice travels through the network for delivery to the remote telephone, the voice signal must be passed from the two-wire local loop to the four-wire connection at the first switch, and from the four-wire connection at the switch to a two-wire local loop at the remote end. If the impedance at

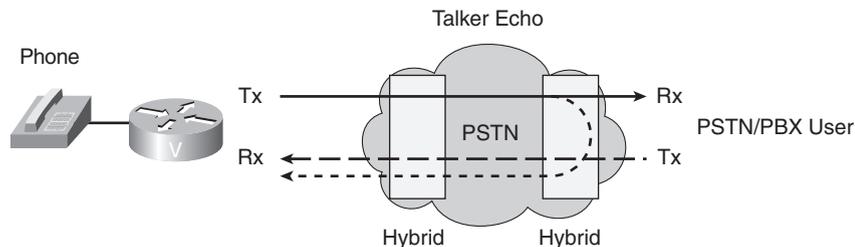
these two-wire to four-wire connections is not matched exactly, some of the voice signal reflects back in the direction of the source. As a result, originating callers hear their own voice reflected back. Sometimes, the reflected signal is reflected again, causing the destination to hear the same conversation twice.

In a traditional voice network, voice can reflect back. It usually goes unnoticed, however, because the delay is so low. In a VoIP network, echo is more noticeable because both packetization and compression contribute to delay. Specifically, for echo to be a problem, all of the following conditions must exist:

- An analog leakage path between analog Tx and Rx paths
- Sufficient delay in echo return for echo to be perceived as annoying
- Sufficient echo amplitude to be perceived as annoying

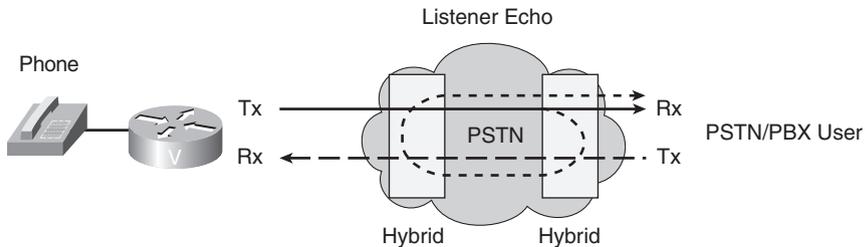
Two types of echo might exist in a telephony network: *talker echo* and *listener echo*. Talker echo, which is the most common type of echo, occurs when the speech energy of a talker, transmitted down the primary signal path, is coupled into the receiving path from the far end (or *tail circuit*), as illustrated in Figure 3-15. Talkers then hear their own voice, delayed by the total echo path delay time. If the “echoed” signal has sufficient amplitude and delay, the result can be annoying to the user and can interfere with the normal speech process. Talker echo is usually a direct result of the two-wire to four-wire conversion that takes place through “hybrid” transformers.

Figure 3-15 *Talker Echo*



Listener echo is less common than talker echo. As shown in Figure 3-16, listener echo occurs at the far end by circulating voice energy and is generally caused by the two-wire and four-wire hybrid transformers (caused by the “echo being echoed”). The voice of the talker is echoed by the far-end hybrid, and when the echo comes back to the listener, the hybrid on the side of the listener echoes the echo back toward the listener. The effect is that the person listening hears both the talker and an echo of the talker.

Figure 3-16 *Listener Echo*



Signal Strength

Another problem is inconsistent volume at different points in the network. Both echo and volume inconsistency might be caused by a voice port generating a signal level that is too high or too low. You can adjust signal strength, either in the inbound direction from an edge telephone or switch into the voice port, or in the outbound direction from the voice port to the edge telephone or switch. Echo results from incorrect input or output levels, or from an impedance mismatch. Although these adjustments are available on the Cisco voice equipment, they are also adjustable on PBX equipment.

Too much input gain can cause clipped or fuzzy voice quality. If the output level is too high at the remote router voice port, the local caller hears echo. If the local router voice port input decibel level is too high, the remote side hears clipping. If the local router voice port input decibel level is too low, or the remote router output level is too low, the remote-side voice can become distorted at a very low volume and DTMF tones might be missed.

Calculating Decibel Levels

Change in signal strength is measured in decibels (dBs). You can either boost the signal or attenuate it by configuring the voice port for input gain or output attenuation. You must be aware of what a voice port connects to and know at what dB level that device works best.

Calculating network dB levels is often an exercise in simple number line arithmetic. Table 3-5 provides common dB levels.

Table 3-5 *Calculating Decibel Levels*

Source 1 Out/In	Router 1 Adjustment	Net at Router 1	WAN	Net at Router 2	Router 2 Adjustment	Destination 1 In/Out
0 dB →	-3 dB →	-3 dB	--	-3 dB	+/-6 dB →	→ -9 dB
-9 dB ←	← +/- 6 dB	-3 dB	--	-3 dB	-3 dB	← 0 dB

Baselining Input and Output Power Levels

Considerations for baselining input and output power levels are as follows:

- Analog voice routers operate best when the receive level from an analog source is set at approximately -3 dB.
- In the United States and most of Europe, the receive level normally expected for an analog telephone is approximately -9 dB. In Asian and South American countries, receive levels are closer to -14 dB. To accommodate these differences, the output levels to the router are set over a wide range.
- Overdriving the circuit can cause analog clipping. Clipping occurs when the power level is above available pulse code modulation (PCM) codes, and a continuous repetition of the last PCM value is passed to the DSP.
- Echo occurs when impedance mismatches reflect power back to the source.

Adjustment of decibel levels might be necessary throughout a voice network. A station connected to a PBX might experience one level of loudness when calling a local extension, a different level when dialing an outside line, and different levels when calling remote sites via VoIP. Adjustments might be necessary in this case.

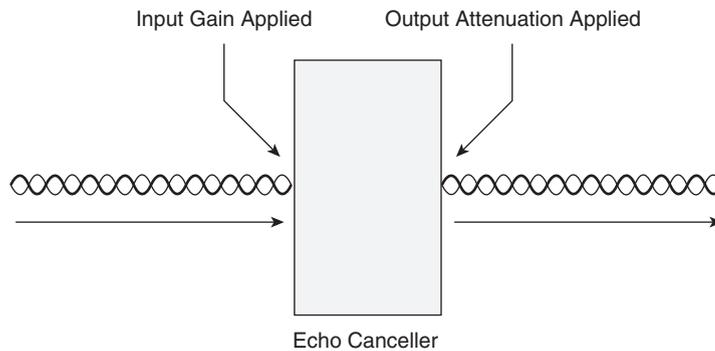
Voice Port Tuning

In an untuned network, a port configuration that delivers perceived good quality for a call between two dial peers might deliver perceived poor quality for a call between two other dial peers.

Voice quality adjustment is a defined, step-by-step procedure that is implemented after the network is up and running. It is ineffective for you to begin changing the default voice port configurations until full cross-network calls are established; a correctly implemented procedure results in a quality compromise between various sources that the customer accepts as good overall quality.

A variety of different factors, including input gain and output attenuation, can affect voice quality, as illustrated in Figure 3-17. Notice that *input gain* occurs as the signal goes into the echo canceller, and *output attenuation* occurs as the signal comes out of the echo canceller.

A loss plan looks at the required dB levels at specific interfaces, such as an analog FXS port connecting to a telephone or an FXO port connecting to the PSTN. An analog voice router works best with a receive level of -3 dB. An analog telephone in North America and Europe works best with a receive level of -9 dB. Therefore, if the device connecting to that router provides a different level than the expected -3 dB, then input gain can be set to equalize it to -3 dB. If the output at the other end is a telephone that expects -9 dB, then the output voice port has to provide -6 dB output attenuation in addition to the -3 dB to send signaling to the telephone at the expected -9 dB levels. A system-wide loss plan looks at the dB levels

Figure 3-17 *Configuring Voice Port Voice-Quality Tuning*

of the initial input and the remote output ports and plans for the appropriate adjustments for end-to-end signal levels. You must consider other equipment (including PBXs) in the system when creating a loss plan.

Voice Port Tuning Configuring Parameters

Parameters for configuring voice port voice quality tuning are as follows:

- **input-gain**—Configures a specific input gain, in decibels, to insert into the receiver side of the interface. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of -6 dB between telephones. The standard transmission plan defines country-specific dB levels and assumes that interfaces already provide the expected dB levels. For example, there must be attenuation of -6 dB between two telephones so that the input gain and output attenuation is 0, if the interfaces provide the required -6 dB attenuation.
- **output-attenuation**—Configures the output attenuation value in decibels for the transmit side. The value represents the amount of loss to be inserted at the transmit side of the interface.
- **impedance**—Configures the terminating impedance of a voice port interface. The impedance value selected must match the setting from the specific telephony system or device to which it is connected. Impedance standards vary between countries. CO switches in the United States are predominantly 600 ohms real (600r). PBXs in the United States are normally 600r or 900 ohms complex (900c).

Incorrect impedance settings or an impedance mismatch generates a significant amount of echo. You can mask the echo by entering the **echo-cancel** command. In addition, gains often do not work correctly if there is an impedance mismatch.

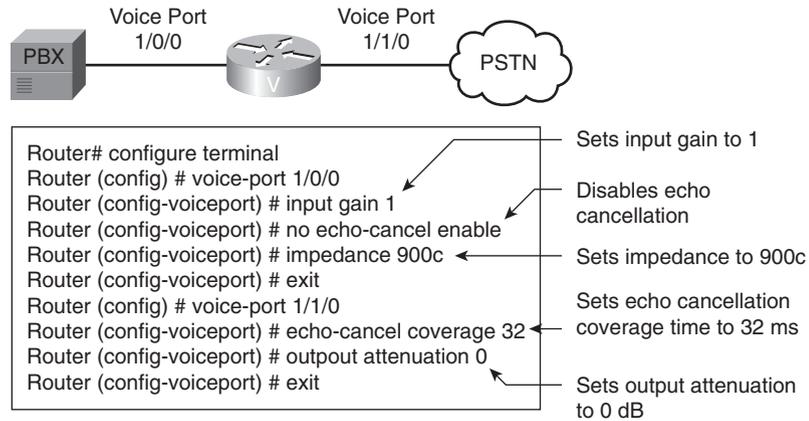
NOTE

The **input-gain** and **output-attenuation** commands accommodate network equipment and are not end-user volume controls for user comfort.

Configuring Voice Port Tuning

Figure 3-18 shows a voice-enabled router connecting to a PBX via an E&M port, specifically, port 1/0/0. Additionally, the router connects into the PSTN via FXO port 1/1/0.

Figure 3-18 *Voice Port Tuning Example*



Example 3-8 illustrates voice port tuning parameters on the E&M and FXO ports in the router shown in Figure 3-18. In the example, the PBX output is -4 dB, whereas the voice router functions best at -3 dB. Therefore, the adjustment is made in the inbound path to the router using the **input-gain** command. The impedance setting on the router needs to be changed from the default of 600r to match the 900c impedance setting of the PBX. Because this is an E&M port, echo cancellation is disabled. The FXO port connecting to the PSTN has an adjustment for echo coverage that allows for longer-distance echo cancellation.

Example 3-8 *Configuring Voice Port Tuning*

```

Router#configure terminal
Router(config)#voice-port 1/0/0
Router (config-voiceport)#input gain 1
!Sets input gain to 1
Router (config-voiceport)#no echo-cancel enable
!Disables echo cancellation
Router (config-voiceport)#impedance 900c
!Sets impedance to 900c
Router (config-voiceport)#exit
Router(config)#voice-port 1/1/0
Router (config-voiceport)#output attenuation 0
!Sets output attenuation to 0 decibels
Router (config-voiceport)#exit
  
```

E&M voice port parameters used in Example 3-8 include:

- **input-gain**—Increases the inbound voice level by 1 dB before the voice is transmitted across the network
- **no echo-cancel enable**—Disables echo cancellation
- **impedance**—Sets the impedance to match the connecting hardware

FXO voice port parameters used in Example 3-8 include:

- **echo-cancel coverage**—Adjusts the cancellation coverage time to 32 ms. This allows for cancellation of echo that has greater delay.
- **output-attenuation**—Specifies that there is no attenuation as the signal is passed out of the interface to the PSTN.

Configuring Echo Cancellation

Echo cancellation is configured at the voice port level. It is enabled by default, and its characteristics are configurable. Echo cancellation commands are as follows:

- **echo-cancel enable**—Enables cancellation of voice that is sent out through the interface and received back on the same interface. Sound that is received back in this manner is perceived by the listener as echo. Echo cancellation keeps a certain-sized sample of the outbound voice and calculates what that same signal looks like when it returns as an echo. Echo cancellation then attenuates the inbound signal by that amount to cancel the echo signal. If you disable echo cancellation, it will cause the remote side of a connection to hear echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, you should disable this command if it is not needed. There is no echo path for a four-wire E&M interface. The echo canceller should be disabled for this interface type.
- **echo-cancel coverage**—Adjusts the coverage size of the echo canceller. This command enables cancellation of voice that is sent out through the interface and received back on the same interface within the configured amount of time. If the *local loop* (the distance from the interface to the connected equipment that is producing the echo) is longer, the configured value of this command should be extended.

If you configure a longer value for this command, it takes the echo canceller longer to converge. In this case, the user might hear a slight echo when the connection is initially set up. If the configured value for this command is too short, the user might hear some echo for the duration of the call because the echo canceller is not canceling the longer-delay echoes. There is no echo or echo cancellation on the network side (for example, the non-POTS side of the connection).

- **non-linear**—The function enabled by the **non-linear** command is also known as residual echo suppression. This command effectively creates a half-duplex voice path. If voice is present on the inbound path, then there is no signal on the outbound path. This command is associated with the echo canceller operation. The **echo-cancel enable** command must be enabled for the **non-linear** command to take effect. Use the **non-linear** command to shut off any signal if near-end speech is not detected.

Enabling the **non-linear** command normally improves performance. However, some users encounter truncation of consonants at the ends of sentences when this command is enabled. This occurs when one person is speaking and the other person starts to speak before the first person finishes. Because the nonlinear cancellation allows speech in one direction only, it must switch directions on the fly. This might clip the end of the sentence spoken by the first person or the beginning of the sentence spoken by the second person.

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

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

ITU standard G.168 allows more rigorous testing and satisfies more testing requirements. White noise is replaced with a pseudo-speech signal for the convergence tests. Most echo cancellation algorithms use a least mean square algorithm to adapt the echo cancellation filter. This algorithm works best with random signals and slows down with more correlated signals such as speech. Use of the pseudo-speech signal in testing provides a more realistic portrayal of the echo canceller's performance in real use.

If you speak into your telephone and hear your own voice a short time later, you are experiencing talker echo. As you learned earlier in the “Echo” section, talker echo is caused by the remote telephony circuitry's two-wire to four-wire hybrid circuit. Enabling echo-cancellation on your voice port will eliminate the problem. Depending on the return time of the echoed voice, you can further adjust using the **echo-cancel coverage** command. Table 3-6 compares echo cancellation standards.

Table 3-6 *Comparing Echo Cancellation Standards*

	G.165 EC	G.168 EC
Tail Coverage	Up to 32 ms	Up to 64 ms
Minimum Echo Return Loss (ERL)	Greater than or equal to -6 dB	Configurable to greater than or equal to -0 dB, -3 dB, or -6 dB
Echo Suppression	Up to 10 seconds	Not required due to faster convergence
Minimum Cisco IOS Software Release	12.2(11)T, 12.2(8)T5, 12.2(12), and higher	12.2(13)T, 12.2(8)YN, 12.2(15)T, 12.3(4)T, 12.3(4)XD, and higher

Summary

This chapter discussed the configuration of various analog and digital voice ports. For example, analog voice ports, such as FXS and FXO ports, need an appropriate signaling type (that is, loop-start or ground-start) to communicate with attached equipment. E&M port configurations also need to match the configurations of the equipment connecting to the E&M ports (for example, a PBX E&M interface). This chapter also reviewed various analog port timing parameters.

Digital port configuration included parameters such as line coding and framing. Additionally, this chapter introduced how DS-0 groups logically combine multiple channels in a digital circuit (for example, a T1), thus reducing the number of required configuration steps. Several **show**, **debug**, and **test** commands assist in the monitoring and troubleshooting of voice ports.

Echo often causes significant quality issues in VoIP networks. Typically, echo results from an impedance mismatch in a two-wire to four-wire hybrid circuit. Fortunately, Cisco voice-enabled routers combat echo via echo suppression and echo cancellation technologies. This chapter explored the commands used to influence the effect of echo on voice calls.

Chapter Review Questions

The following questions test your knowledge of topics explained in this chapter. You can find the answers to the questions in Appendix A, “Answers to Chapter Review Questions.”

1. If a client picked up a customer service handset and was automatically connected to customer service without dialing any digits, what kind of call would it be?
 - a Cisco CallManager-to-Cisco CallManager call
 - b PBX-to-PBX call
 - c on-net call

- d local call
 - e PLAR call
2. Which configuration parameter would you change to set the dial tone, busy tone, and ringback tone on an FXS port?
- a cptone
 - b ring frequency
 - c ring cadence
 - d description
 - e signal
3. What are the two options for the **linecode** command on an E1 connection?
- a SF
 - b ESF
 - c CRC4
 - d AMI
 - e B8ZS
 - f HDB3
4. What are the two options for the **framing** command on a T1 connection?
- a SF
 - b ESF
 - c CRC4
 - d AMI
 - e B8ZS
 - f HDB3
5. What is the purpose of T-CCS?
- a To route calls between PBXs

- b** To provide point-to-point connections between PBXs
 - c** To pass proprietary signaling between PBXs
 - d** To specify a channel for standards-based call signaling
- 6.** What is the best time to change default voice port configurations to tune voice quality?
 - a** Before you set up the network
 - b** After the network is up and running
 - c** After two dial peers experience poor quality
 - d** When there is a network failure
- 7.** Which command is used to enable residual echo suppression?
 - a** **echo-cancel enable**
 - b** **echo-cancel coverage**
 - c** **non-linear**
 - d** **no echo-cancel enable**
- 8.** Which two of the following are echo cancellation standards?
 - a** G.114
 - b** G.165
 - c** G.168
 - d** G.400
- 9.** Which of the following statements best describes when echo occurs?
 - a** When there is a mismatch in signaling types
 - b** When impedance mismatches reflect power back to the source
 - c** When the far-end impedance absorbs the signal
 - d** When the router processor utilization is too high to run the companding algorithm in a timely manner
- 10.** Why does echo usually go unnoticed in a traditional telephony network?
 - a** The volume of the echo is high.

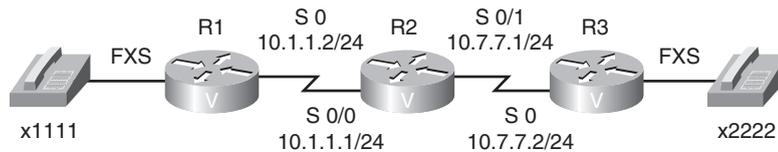
- b The delay between the original voice signal and the echo is low.
- c Traditional telephony networks do not suffer from impedance mismatches.
- d Traditional telephony networks negate inductance through the use of load coils.

Lab Exercise: Voice Port Configuration

In this lab, you will configure and test router R1's FXS port using the topology you constructed in the lab exercise for Chapter 2, "Analog and Digital Voice Connections." Recall that an FXS port allows you to connect end stations, such as analog phones, fax machines, and modems, into a voice-enabled router.

Figure 3-19 illustrates the lab's physical topology and IP addressing.

Figure 3-19 *Lab Topology*



Task 1: Configure FXS Port Parameters

In this task, you will issue commands to configure various parameters on an FXS port.

Complete these steps:

1. Enter voice port configuration mode for the FXS port on router R1.
2. To ensure the appropriate signaling type on this voice port, configure the signaling to the default value of **loopstart**.
3. Assume the analog phone connected to this voice port is located in one of several adjoining cubicles and you wish to create a distinctive ring for this phone so that the phone's user can distinguish their phone's ringing pattern from other phones' ringing patterns. Create a distinctive ringing pattern that causes the analog phone to ring for 5 seconds, be silent for 1 second, and then repeat this pattern. To make sure this change takes effect, shut down and then bring up the voice port, using the **shutdown** command, followed by the **no shutdown** command.
4. Assume that several thousand feet of cable separate the analog phone from the FXS port, and as a result, some electrical capacitance has built up in the line. This capacitance can lead to an impedance mismatch, thus causing echo. To help

counteract this capacitance, change the impedance value of the voice port from the default value of 600r (that is, 600 purely resistive ohms) to 600c (that is, 600 ohms of impedance that contains both a resistive and a capacitive component).

NOTE At this point, you have not configured your voice-enabled routers with call routing intelligence. Therefore, you cannot place a call between the two analog phones. However, the parameters you configured in this lab will be in effect at the completion of the lab exercise for Chapter 4.

Task 2: Exercise Verification

In this task, you will verify the parameters you configured are in effect and that the FXS port is functioning properly.

Complete these steps:

1. To verify that the appropriate signaling, ring cadence, and impedance settings are applied to R1's FXS port, use the **show voice port *port_id*** command to view the port settings.
2. To verify that the voice port is functioning, lift the handset of the attached analog phone. You should hear dial tone. If you do not hear dial tone, you might need to administratively bring up the port with the **no shutdown** command.
3. To verify that the voice port can send a ringing signal at an appropriate voltage and frequency as to alert the attached analog phone, enter the **test voice port *port_id* relay ring on** command, followed by the **test voice port *port_id* relay ring disable** command.

Suggested Solution

Although your physical hardware might differ, Example 3-9 offers one solution to the preceding exercise.

Example 3-9 Router R1's Configuration

```

R1#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
R1(config)#voice-port 1/1
R1(config-voiceport)#signal loopstart
R1(config-voiceport)#ring cadence define 50 10
R1(config-voiceport)#shutdown
R1(config-voiceport)#no shutdown
R1(config-voiceport)#impedance 600c
R1(config-voiceport)#end

```

The results of the **show voice port** *port_id* command, issued on router R1, should be similar to the output shown in Example 3-10. Note that the shaded lines in the output reflect the parameters configured in this exercise

Example 3-10 Router R1's show voice port 1/1 Output

```

R1#show voice port 1/1
FXS 1/1 Slot is 1, Port is 1
  Type of VoicePort is FXS
  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
  Non Linear Mute is disabled
  Non Linear Threshold is -21 dB
  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 Cancellation NLP mute is disabled
  Echo Cancellation NLP threshold is -21 dB
  Echo Cancel Coverage is set to 64 ms
  Echo Cancel worst case ERL is set to 6 dB
  Playout-delay Mode is set to adaptive
  Playout-delay Nominal is set to 60 ms
  Playout-delay Maximum is set to 200 ms
  Playout-delay Minimum mode is set to default, value 40 ms
  Playout-delay Fax is set to 300 ms
  Connection Mode is normal
  Connection Number is not set
  Initial Time Out is set to 10 s
  Interdigit Time Out is set to 10 s
  Call Disconnect Time Out is set to 60 s
  Supervisory Disconnect Time Out is set to 750 ms
  Ringing Time Out is set to 180 s
  Wait Release Time Out is set to 30 s
  Companding Type is u-law

```

continues

Example 3-10 Router R1's show voice port 1/1 Output (Continued)

```
Coder Type is g729ar8
Voice Activity Detection is enabled
Nominal Playout Delay is 60 milliseconds
Maximum Playout Delay is 200 milliseconds
Region Tone is set for US

Analog Info Follows:
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600c Ohm
Analog interface A-D gain offset = -3.0 dB
Analog interface D-A gain offset = -3.0 dB
FXS idle voltage set to low
Station name None, Station number None
Translation profile (Incoming):
Translation profile (Outgoing):

Voice card specific Info Follows:
Signal Type is loopStart
Ring Frequency is 20 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is active
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
Hookflash-in Timing is set to 1000 ms
Hookflash-out Timing is set to 400 ms
No disconnect acknowledge
Ring Cadence is User defined
Ring Cadence are [50 10] * 100 msec
Ringer Equivalence Number is set to 1
InterDigit Pulse Duration Timing is set to 500 ms
```