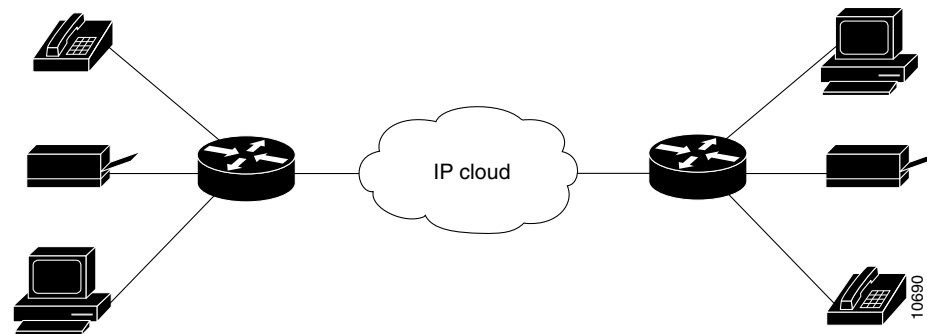




Voice-over-IP Quick Start Guide

Voice-over-IP (VoIP) enables a Cisco 1750 router (hereafter referred to as the router) to carry voice traffic (for example, telephone calls and faxes) over an IP network, simultaneously with data traffic. (See Figure 1.) This guide describes the configuration tools and the major configuration tasks to configure the router.

Figure 1 Voice and Data Traffic on an IP Network



Configuration Tools

You can configure VoIP by using one of the following tools:

- Cisco ConfigMaker application
- Cisco IOS command-line interface (CLI)

Cisco ConfigMaker Application

Use the Cisco ConfigMaker application to address and configure one or more devices or to create an entire network. A series of wizards and dialog boxes can make your voice-configuration task simple and fast.



Follow these steps to install the Cisco ConfigMaker application on your PC:

-
- Step 1** Disable any virus-scanning software you have running.
- Step 2** Close all programs active on your PC.
- Step 3** Insert the CD into your CD-ROM drive.
- Step 4** The Cisco ConfigMaker installer starts automatically.
If the installer does not start after a few seconds, select **Start>Run**, and enter **d:\setup** where **d** is your CD-ROM drive.
- Step 5** Follow the instructions on the screen to install the Cisco ConfigMaker application.
To learn about the Cisco ConfigMaker application, view the online movie, or use *Cisco ConfigMaker Help Topics* from the help menu.
You need administrative privileges to install the Cisco ConfigMaker application on Windows NT.
-

Command-Line Interface

To configure your router from the command-line interface (CLI), refer to this guide, the *Cisco 1700 Series Software Configuration Guide*, or the *Cisco 1750 Router Voice-over-IP Configuration Guide* available on the Documentation CD-ROM.

Major Configuration Tasks

Follow these major steps to configure VoIP on your router:

-
- Step 1** Configure your IP network to support real-time voice traffic. Before you move to the next step, ping the remote router to make sure you have an IP connection.
- Step 2** Configure your router to support voice ports. Follow these steps for your specific requirements:
- Foreign exchange station (FXS) and foreign exchange office (FXO) voice ports
The default voice-port values are usually adequate for both FXS and FXO voice ports. However, if you need to change the default values, refer to the *Cisco 1750 Router Voice-Over-IP Configuration Guide*.
 - E&M voice port
The default values for an E&M voice port are not sufficient to enable voice and data transmission over the IP network. E&M values must match those of the connected PBX device. Refer to the “Connecting VICs to the Network” section on page 6.
- Step 3** Enter configuration mode to configure your router for VoIP traffic. Refer to the “Entering Configuration Mode” section on page 10.
- Step 4** Save the configuration. Refer to the “Saving the Configuration” section on page 11.

- Step 5** Establish a dial plan to determine how the call can be routed. Use the **dial-peer voice** command to define dial peers and to implement the dial plan. Refer to the “Dial Plan” section on page 12.
- Step 6** Fine-tune your network to adequately support VoIP by configuring a series of protocols and features for quality of service (QoS). Refer to the “Tuning Parameters for Real-Time Voice Traffic” section on page 26.
-

Information for International Users

International users must set the values of certain voice-port commands, like **cptone**, that are specific to their country. Refer to the *Cisco 1750 Router Voice-over-IP Configuration Guide* for details.

Organization

In the following sections, this guide explains how to install the voice interface cards (VICs) and how to set up basic configurations for a VoIP network:

- “Audience” section on page 4 describes what you should know before starting, what you should do before configuring your voice network, conventions used in this guide, and how to find more information.
- “Installing Voice Interface Cards” section on page 5 describes the VICs and explains how to install them.
- “Voice Port Numbering” section on page 8 explains the interface numbering convention for voice ports.
- “Entering Configuration Mode” section on page 10 offers hints for using Cisco IOS software to configure a VoIP network.
- “Saving the Configuration” section on page 11 explains how to save the configuration.
- “Dial Plan” section on page 12 explains how to map the destination telephone numbers with the voice ports on the router, and describes two types of dial peers.
- “Configuring FXS Interfaces” section on page 14 describes how to configure FXS VICs for connecting to a telephone, fax machine, and similar device.
- “Configuring FXO Interfaces” section on page 21 describes how to configure FXO VICs for connecting to a PSTN central office.
- “Configuring E&M Interfaces” section on page 23 describes how to configure E&M VICs for connecting to a PBX.
- “Tuning Parameters for Real-Time Voice Traffic” section on page 26 describes how to improve the QoS in VoIP connections.
- “List of Terms” section on page 31 provides a list of terms and abbreviations used in this guide.
- “Cisco Connection Online” section on page 33 explains how to get service and support.

For further information on VoIP features, refer to the *Cisco 1750 Router Voice-over-IP Configuration Guide*, the *Cisco WAN Interface Cards Hardware Installation Guide*, and the Cisco IOS configuration guides and command references.

Audience

This guide is intended primarily for users who configure and maintain routers, but are not necessarily familiar with tasks, the relationship between tasks, or the commands necessary to perform particular tasks to configure VoIP. We assume that you are familiar with Cisco IOS software, IP and telephony networks, and PBX operation for E&M installations.

Conventions

This document uses the following conventions:

- Commands and keywords are in **boldface**.
- Optional keywords or arguments are in square brackets.
- Variables for which you supply values are in *italic*.
- Titles of publications and new words or concepts are in *italic*.
- Information the router displays on the console screen is in `screen font`.
- Information that you enter at the CLI is in **boldface screen font**.

Additional Information

Cisco documentation and additional literature are available in a CD-ROM package, which ships with your product. The Documentation CD-ROM, a member of the Cisco Connection Family, is updated monthly. Therefore, it might be more up to date than printed documentation. To order additional copies of the Documentation CD-ROM, contact your local sales representative or call customer service. The CD-ROM package is available as a single package or as an annual subscription.

You can also read Cisco documentation on the World Wide Web at <http://www.cisco.com>, <http://www-china.cisco.com>, or <http://www-europe.cisco.com>. From here, you can submit comments electronically. Click **Feedback** on the title bar, and then select **Documentation**. After you complete the form, click **Submit** to send it to Cisco. We appreciate your comments.

If you have questions or need help, refer to the section “Cisco Connection Online” at the end of this guide for further information.

Installing Voice Interface Cards

The router supports one to three Cisco voice interface cards (VICs). Each VIC provides two ports. You need one VIC port for each voice connection.



Note

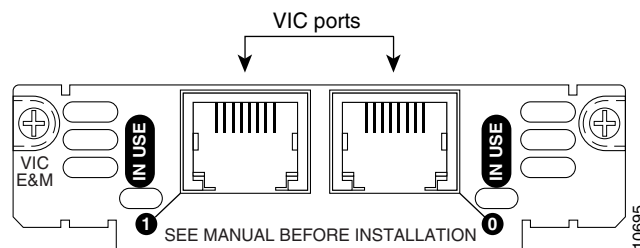
To transmit voice calls over an IP WAN, you need (in addition to the VICs) at least one WAN interface card (WIC) in the router to provide the connection to the WAN.

There are three types of VIC interfaces. (See Figure 4.)

- FXS (foreign exchange station) interface connects directly to a standard telephone, fax machine, or similar device. The FXS VIC interface supplies ringing voltage, dial tone, and similar signals to the station. Ports on this VIC are color-coded gray.
- FXO (foreign exchange office) interface connects local calls to a PSTN central office or to a PBX that does not support E&M signaling. This is the interface a standard telephone provides. Ports on this VIC are color-coded pink.
- E&M is a signaling technique for two-wire and four-wire telephone and trunk interfaces. The E&M VIC connects remote calls from an IP network to a PBX for local distribution. Ports on this VIC are color-coded brown.

Figure 2 shows a typical VIC.

Figure 2 Voice Interface Card



You should install and cable VICs before performing the software configuration tasks explained later in this guide.



Warning

Be sure to observe all warnings and safety precautions in the installation guide.



Caution

VICs do not support online insertion and removal (hot swap). To avoid equipment damage, turn off electrical power and disconnect network cables before you insert a VIC in the router.

Follow these steps, and refer to Figure 3 to insert a VIC in the router:

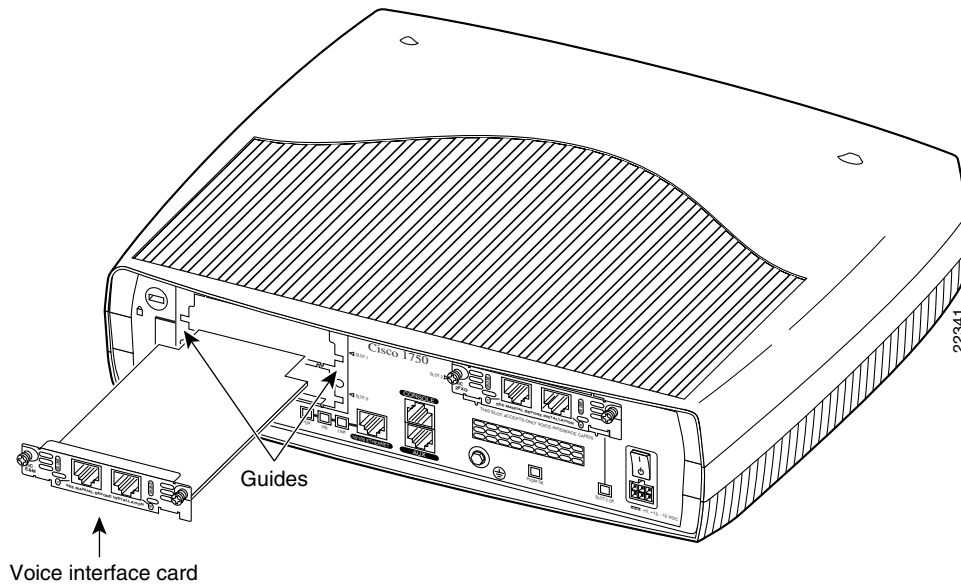
Step 1 Make sure the router is turned off and is disconnected from the power supply.

Step 2 Loosen the thumbscrews on the VIC slot cover on the rear panel.

You should be able to loosen the screws using your fingers; however, if the screws are very tight, you might need to use a Phillips screwdriver.

- Step 3** Remove the metal plate that covers the card slot.
- Step 4** Hold the card by the edges on either side of the card front panel, and line up the card edges with the guides inside the card slot, as shown in Figure 3.
- Step 5** Insert the card in the slot and gently push it into the router until the front panel of the card is flush with the rear panel of the router.
- Step 6** Tighten the screws.

Figure 3 Inserting a VIC in the Router

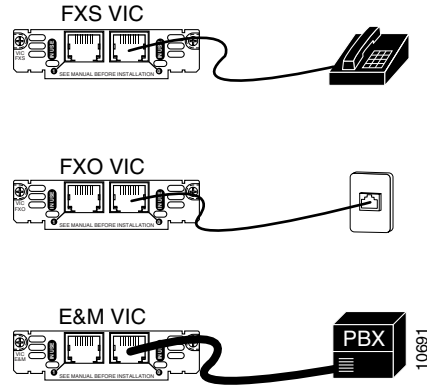


If you need more detailed installation instructions, refer to the *Cisco WAN Interface Cards Hardware Installation Guide*.

Connecting VICs to the Network

Use the following cables, and refer to Figure 4 to connect the VICs to the network:

- Standard RJ-11 modular telephone cable to connect FXS VIC ports (color-coded gray) to a telephone or fax machine.
- Standard RJ-11 modular telephone cable to connect FXO VIC ports (color-coded pink) to the PSTN, or to a PBX that does not support E&M signaling, through a telephone wall outlet.
- Standard RJ-48S connector and cable to connect E&M VIC ports (color-coded brown) to a PBX line. The cable wiring depends on the type of connection. For details, refer to the *Cisco WAN Interface Cards Hardware Installation Guide*.

Figure 4 Connecting VICs to the Network

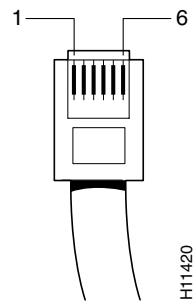
When you are finished, reinstall any network interface cables you removed, and turn on power to the router.

Connector Pinouts

Figure 5 shows the wiring patterns for FXS and FXO VICs connector, and Table 1 lists the pinouts.


Note

Pins that are not used should not be connected.

Figure 5 RJ-11 Wiring**Table 1** FXS and FXO Pinouts

Pin	Signal
1	–
2	–
3	Ring
4	Tip
5	–
6	–

Figure 6 shows the wiring pattern for an E&M connector, and Table 1 lists the pinout.

**Note**

The E&M VIC pinout depends on the PBX type and connection. Pins that are not used should not be connected.

Figure 6 RJ-48S Wiring

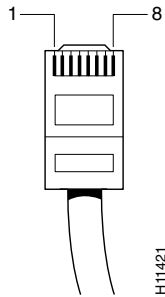


Table 2 E&M Pinouts

Pin	Signal	Description
1	SB	-48V signaling battery
2	M-lead	Signaling input
3	R	Ring, audio input
4	R or R1	Ring, audio input/output, or output
5	T or T1	Tip, audio input/output, or output
6	T	Tip, audio input
7	E-lead	Signaling output
8	SG	Signaling ground return

Checking the Installation

If you installed an FXS VIC, connect a handset to the VIC port. When router power is on, you should be able to hear the dial tone when you lift the handset. The dial tone should stop after you dial a digit. If you have trouble, use the **show voice port** command to make sure that the VIC is installed correctly, or try connecting a different handset to the VIC.

Voice Port Numbering

Enter the **show voice port** command to identify the port numbers of voice interfaces installed in your router:

```
Router# show voice port slot-number/port
```

Figure 7 shows Cisco 1750 router slot numbering.

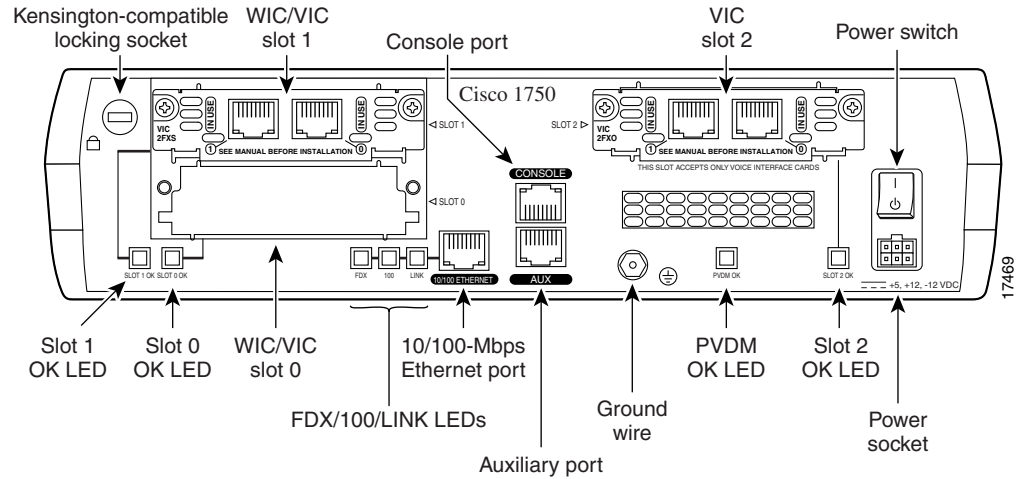
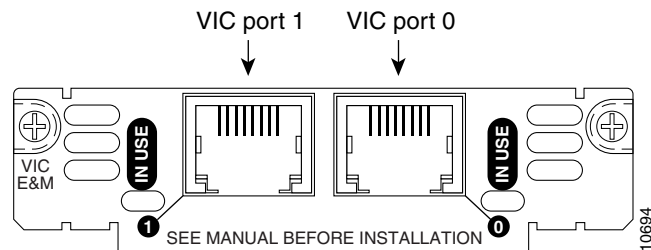
Figure 7 Cisco 1750 Router Slot Numbering

Figure 8 shows the VIC port numbering.

Figure 8 VIC Port Numbering

Example

If you install a VIC in both SLOT 0 and SLOT 1 of the router, the ports in each of these slots would be numbered as follows:

SLOT 0—0/0 and 0/1

SLOT 1—1/0 and 1/1

Entering Configuration Mode

You can configure your router for VoIP by entering commands at the CLI. This method of entering commands is called configuration mode.



Note

VoIP requires one of the following IOS Release 11.3(1)T (or later) feature sets: IP Plus, Desktop Plus, or Enterprise Plus.

Follow these steps to enter configuration mode:

Step 1

Connect a terminal or PC to the router.

To configure the router with a PC, the PC must have some type of terminal emulation software installed. The software should be configured with the following parameters: 9600 baud, 8 data bits, no parity, 1 stop bit.

Step 2

Power up the router.

Step 3

If the current configuration is no longer valid (for instance, because you added an interface), after about 1 minute you will see the following prompt:

```
Would you like to enter the initial configuration dialog? [yes]:
```

Do not use the **yes** default. You need to enter **no** to enter the normal operating mode of the router.



Note

If the current configuration is valid, you enter the normal operating mode automatically.

Step 4

At the user EXEC prompt (`Router>`), enter **enable** and the password to enter enable mode:

```
Router> enable
Password:
```

Configuration changes can be made only in enable mode. The prompt changes to the privileged EXEC (enable) prompt (`Router#`):

```
Router#
```

Step 5

Enter the following command to enter global configuration mode:

```
Router# config terminal
Router(config)#
```

Step 6

If you have not configured the router before or want to change the configuration, you need to configure global parameters, passwords, network management, and routing protocols. In this example, IP routing, AppleTalk routing, and IPX routing are all enabled:

```
Router(config)# ip routing
Router(config)# appletalk routing
Router(config)# ipx routing
```

For complete information about the global configuration commands and about configuring LAN and WAN interfaces on your router, refer to the Cisco IOS configuration guides and command references.

The rest of this guide explains how to configure your router for VoIP traffic.

**Note**

Voice configuration uses a number of Cisco IOS commands. For complete information about these commands, refer to the *Cisco 1750 Router Voice-Over-IP Configuration Guide*. You can also enter a question mark after a command or partial command at the `Router(config)#` prompt to get syntax and argument help.

**Note**

At any point, you can see the operating configuration, including changes you just made, by entering the **show running-config** command. To enter this command, you have to exit from configuration mode by pressing **Ctrl-Z** or entering the **end** global configuration command.

Saving the Configuration

You must save the new configuration to nonvolatile RAM (NVRAM) periodically during the configuration process and also when you finish configuring the router.

Cisco IOS software uses two configuration files—startup config and running config. The startup-config file is a backup file that has all the information you specified about the router interfaces and is used by the router on the next restart. The running-config file has the current operating configuration with the changes you have just made. You can make any changes to the running-config file; however, these changes will be lost when the router powers down. To make these changes permanent, copy the running-config file to the startup-config file stored in NVRAM.

Follow these steps for each router to write the new configuration to NVRAM:

-
- Step 1** Exit configuration mode, and return to the enable prompt by pressing **Ctrl-Z** or entering the **end** global configuration command.
- Step 2** To see the startup configuration stored in NVRAM, enter the following command at the enable prompt:
- ```
Router# show startup-config
```
- Step 3** To see the operating configuration, including any changes you have just made, enter the following command:
- ```
Router# show running-config
```
- Step 4** To write your changes to NVRAM and make them permanent, enter the following command:
- ```
Router# copy running-config startup-config
Building configuration...
[OK]
Router#
```

The router is now configured to boot in the new configuration.

If you have questions or need help, refer to the last section, “Cisco Connection Online.”

---

# Dial Plan

Use a dial plan to map the destination telephone numbers with the voice ports on the router. In North America, the North American Numbering Plan (NANP) is used, which consists of an area code, an office code, and a station code. Area codes are assigned geographically, office codes are assigned to specific switches, and station codes identify a specific port on that switch. The format in North America is 1Nxx-Nxx-xxxx, with N = digits 2 through 9 and x = digits 0 through 9. Internationally, each country is assigned a one- to three-digit country code; the country's dialing plan follows the country code.

In corporate environments, VoIP can be configured to recognize extension numbers and expand them into their full E.164 dialed number by using two commands in tandem: **destination-pattern** and **num-exp**. Before you configure these two commands, map individual telephone extensions with their full E.164 dialed numbers. This can be done easily by creating a number expansion table.

For Cisco voice implementations, two types of dial peers are used to match a dialed number to either a local telephony port or a remote IP address:

- POTS dial peer associates a physical voice port with a local telephone device, and the key commands you need to configure are the **port** and **destination-pattern** commands as shown in the following example:

```
West(config)# dial-peer voice 401 pots
West(config-dial-peer)# destination-pattern 14085553737
West(config-dial-peer)# port 0/0
```

The **destination-pattern** command defines the telephone number associated with the POTS dial peer. The **port** command associates the POTS dial peer with a specific logical dial interface, normally the voice port connecting your router to the local POTS network.

- VoIP dial peer associates a telephone number with an IP address, and the key commands you need to configure are the **destination-pattern** and **session target** commands as shown in the following example:

```
West(config)# dial-peer voice 501 voip
West(config-dial-peer)# destination-pattern 1919555....
West(config-dial-peer)# session target ipv4:192.168.11.3
```

The **destination-pattern** command defines the telephone number associated with the VoIP dial peer. The **session target** command specifies a destination IP address for the VoIP dial peer.

Use the **dial-peer voice** command to define dial peers and change to dial peer configuration mode. To see an example, refer to the individual sections on configuring FXS, FXO, and E&M interfaces later in this guide.

## Wildcards and Number Expansion

Office PBXs are configured so that a user can dial a local call (within the same PBX) by dialing the extension only—for instance, the four-digit extension 3737 or the five-digit extension 53737 instead of the full telephone number, 1 408 555-3737.

You can provide the same shortcut on a VoIP network by using the **number-expansion (num-exp)** command.

```
East(config)# num-exp 5.... 1408555....
```

This command tells the router to expand a particular sequence of dialed numbers into a complete telephone number (destination pattern) as shown in Table 3.

**Table 3 Sample Number Expansion Table**

| Extension | Destination Pattern | Num-Exp Command Entry     | Description                                                                                           |
|-----------|---------------------|---------------------------|-------------------------------------------------------------------------------------------------------|
| 3737      | 14085553737         | num-exp 3737 14085553737  | To expand 3737 to 14085553737                                                                         |
| ....      | 1408555....         | num-exp .... 1408555....  | To expand any dialed sequence of four digits (represented by four periods) by prefixing 1408555 to it |
| 5....     | 1408555....         | num-exp 5.... 1408555.... | To expand a five-digit extension beginning with the numeral 5 by prefixing 140855 to it               |

You can use a period (.) as a wildcard character representing a single digit in a telephone number.

You can use the **show num-exp** command to verify the number expansion information.

```
East# show num-exp [dialed-number]
```

After you have configured dial peers and assigned destination patterns to them, you can use the **show dialplan number** command to see how a telephone number maps to a dial peer.

```
East# show dialplan number [dialed-number]
```

**Note**

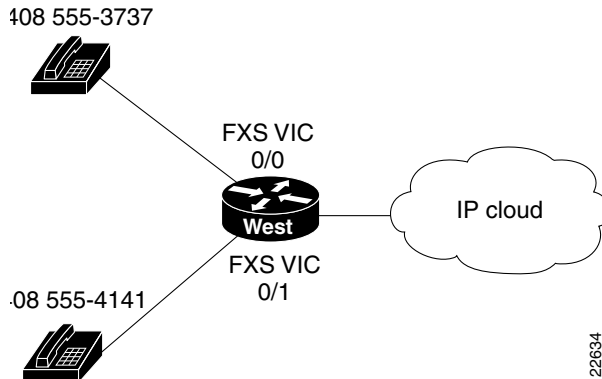
You must still configure each telephone number in full on a local dial peer so that the router can find the voice port it belongs to.

# Configuring FXS Interfaces

This section explains how to configure ports on FXS VICs that connect directly to a standard telephone, fax machine, or similar device.

Figure 9 shows a basic voice network. A small business uses a Cisco 1750 router (named *West*) to provide telephone and fax connections among employees in its office. Two of these telephones are connected to an FXS VIC port in the *West* router.

**Figure 9 Basic Voice Network (West Router)**



**Note**

You can name your router by using the global configuration **hostname** command.

(For information about port numbering, refer to the section on “Voice Port Numbering” earlier in this guide.)

**Table 4 West Router Telephone Numbers and Voice Ports**

| Telephone Number | Voice Port |
|------------------|------------|
| 408 555-3737     | 0/0        |
| 408 555-4141     | 0/1        |

**Note**

If your router is configured with three VICs, a total of six telephones and fax machines can be connected to it. As the router has only three slots, you need to replace one VIC with a WIC to provide an interface for IP connectivity to the WAN and for data traffic. To accommodate more than six voice devices, you need to add more routers or use an E&M VIC and a local PBX, rather than connecting every telephone to its own FXS VIC.

## Local Dial Peers

To route a received voice call to the right destination, the router needs to know which telephone number belongs to each voice port. For instance, if a call comes in for 408 555-3737, the router needs to know that this telephone is connected to voice port 0/0 (as shown in Figure 9.) In other words, the router needs to know the information in Table 4.

To hold this information, Cisco IOS software uses objects called *dial peers*. A telephone number, a voice port, and other call parameters are tied together by associating them all with the same dial peer. Configuring dial peers is similar to configuring static IP routes—you are telling the router what path to follow to route the call. All voice technologies use dial peers to define the characteristics associated with a call leg. A call leg is a segment of a call path, for instance, between a telephone and a router, a router and a network, a router and a PBX, or a router and the PSTN. Each call leg corresponds to a dial peer.

Dial peers are identified by numbers, but they are usually referred to as *tags* to avoid confusion with telephone numbers. Dial-peer tags are arbitrary integers that can range from 1 to  $2^{31}-1$  (2147483647). Within the allowed range, you can choose any dial-peer tag that is convenient or makes sense to you. Dial peers on the same router must have unique tags, but you can reuse the tags on other routers.

Table 4 assigns a dial-peer tag to each telephone number and its associated voice port on the *West* router. This type of dial peer is called a *POTS dial peer* or a *local dial peer*. The term “POTS” (plain old telephone service) means that the dial peer associates a physical voice port with a local telephone device. (VoIP dial peer is explained in the section “Calling Between Routers” later in this guide.)

**Table 5** *West Router Local Dial Peers*

| Telephone Number | Voice Port | Dial-Peer Tag |
|------------------|------------|---------------|
| 408 555-3737     | 0/0        | 401           |
| 408 555-4141     | 0/1        | 402           |

You should construct a table similar to Table 4 for your own routers, assigning your own telephone numbers and dial-peer tags.



**Note**

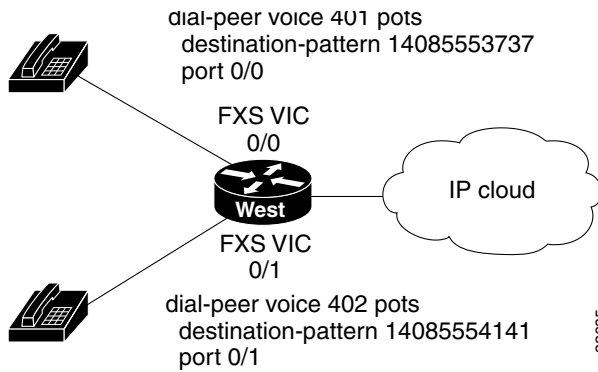
The telephone numbers used in this guide are only examples and are invalid for public use in the United States. When you configure your network, be sure to substitute your own telephone numbers.

To configure the router with the dial-peer information in Table 4, enter the following global configuration commands:

```
West> enable
Password:
West# configure terminal
West(config)# dial-peer voice 401 pots
West(config-dial-peer)# destination-pattern 14085553737
West(config-dial-peer)# port 0/0
West(config)# dial-peer voice 402 pots
West(config-dial-peer)# destination-pattern 14085554141
West(config-dial-peer)# port 0/1
West(config-dial-peer)# exit
West(config)#
```

These commands are summarized in Figure 10.

**Figure 10 West Router Configured for Local Dial Peers**



The **dial-peer** command always takes the argument **voice**. The number following it is the dial-peer tag, and **pots** is the type of dial peer.

Cisco IOS software refers to a telephone number as a *destination pattern* because it is the destination for an incoming or outgoing call. Enter these numbers with the **destination-pattern** command. A destination pattern can include asterisks (\*) and pound signs (#) from the telephone keypad, and commas (,) and periods (.), which have special meanings. Parentheses ( ( ) ), hyphens (-), slashes (/), and spaces ( ), which are often used to make telephone numbers easier for humans to read, are not allowed.

Notice that the commands in the examples puts the prefix 1 (used in the United States to indicate a long-distance number) and an area code in front of the remaining numbers to complete the destination pattern. You need to include similar codes for your country if the VoIP equipment needs to establish a connection to the PSTN.

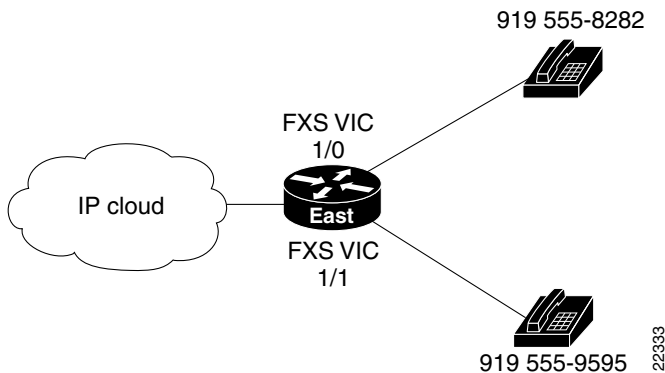


**Note**

The Cisco IOS software does not check the validity of the telephone number. It accepts any string of permitted characters as a valid number.

The business that owns the *West* router also has a branch office in the *East*. Figure 11 shows the *East* office network, and Table 6 lists the phone numbers, voice ports, and dial-peer tags for this office.

**Figure 11 Basic Voice Network (East Router)**



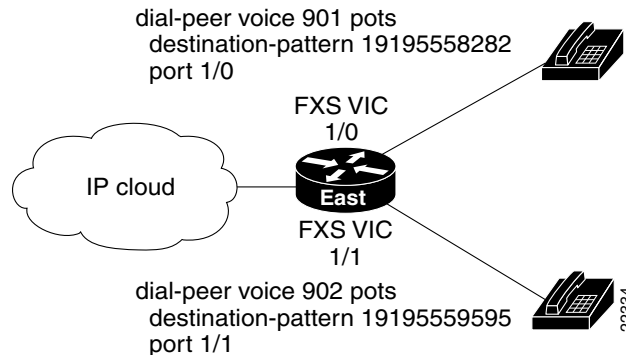
**Table 6** East Router Local Dial Peers

| Telephone Number | Destination Pattern | Voice Port | Dial-Peer Tag |
|------------------|---------------------|------------|---------------|
| 919 555-8282     | 19195558282         | 1/0        | 901           |
| 919 555-9595     | 19195559595         | 1/1        | 902           |

Enter the following commands to configure the local ports on the *East* Router with the dial-peer information in Table 6:

```
East(config)# dial-peer voice 901 pots
East(config-dial-peer)# destination-pattern 19195558282
East(config-dial-peer)# port 1/0
East(config)# dial-peer voice 902 pots
East(config-dial-peer)# destination-pattern 19195559595
East(config-dial-peer)# port 1/1
East(config-dial-peer)# exit
East(config)#
```

These commands are summarized in Figure 12.

**Figure 12** East Router Configured for Local Dial Peers

## Checking the Configuration

If you configured POTS dial peers on your router by following these examples, you can place calls between telephones connected to the same router. You can also use the **show dial-peer voice** command to verify that the data you configured is correct.



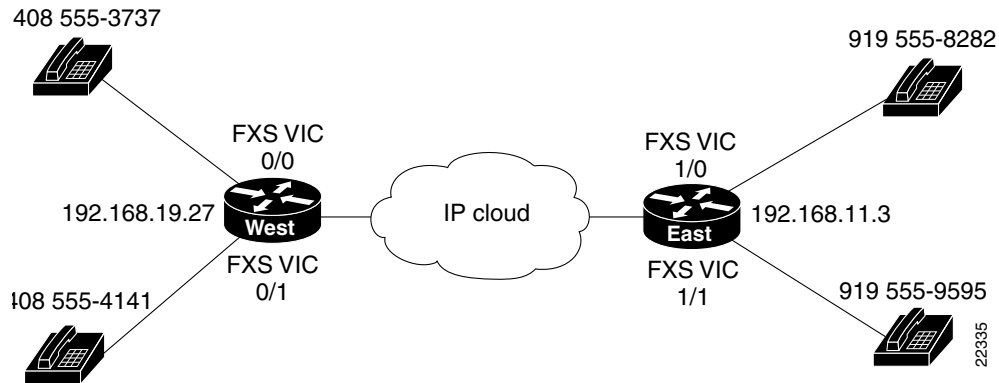
### Note

If the voice port is offline, use the interface configuration **no shutdown** command to enable it.

## Calling Between Routers

To enable the *West* and *East* offices to send voice traffic to each other over the same IP network they use for data traffic, use a WIC on each router to provide a connection to the IP network, as shown in Figure 13.

**Figure 13 IP Connection Between Routers**



Look at the connection between the *West* router and the IP network. This connection does not include a voice port or an attached telephone—it leads from a WAN interface to a remote destination somewhere on the IP network. IP routers know how to locate IP addresses on the network, but they do not know how to locate telephone numbers. To route an outgoing voice call over this connection, the *West* router has to associate a telephone number in the *East* office with the IP address of the *East* router.

Table 7 assigns a dial-peer tag to each telephone number and its associated IP address on the *West* router. This type of dial peer is called a *remote dial peer* or *VoIP dial peer*. (Remember, the dial-peer tags are arbitrary.) The term *VoIP* means that the dial peer associates a telephone number with an IP address.

**Table 7 West Router Remote Dial Peers**

| Remote Location | Telephone Number | Destination Pattern | IP Address   | Dial-Peer Tag |
|-----------------|------------------|---------------------|--------------|---------------|
| <i>East</i>     | 919 555-8282     | 19195558282         | 192.168.11.3 | 501           |
| <i>East</i>     | 919 555-9595     | 19195559595         | 192.168.11.3 | 502           |

Create a VoIP dial peer on the *West* router for every telephone on the *East* router, all associated with the same IP address. But it is much easier to use periods as wildcards, as shown in Table 8.

**Table 8 West Router Remote Dial Peers with Wildcards**

| Remote Location | Telephone Number | Destination Pattern | IP Address   | Dial-Peer Tag |
|-----------------|------------------|---------------------|--------------|---------------|
| <i>East</i>     | 919 555-xxxx     | 1919555....         | 192.168.11.3 | 501           |

Construct a table similar to Table 8 for your own routers, assigning your own telephone numbers, IP addresses, and dial-peer tags.



**Note**

The IP addresses shown in this guide are meant only as examples. When you configure your network, be sure to substitute your own IP addresses.

Enter the following information on the *West* router to create the dial-peer configuration given in Table 8:

```
West(config)# dial-peer voice 501 voip
West(config-dial-peer)# destination-pattern 1919555....
West(config-dial-peer)# session target ipv4:192.168.11.3
```

Cisco IOS software describes the remote network as the *session target*. This command is followed by the IP address of the remote router. The prefix **ipv4** means IP version 4. Alternatively, you can use the prefix **dns** followed by the Domain Name System (DNS) name as follows:

```
West(config-dial-peer)# session target dns:voice.eastrouter.com
```

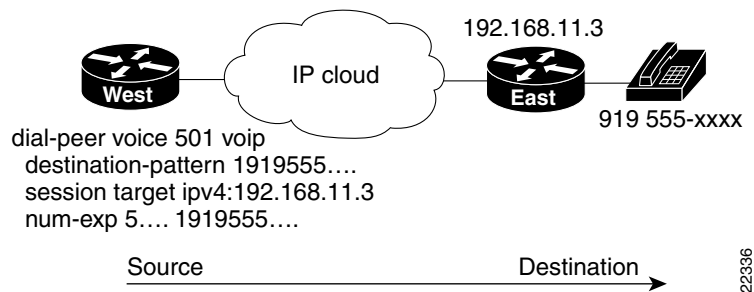
Configure a dial peer on each router for each telephone number on every other router connected to it. Make things easier by configuring number expansion for *East* router telephone numbers on the *West* router:

```
West(config)# num-exp 5.... 1919555....
```

Now users can dial a five-digit extension beginning with 5 from a telephone on the *West* router to reach a telephone on the *East* router.

These commands are summarized in Figure 14.

**Figure 14 West Router Configured for Remote Dial Peers**



The *West* router is now configured to send calls to the *East* router.

Table 9 shows how to configure the *East* router to send calls to the *West* router.

**Table 9 East Router Remote Dial Peers with Wildcards**

| Remote Location | Telephone Number | IP Address    | Dial-Peer Tag |
|-----------------|------------------|---------------|---------------|
| West            | 408 555-xxxx     | 192.168.19.27 | 801           |

Enter the following information on the *East* router to create the dial-peer configuration given in Table 9:

```
East(config)# num-exp 5.... 1408555....

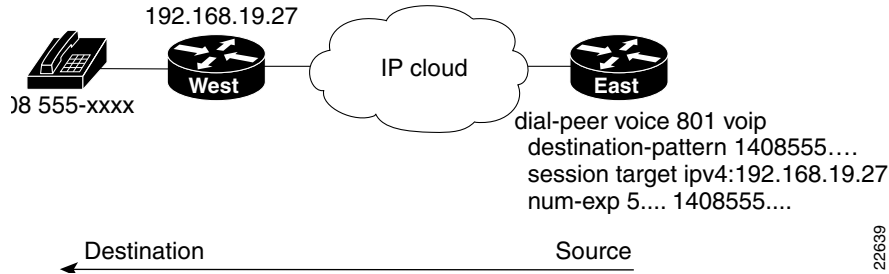
East(config)# dial-peer voice 801 voip

East(config-dial-peer)# destination-pattern 1408555....

East(config-dial-peer)# session target ipv4:192.168.19.27
```

These commands are summarized in Figure 15.

**Figure 15 East Router Configured for Remote Dial Peers**



## Other Routers on the Network

If the path between endpoints of a voice call travels through intermediate routers, configure those routers for VoIP traffic, as described in the section “Configuring FXS Interfaces” earlier in this guide.

You need to configure POTS or VoIP dial peers on an intermediate router only if that router also has voice devices attached to it.

## Checking the Configuration

If you configured VoIP dial peers on your router by following these examples, you can place calls from that router to telephones on the remote routers (using just the extension if you configured number expansion). If you have trouble placing calls, ping the remote router to make sure you have IP connectivity, or use the **show dial-peer voice** command to verify that the data you configured is correct.



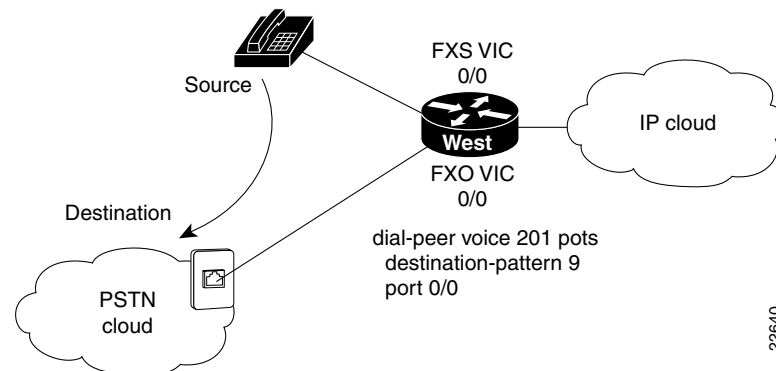
**Note**

Refer to the section “Tuning Parameters for Real-Time Voice Traffic” if you need to improve the quality of voice connections.

# Configuring FXO Interfaces

FXO interfaces provide a gateway from the VoIP network to the analog PSTN or to a PBX that does not support E&M signaling so that users can reach telephones and fax machines outside the VoIP network. Figure 16 shows a typical FXO gateway attached to the *West* router.

**Figure 16** FXO Gateway to PSTN



To create a POTS dial peer for an FXS interface as explained earlier, you enter the complete telephone number of the attached telephone as the destination pattern for incoming calls. However, to create a POTS dial peer for an FXO interface, the destination pattern refers to outgoing calls, and you can include wildcards in it because the PSTN performs the switching.

The VoIP feature can also remove digits that you do not want to send to the PSTN. For instance, to dial 9 to reach an outside line (that is, the analog PSTN), enter the following commands:

```
West> enable
Password:
West# configure terminal
West(config)# dial-peer voice 201 pots
West(config-dial-peer)# destination-pattern 9
West(config-dial-peer)# port 0/0
```

When you dial 9, the router makes a connection to the PSTN through voice port 0/0. The PSTN then provides a dial tone, and any digits you enter on the telephone thereafter are interpreted on the PSTN.

To enable *East* router users to make calls over the *West* router local PSTN, enter the following commands:

```
East(config)# dial-peer voice 701 voip
East(config-dial-peer)# destination-pattern 7
East(config-dial-peer)# session target ipv4:192.168.19.27

West(config)# dial-peer voice 601 pots
West(config-dial-peer)# destination-pattern 7
West(config-dial-peer)# port 0/0
```

When you dial 7 on the *East* router, the call is connected to the PSTN on the *West* router. The PSTN then provides a dial tone, and any digits you enter on the telephone thereafter are interpreted on the PSTN.

**Note**

---

In this example, *West* router voice port 0/0 has two separate POTS dial peers associated with it. Dial peer 201 matches calls beginning with the digit 9 and handles PSTN calls originating from the *West* router. Dial peer 601 matches calls beginning with the digit 7 and handles calls to the PSTN originating from the *East* router.

---

## Checking the Configuration

If you configured your FXO interface according to this example, you can place outgoing calls over the PSTN. If you have trouble placing calls, use the **show voice port** command to make sure that the VIC is installed correctly. Use the **show dial-peer voice** command to make sure that the data you configured is correct, and test the PSTN by connecting a handset directly to the PSTN outlet and placing a call.

**Note**

---

Refer to the section “Tuning Parameters for Real-Time Voice Traffic” if you need to improve the quality of voice connections.

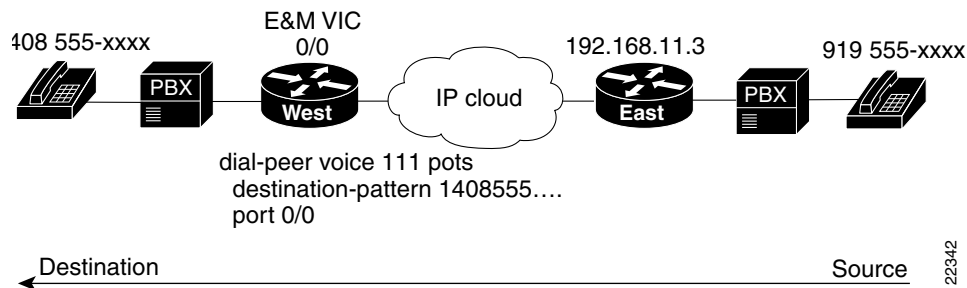
---

## Configuring E&M Interfaces

If you have more than a few voice users at each location, the cost of voice ports and routers and the effort needed to configure dial peers for all the combinations of origins and destinations increases rapidly. In this situation, it might be more efficient to use a PBX at each location to switch local traffic and direct incoming calls and then use E&M VICs to connect the PBXs over an IP network.

Figure 17 shows a company with two offices, *West* and *East*. Each office has a PBX to operate its internal telephone network, and the IP network carries voice traffic between the offices. Each PBX connects to an E&M VIC port in the router.

**Figure 17** Linking PBXs over the IP Network (Local Dial Peers)



To configure E&M voice ports, you need to use the following commands beginning in privileged EXEC mode:

| Command                                                                                                          | Task                                                                |
|------------------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------|
| <b>dial-type</b> { dtmf   pulse }                                                                                | Select the appropriate dial type for out-dialing.                   |
| <b>signal</b> { wink-start   immediate   delay-dial }                                                            | Select the appropriate signal type for this interface.              |
| <b>cptone</b> { australia   brazil   china   finland   france   germany   japan   northamerica   unitedkingdom } | Select the appropriate voice call progress tone for this interface. |
| <b>operation</b> { 2-wire   4-wire }                                                                             | Select the appropriate cabling scheme for this voice port.          |
| <b>type</b> { 1   2   3   5 }                                                                                    | Select the appropriate E&M interface type.                          |

Both PBXs in this example use E&M interface Type 2, with four-wire operation and immediate-start signaling. The values for your configuration depend on your PBX and are available from your telecommunications department or the PBX manufacturer. For more information about E&M interface configuration commands, refer to the “VoIP Commands” chapter of the *Cisco 1750 Router Voice-over-IP Configuration Guide*.

In this example, *West* users can dial 5 and a four-digit extension to reach telephones in the *East* Office. *East* users can dial 5 and a four-digit extension to reach telephones in the *West* office.

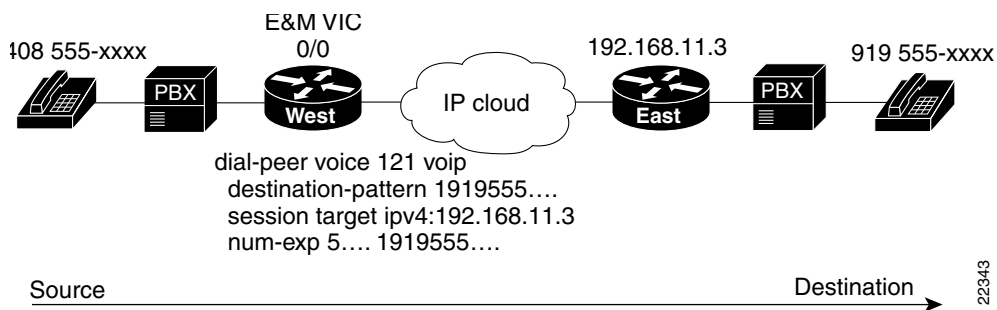
The *West* router connects to the PBX through an E&M VIC port 0/0. This port is associated with a POTS dial peer for incoming calls. But you no longer need to associate every telephone number with its own port. Instead, you can configure a local dial peer as if all the *West* telephones (represented by a wildcard destination pattern) are connected directly to this port, as shown in the following commands:

```
West> enable
Password:
West# configure terminal
West(config)# dial-peer voice 111 pots
West(config-dial-peer)# destination-pattern 1408555....
West(config-dial-peer)# port 0/0
```

Configure VoIP dial peers for outgoing calls and associate destination phone numbers on the *East* router with that router IP address, as shown in Figure 18, and in the following commands:

```
West(config)# dial-peer voice 121 voip
West(config-dial-peer)# destination-pattern 1919555....
West(config-dial-peer)# session target ipv4:192.168.11.3
West(config-dial-peer)# exit
West(config)#
```

**Figure 18** Linking PBXs over the IP Network (Remote Dial Peers)



Now configure number expansion so that numbers beginning with 5 (belonging to the *East* office) and sent by the *West* PBX to the *West* router are expanded into the full destination pattern:

```
West(config)# num-exp 5.... 1919555....
```



**Note**

You do not need to configure number expansion for calls from one *West* telephone to another *West* telephone because the PBX switches those calls.

Finally, enter the following global configuration **voice-port** command to configure the E&M port:

```
West(config)# voice-port 0/0
West(config-voice-port)# signal immediate
West(config-voice-port)# operation 4-wire
West(config-voice-port)# type 2
```



**Note**

For these commands to take effect, you have to cycle the port by using the **shutdown** and **no shutdown** commands.



**Note**

Configure the PBX to pass all dual-tone multifrequency (DTMF) signals to the router.

Configure the *East* router similar to the *West* router. The *East* router connects to the PBX through an E&M VIC port 0/1. Enter the following commands to configure a POTS dial peer for all *East* telephones:

```
East(config)# dial-peer voice 211 pots
East(config-dial-peer)# destination-pattern 1919555....
East(config-dial-peer)# port 0/1
```

Enter the following commands to configure a VoIP dial peer for telephones on the *West* router:

```
East(config)# dial-peer voice 221 voip
East(config-dial-peer)# destination-pattern 1408555....
East(config-dial-peer)# session target ipv4:192.168.19.27
East(config-dial-peer)# exit
East(config)#
```

Enter the following commands to configure number expansion and to make it easy for *East* users to dial numbers on the *West* router:

```
West(config)# num-exp 5.... 1408555....
```

Finally, configure the E&M port:

```
East(config)# voice-port 0/1
East(config-voice-port)# signal immediate
East(config-voice-port)# operation 4-wire
East(config-voice-port)# type 2
```

## Checking the Configuration

If you configured the E&M interfaces correctly, you can place calls from a telephone served by one PBX to a telephone served by the other PBX (using just the extension, if you configured number expansion). If you have trouble placing calls, ping the remote router to make sure you have IP connectivity.



### Note

---

Refer to the section “Tuning Parameters for Real-Time Voice Traffic” if you need to improve the quality of voice connections.

---

# Tuning Parameters for Real-Time Voice Traffic

Voice traffic is much more sensitive to timing variations than data traffic. For good voice performance, you need to configure your data network so that voice packets are not lost or delayed. To achieve the level of network performance needed for VoIP connections, use the following features to improve QoS:

- IP precedence
- Resource Reservation Protocol (RSVP)
- Multilink Point-to Point Protocol (PPP) interleaving
- Real-Time Transport Protocol (RTP) header compression

Cisco IOS software provides many other tools for ensuring QoS, such as custom queuing, priority queuing, and weighted fair queuing. For further information and more detailed examples of QoS configuration, refer to the *Cisco 1750 Router Voice-over-IP Configuration Guide*.


**Note**

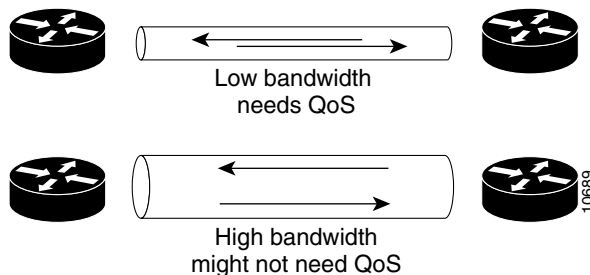
QoS measures the level of network performance. It does not directly measure the quality of the voice signal.

Configuring VoIP on a Frame Relay link involves special considerations. These are discussed in the section “Configuring Frame Relay for VoIP.”

## Need for Quality of Service

On a relatively low-bandwidth connection, such as a PPP or High-Level Data Link Control (HDLC) serial link, you should consider using methods to ensure QoS. If you have a high-bandwidth network, such as Ethernet or Fast Ethernet, and voice and data traffic together occupy only a small fraction of the bandwidth available, you might not need to provide QoS. (See Figure 19.)

**Figure 19** *Bandwidth v. Quality of Service*



## IP Precedence

Use the **ip precedence** command to give voice packets a higher priority over other IP data traffic. Every IP packet is given a precedence level. In IP precedence, the numbers 1 through 5 identify classes for IP flows; the numbers 6 through 7 are used for network and backbone routing and updates. You can configure voice packets for higher priority by setting the IP precedence value to 5. Internal routers using weighted fair queuing will give these packets priority. This command only applies to VoIP dial peers. The following example sets the IP precedence to 5:

```
Router> enable
Password:
Router# configure terminal
Router(config)# dial-peer voice 221 voip
Router(config-dial-peer)# ip precedence 5
```

The **ip precedence** command should also be used if RSVP is not enabled and the user would like to give voice packets a higher priority over other IP data traffic.

## RSVP

Resource Reservation Protocol (RSVP) enables routers to reserve enough bandwidth on an interface for reliable and quality performance. RSVP works well on PPP, HDLC, and similar serial line interfaces. It does not work well on multi-access LANs.

You should configure RSVP if you have a serial interface and any of the following:

- Links slower than 2 Mbps
- Links with high utilization
- Need for the best possible voice quality



### Note

---

If you configure multilink PPP interleaving, you can use the **ip rtp reserve** command instead of configuring RSVP. Refer to the next section on “Multilink PPP Interleaving.”

---

## Configuring RSVP

By default, RSVP is disabled so that it is backwards compatible with routers that do not implement it. To enable RSVP on an IP network, enter the **ip rsvp bandwidth** command from interface configuration mode. The following example enables RSVP on serial interface 0/0 and sets the maximum bandwidth to 96 kbps and the maximum bandwidth per single request to 24 kbps:

```
Router(config)# interface serial 0/0
Router(config-if)# ip rsvp bandwidth 96 24
```

The default maximum bandwidth is 75 percent of the bandwidth available on each interface.

RSVP must be enabled at each LAN or WAN interface that voice packets will travel across. After enabling RSVP you must also use the **req-qos** dial-peer configuration command to request an RSVP session on each VoIP dial peer. Otherwise, no bandwidth is reserved for voice traffic.

```
Router(config-dial-peer)# req-qos controlled-load
```

For further information about this command, refer to the *Cisco 1750 Router Voice-over-IP Configuration Guide*.

## Multilink PPP Interleaving

Multilink PPP interleaving allows large packets to be multilink-encapsulated and fragmented into smaller packets to make sure that voice packets are transmitted without delay; small real-time packets, which are not multilink-encapsulated, are transmitted between fragments of the large packets. The interleaving feature also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be transmitted earlier than other flows.

You should configure multilink PPP interleaving if you have a dialer, ISDN PRI or BRI interface, or a virtual template, and either of the following:

- Point-to-point connections using PPP encapsulation
- Links slower than 2 Mbps



### Note

---

Do not use multilink PPP on links faster than 2 Mbps.

---

## Configuring Multilink PPP Interleaving

To configure multilink PPP and interleaving on a dialer, ISDN PRI or ISDN BRI interface, or a virtual template, you must first configure multilink PPP and interleaving on the interface or template by entering the following commands in interface configuration mode:

```
Router(config-if)# encapsulation ppp
Router(config-if)# ppp multilink
Router(config-if)# ppp multilink interleave
```

Optionally, configure a maximum fragment delay:

```
Router(config-if)# ppp multilink fragment-delay milliseconds
```

You can also reserve a special queue for real-time packet flows to specified destination User Datagram Protocol (UDP) ports, allowing real-time traffic to have higher priority than other flows. You need the following command only if you have not configured RSVP:

```
Router(config-if)# ip rtp reserve lowest-UDP-port range-of-ports
```

For virtual templates only, apply the virtual template to the multilink bundle:

```
Router(config-if)# multilink virtual-template 1
```

To create a virtual template interface, enter the following **interface virtual-template** global configuration command:

```
Router(config)# interface virtual-template 1
```

## RTP Header Compression

RTP header compression on a PPP, HDLC, or similar serial interface compresses the packet header to reduce network overhead.

You should configure RTP header compression on a serial interface if you have either of the following:

- Links slower than 2 Mbps
- Need to save bandwidth



**Note**

---

Do not use RTP header compression on links faster than 2 Mbps.

---

## Configuring RTP Header Compression

Enable RTP header compression at both ends of the serial link by entering the following **ip rtp header-compression** interface configuration command:

```
Router(config-if)# ip rtp header-compression
```



**Note**

---

When you enter the **show running-config** command, the format of the **ip rtp header-compression** command will change to **ip rtp header-compression iphc-format**.

---

## Configuring Frame Relay for VoIP

Configuring VoIP on a Frame Relay link involves certain special considerations to ensure acceptable voice quality. For Frame Relay links with slow output rates (64 kbps or less) and with data and voice being transmitted over the same permanent virtual circuit (PVC), you should configure the following parameters:

- Lower maximum transmission unit (MTU) size—Voice packets are generally small. If you decrease the MTU size to 300 bytes, large data packets can be broken up into smaller data packets that are more easily interleaved with voice packets.

The following example configures an MTU size of 300 bytes over serial interface 0/0:

```
Router# interface serial 0/0
Router(config-if)# mtu 300
```



### Note

Lowering the MTU size affects data throughput speed.

- RSVP—Configure RSVP on the subinterfaces (which correspond to PVCs) to reserve bandwidth for voice channels. Refer to the section on “RSVP” earlier in this guide.

The following example configures RSVP over serial subinterface 0/0.1:

```
Router(config-if)# interface serial 0/0.1 point-to-point
Router(config-if)# ip address 192.168.19.0 255.0.0.0
Router(config-if)# ip rsvp bandwidth 48 48
```

- RTP header compression—Configure RTP header compression on the subinterfaces to minimize the size of the voice packet. Refer to the section on “RTP Header Compression” earlier in this guide.

The following command configures RTP header compression on the selected subinterface:

```
Router(config-if)# frame-relay ip rtp header-compression
```

- Traffic shaping—Use traffic shaping to control the outbound traffic rate; otherwise, voice packets can be discarded. In Cisco IOS Release 12.0T, Frame Relay traffic shaping is not compatible with RSVP. Use generic traffic shaping instead, and set the committed information rate (CIR) equal to the port speed. This prevents the router from exceeding the CIR rate and discarding the frames.

The following command configures generic traffic shaping with a CIR of 32000 bps:

```
Router(config-if)# traffic-shape rate 32000
```

For further information and more detailed examples of Frame Relay configuration, refer to the *Cisco 1750 Router Voice-over-IP Configuration Guide*.



### Note

When you finish configuring the router, you must save the new configuration to nonvolatile RAM (NVRAM) by copying **running-config** to **startup-config**. You should also save periodically during the configuration process. Refer to the “Saving the Configuration” section on page 11 for details.

# List of Terms

This section defines some of the VoIP terms and concepts used in this guide.

**BRI**—Basic Rate Interface, an ISDN interface.

**CIR**—Committed information rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

**Call leg**—A segment of a call path, for instance, between a telephone and a router, a router and a network, a router and a PBX, or a router and the PSTN. Each call leg corresponds to a dial peer.

**Destination pattern**—The pattern of numbers that identifies the destination of an incoming call; in other words, a phone number.

**Dial peer**—A software object that ties together a voice port and a local telephone number (*local dial peer* or *POTS dial peer*) or an IP address and a remote telephone number (*remote dial peer* or *VoIP dial peer*). Each dial peer corresponds to a call leg.

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

**E.164**—The international public telecommunications numbering plan. A standard set by ITU-T that addresses telephone numbers.

**E&M**—A signaling technique for two-wire and four-wire interfaces that provide connections to PBX trunk lines (tie lines).

**FXO**—Foreign exchange office, a type of VIC interface. Using a standard RJ-11 modular telephone cable, the FXO VIC connects local calls to a PSTN central office or to a PBX. This is the interface a standard telephone provides.

**FXS**—Foreign exchange station, a type of VIC interface. The FXS VIC connects directly to a standard telephone, fax machine, or similar device over a standard RJ-11 modular telephone cable, and supplies ringing voltage, dial tone, and similar signals to it.

**HDLC**—High-Level Data Link Control. A data link layer protocol that specifies a data encapsulation method on synchronous serial links using frame characters and checksums.

**MTU**—Maximum transmission unit. Maximum packet size that a particular interface can handle.

**Multilink PPP**—Multilink Point-to-Point Protocol, a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

**NANP**—North American Numbering Plan.

**PBX**—Private branch exchange, a private telephone switching system.

**POTS**—Plain old telephone service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the public switched telephone network.

**POTS dial peer**—A software object that ties together a voice port and the telephone number of a device attached to the port (also called *local dial peer*).

**PRI**—Primary Rate Interface, an ISDN interface.

**PSTN**—Public switched telephone network.

**PVC**—Permanent virtual circuit.

**QoS**—Quality of service, a measure of the level of performance needed for a particular application, such as a VoIP connection.

**RSVP**—Resource Reservation Protocol, a network protocol that enables routers to reserve the bandwidth necessary for reliable performance.

**RTP**—Real-Time Transport Protocol, a network protocol used to carry audio and video packet traffic over an IP network.

**session target**—A remote IP or DNS address specified for a dial peer.

**tag**—A positive integer in the range 1 to  $2^{31} - 1$  (2147483647) used to identify a dial peer.

**UDP**—User Datagram Protocol, a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery.

**VIC**—Voice interface card. VICs install in a slot in the router, and provide the connection to the telephone equipment or network. There are three types of VICs: FXS, FXO, and E&M. Each VIC provides two ports of the same type.

**VoIP**—Voice-over-IP, a feature that carries voice traffic, such as telephone calls and faxes, over an IP network, simultaneously with data traffic.

**VoIP dial peer**—A software object that ties together an IP address and a telephone number at a remote site reached over the IP network (also called *remote dial peer*).

# Cisco Connection Online

Cisco Connection Online (CCO) is Cisco Systems' primary, real-time support channel. Maintenance customers and partners can self-register on CCO to obtain additional information and services.

Available 24 hours a day, 7 days a week, CCO provides a wealth of standard and value-added services to Cisco's customers and business partners. CCO services include product information, product documentation, software updates, release notes, technical tips, the Bug Navigator, configuration notes, brochures, descriptions of service offerings, and download access to public and authorized files.

CCO serves a wide variety of users through two interfaces that are updated and enhanced simultaneously: a character-based version and a multimedia version that resides on the World Wide Web (WWW). The character-based CCO supports Zmodem, Kermit, Xmodem, FTP, and Internet e-mail, and it is excellent for quick access to information over lower bandwidths. The WWW version of CCO provides richly formatted documents with photographs, figures, graphics, and video, as well as hyperlinks to related information.

You can access CCO in the following ways:

- WWW: <http://www.cisco.com>
- WWW: <http://www-europe.cisco.com>
- WWW: <http://www-china.cisco.com>
- Telnet: [cco.cisco.com](http://cco.cisco.com)
- Modem: From North America, 408 526-8070; from Europe, 33 1 64 46 40 82. Use the following terminal settings: VT100 emulation; databits: 8; parity: none; stop bits: 1; and connection rates up to 28.8 kbps.

For a copy of CCO's Frequently Asked Questions (FAQ), contact [cco-help@cisco.com](mailto:cco-help@cisco.com). For additional information, contact [cco-team@cisco.com](mailto:cco-team@cisco.com).



## Note

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