

Configuring Voice over IP

This chapter provides an overview of Voice over IP (VoIP) technology and gives step-by-step configuration tasks. The chapter contains the following sections:

- [VoIP Benefits, page 12](#)
- [VoIP Call Processing, page 12](#)
- [VoIP Prerequisite Tasks, page 13](#)
- [VoIP Network Design Considerations, page 14](#)
- [VoIP Configuration Task List, page 15](#)
- [Configuring VoIP over Frame Relay, page 17](#)
- [VoIP Configuration Examples, page 18](#)

To identify the hardware platform or software image information associated with a feature in this chapter, use the [Feature Navigator](#) on Cisco.com to search for information about the feature or refer to the software release notes for a specific release. For more information, see the “Identifying Supported Platforms” in the “Using Cisco IOS Software” chapter.

Voice over IP Overview

VoIP is a Layer 3 network protocol that uses various Layer 2 point-to-point or link-layer protocols such as PPP, Frame Relay, or ATM for its transport. VoIP enables Cisco routers, access servers, and multiservice access concentrators to carry and send voice and fax traffic over an IP network. In VoIP, digital signal processors (DSPs) segment the voice signal into frames and store them in voice packets. These voice packets are transported via IP in compliance with a voice communications protocol or standard such as H.323, Media Gateway Control Protocol (MGCP), or Session Initiation Protocol (SIP).

[Table 3](#) shows the relationship between the Open System Interconnection (OSI) reference model and the protocols and functions of VoIP network elements.

Table 3 Relationship of OSI Reference Model to VoIP Protocols and Functions

OSI Layer Number	OSI Layer Name	VoIP Protocols and Functions
7	Application	NetMeeting/Applications
6	Presentation	Codecs
5	Session	H.323/MGCP/SIP
4	Transport	RTP/TCP/UDP

Table 3 Relationship of OSI Reference Model to VoIP Protocols and Functions (continued)

OSI Layer Number	OSI Layer Name	VoIP Protocols and Functions
3	Network	IP
2	Data Link	Frame Relay, ATM, Ethernet, PPP, MLP, and more

Cisco IOS software supports the following call control protocols and standards in Release 12.2:

- H.323—the International Telecommunication Union-Telecommunications Standardization Sector (ITU-T) specification for sending voice, video, and data across a network. The H.323 specification includes several related standards, such as H.225 (call control), H.235 (security), H.245 (media path and parameter negotiation), and H.450 (supplementary services). For more information, see the “H.323 Overview” chapter in this configuration guide.
- MGCP—Media Gateway Control Protocol, an Internet Engineering Task Force (IETF) draft standard for controlling voice gateways through IP networks. For more information, see the “Configuring MGCP and Related Protocols” chapter in this configuration guide.
- SIP—Session Initiation Protocol, defined in IETF RFC 2543. For more information, see the “Configuring SIP” chapter in this guide.

VoIP protocols typically use Real-time Transport Protocol (RTP) for the media stream or speech path. RTP uses User Datagram Protocol (UDP) as its transport protocol. Voice signaling traffic often uses Transmission Control Protocol (TCP) as its transport medium. The IP layer provides routing and network-level addressing; the data-link layer protocols control and direct the transmission of the information over the physical medium.

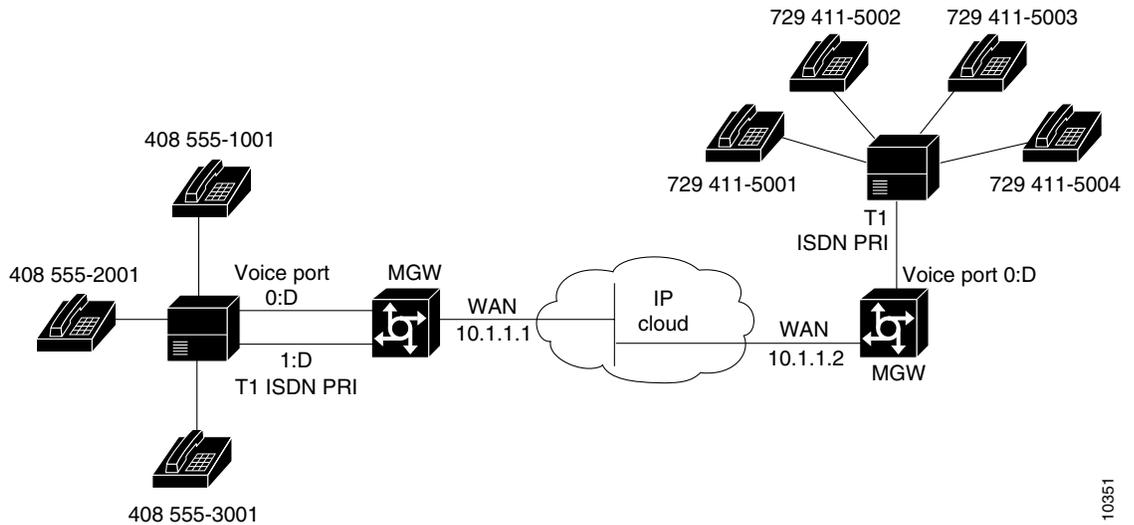
The main factor in choosing between VoIP and the Layer 2 VoFR and VoATM transport alternatives is interworking with other voice or multimedia applications. Generally speaking, Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) are effective WAN transport technologies and are more bandwidth-efficient than VoIP. But VoFR and VoATM cannot be deployed over LANs or to the desktop. VoIP is the predominant form of voice-over-packet deployed today, and, for implementing voice applications, it is usually the only choice even if the first step in network deployment is pure transport between existing PBXs.

VoIP leverages the entire Internet and Intranet IP infrastructure for routing, making it easy to design any-to-any calling in a VoIP network. VoIP also allows multivendor interworking, which is more difficult to achieve with VoFR and VoATM applications because standards for those solutions have only recently emerged.

Cisco VoIP is frequently used in two primary applications:

- To provide a central-site telephony termination facility for voice traffic coming from multiple voice-equipped remote office facilities. [Figure 2](#) illustrates this application using Cisco AS5300 universal access servers as the central-site telephony termination devices.

Figure 2 VoIP Used as a Central-Site Telephony Termination Facility

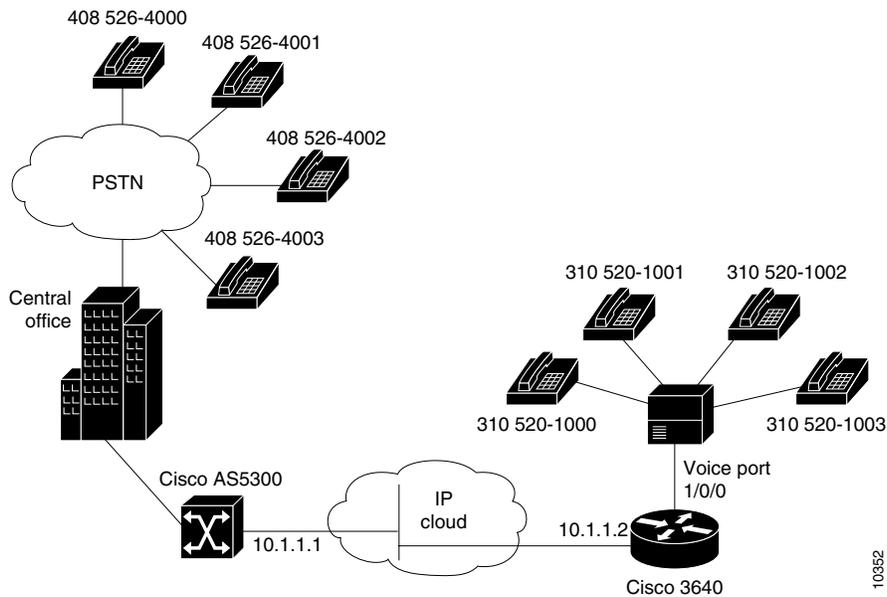


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- To provide Public Switched Telephone Network (PSTN) gateway functionality for Internet telephone traffic. Cisco VoIP used in this scenario leverages the standardized use of H.323-based Internet telephone client applications. In the case of a device with extensive capacity running VoIP (such as the Cisco AS5800 universal access server), the functionality provided is equivalent to that of a carrier-class switch.

Figure 3 illustrates this application, using a Cisco AS5300 as the PSTN gateway.

Figure 3 VoIP Used as a PSTN Gateway for Internet Telephone Traffic



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To use VoIP, you must install the appropriate hardware in your Cisco device: for example, a voice-specific port adapter or network module. The specific voice hardware required depends on the router or access server used. The number of ports or channels available for sending VoIP data depends on the capacity of the specific voice hardware installed. For more information about the physical characteristics, capacity, installation, or configuration of voice hardware, refer to the online documentation for your router or access server.

VoIP Benefits

VoIP offers the following benefits:

- Toll bypass (either one- or two-stage toll bypass, depending on the environment in which VoIP is deployed)
- Remote PBX presence over WANs
- PSTN voice-traffic and fax-traffic offload
- Universally accessible voice-mail and fax-mail services
- Unified voice and data trunking
- Plain old telephone service (POTS)-Internet telephony gateways
- Support for Microsoft NetMeeting when a Cisco router is used as a voice gateway

VoIP Call Processing

Before configuring VoIP on a Cisco router or access server, it helps to have a high-level understanding of what happens when you place a VoIP call. The following sequence outlines the general flow of a two-party VoIP voice call using H.323:

1. The caller picks up the handset, signaling an off-hook condition to the signaling application layer of VoIP.
2. The session application layer of VoIP issues a dial tone and waits for the caller to dial.
3. When the caller dials the number, the dialed digits are accumulated and stored by the session application.
4. After enough digits are accumulated to match a configured destination pattern, the telephone number is mapped to an IP host via the dial plan mapper. The IP host has a direct connection to the destination telephone number or a PBX that is responsible for completing the call to the configured destination pattern.
5. The session application runs the H.323 session protocol to establish a transmission and a reception channel for each direction over the IP network. If the call is being handled by a PBX, the PBX forwards the call to the destination telephone. If Resource Reservation Protocol (RSVP) has been configured, the RSVP reservations are put into effect to achieve the desired quality of service (QoS) over the IP network.
6. The coder-decoders (codecs) are enabled for both ends of the connection and the conversation proceeds using RTP/UDP/Internet Protocol (IP) as the protocol stack. Voice signals are digitized, compressed, packaged into discrete packets, and transported over the network.

7. Any call-progress indications or other signals that can be carried in-band are cut through the voice path as soon as the end-to-end audio channel is established. Signaling that can be detected by the voice ports (for example, in-band dual tone multifrequency [DTMF] digits after the call setup is complete) is also trapped by the session application at either end of the connection and carried over the IP network encapsulated in Real Time Conferencing Protocol (RTCP) using the RTCP APP extension mechanism.
8. When either end of the call hangs up, the RSVP reservations are torn down (if RSVP is used) and the session ends. Each end becomes idle, waiting for the next off-hook condition to trigger another call setup.

VoIP Prerequisite Tasks

Before configuring a Cisco router, access server, or gateway to use VoIP, complete the following tasks:

- Establish a working IP network in which delay (as measured by ping tests) and jitter are minimized. For more information about configuring IP, refer to the “IP Overview,” “Configuring IP Addressing,” and “Configuring IP Services” chapters in the *Cisco IOS IP Configuration Guide*, Release 12.2.
- Install a voice network module (VNM), voice feature card (VFC), or universal port dial feature card into the appropriate slot of your Cisco router, access server, or gateway. For more information about the physical characteristics, capacity, memory requirements, and installation instructions for the hardware you are installing, refer to the appropriate platform-specific hardware documentation.
- Make sure your router, access server, or gateway has sufficient DRAM installed to support VoIP, and make sure you are running a version and image of Cisco IOS software that supports VoIP. For more information, refer to the release notes for the platform you are using and the version of Cisco IOS you are running, or use the Feature Navigator tool on Cisco.com.
- Complete basic configuration of your router, access server, or gateway. For more information about these basic configuration tasks, refer to the “Configuring H.323 Gateways,” “Configuring Voice Ports,” and “Configuring Dial Plans, Dial Peers, and Digit Manipulation” chapters of this configuration guide.
- Formulate the beginning of a dial plan that includes the following:
 - Logical network diagram showing voice ports and components to which they connect, including telephones, fax machines, PBX or key systems, other voice devices that require connection, and voice-enabled routers.
 - Connection details, including physical interfaces, relevant LAN and WAN ports, and all voice ports; for each WAN, the type (Frame Relay, PPP, etc.); for Frame Relay, relevant PVCs and link access rates.
 - Phone numbers or extensions for each voice port, logically laid out and consistent with existing private dial plans and external dialing schemes.
- Establish a working telephony network based on your company dial plan.
- Integrate your dial plan and telephony network into your existing IP network topology. In general, we recommend the following practices:
 - Make routing or dialing transparent to users; for example, avoid secondary dial tones from secondary switches, where possible.
 - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

VoIP Network Design Considerations

You must have a well-engineered network end-to-end when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features geared toward improving quality of service (QoS).

Quality of service refers to the ability of a network to provide differentiated service to selected network traffic over various underlying technologies. QoS is not inherent in a network infrastructure. Rather, you institute QoS by strategically enabling appropriate QoS features throughout your network.

Cisco IOS software provides many tools for enabling QoS on your backbone, such as Random Early Detection (RED), Weighted Random Early Detection (WRED), fancy queueing (meaning custom, priority, or weighted fair queueing), IP RTP priority, low-latency queueing (LLQ), and IP precedence. To configure your IP network for real-time voice traffic, you must take into consideration the entire scope of your network and then select the appropriate QoS tool or tools. For complete information about any of these topics, refer to the *Cisco IOS Quality of Service Solutions Configuration Guide*, Release 12.2. In addition, refer to the “Configuring QoS for Voice” chapter in this configuration guide.

Remember that to improve voice network performance, QoS must be configured throughout your network, not just on the Cisco devices running VoIP. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations; the QoS tasks they perform might differ as well. To configure your IP network for real-time voice traffic, you must consider the functions of both edge and backbone routers in your network and then select the appropriate QoS tool or tools.

VoIP Quality of Service Tips

This section explains the quality issues that you should consider when building VoIP networks and offers a few tips about configuring VoIP with the appropriate QoS. For detailed information on these topics, refer to “Voice Quality Tuning Commands” in the “Configuring Voice Ports” chapter.

Voice traffic differs from data traffic in the following ways:

- Data is often bursty by nature; voice is deterministic (smooth).
- Data applications resend dropped packets; voice applications can only conceal dropped packets.
- Data applications can usually tolerate some delay; voice applications must minimize delay, so that the recipient does not hear clips in the transmission.

These differences mandate the use of QoS strategies to give strict priority to voice traffic, ensuring reliable delivery and minimal delay for networks that carry both voice and data.

Delay

Delay is the time it takes for VoIP packets to travel between two endpoints. Because of the speed of network links and the processing power of intermediate devices, some delay is expected; however, you should attempt to minimize this delay.

The human ear normally accepts a delay of about 150 milliseconds (ms) without noticing problems. (The ITU G.114 standard recommends no more than 150 ms of one-way delay.) When delay exceeds 150 ms, a conversation becomes more and more like a citizens band (CB) radio interchange in which one person must wait for the other to stop speaking before beginning to talk. This type of delay is often evident on international long-distance calls. You can measure delay fairly easily by using ping tests at various times of the day with different network traffic loads. If network delay is excessive, reduce it before deploying VoIP in your network.

Jitter

Although delay can cause unnatural starting and stopping of conversations, variable-length delays (also known as *jitter*) can cause a conversation to break and become unintelligible. Jitter is not usually a problem with PSTN calls because the bandwidth of calls is fixed. However, in VoIP networks in which existing data traffic might be bursty, jitter can become a problem. Cisco voice gateways have built-in dejitter buffering to compensate for a certain amount of jitter, but if jitter is constant on a network, identify the source and control it before deploying a VoIP network.

Serialization

Serialization is a term that describes what happens when a router attempts to send both voice and data packets through an interface. In general, voice packets are very small (80 to 256 bytes), and data packets can be very large (1500 to 18,000 bytes). On relatively slow links, such as WAN connections, large data packets can take a long time to send onto the wire. When these large packets are mixed with smaller voice packets, the excessive transmission time can lead to both delay and jitter. You can use fragmentation to reduce the size of the data packets so that the delay and jitter also decrease.

Bandwidth Consumption

Traditional voice conversations consume 64 kbps of network bandwidth. When this voice traffic is run through a VoIP network, it can be compressed and digitized by digital signal processors (DSPs built into the routers). This compression can reduce the calls to sizes as small as 5.3 kbps for voice samples. After the packets go onto the IP network, the appropriate IP/UDP/RTP headers must be added. This can add a substantial amount of bandwidth to each call (about 40 bytes per packet). Technologies such as RTP header compression, however, can reduce the IP header overhead to about two bytes. In addition, VAD does not send any packets unless there is active speech.

VoIP Configuration Task List

To configure VoIP on a Cisco router or access server, complete the following tasks:

-
- Step 1** Configure your IP network for real-time voice traffic. Fine-tuning your network to adequately support VoIP involves a series of protocols and features designed to improve QoS. To configure your IP network for real-time voice traffic, consider the entire scope of your network. Then select and configure the appropriate QoS tool or tools.
- Refer to [“Configuring VoIP over Frame Relay” section on page 17](#), and the “Configuring QoS for Voice” chapter for information about how to select and configure the appropriate QoS tools to optimize voice traffic on your network.
- Step 2** If you plan to run VoIP over Frame Relay, you must take certain factors into consideration when configuring VoIP for it to run smoothly over Frame Relay. For example, a public Frame Relay cloud provides no guarantees for QoS. Refer to the [“Configuring VoIP over Frame Relay” section on page 17](#) for information about deploying VoIP over Frame Relay.

Step 3 Configure dial peers. Use the **dial-peer voice** command to define dial peers and switch to the dial-peer configuration mode. Each dial peer defines the characteristics associated with a call leg. A call leg is a discrete segment of a call connection that lies between two points in the connection. An end-to-end call consists of four call legs, two from the perspective of the source access server, and two from the perspective of the destination access server. Dial peers are used to apply attributes to call legs and to identify call origin and destination. There are two types of dial peers used for VoIP:

- POTS—Dial peer describing the characteristics of a traditional telephony network connection. POTS dial peers point to a particular voice port on a voice network device. To configure a POTS dial peer, you must configure the associated telephone number and the logical interface. Use the **destination-pattern** command to associate a telephone number with a POTS peer. Use the **port** command to associate a specific logical interface with a POTS peer. In addition, you can specify direct inward dialing for a POTS peer by using the **direct-inward-dial** command.
- VoIP—Dial peer describing the characteristics of the IP network connection. VoIP dial peers point to specific VoIP devices. To configure a VoIP dial peer, you must configure the associated destination telephone number and a destination IP address. Use the **destination-pattern** command to define the destination telephone number associated with a VoIP peer. Use the **session target** command to specify a destination IP address for a VoIP peer.

Refer to the “Configuring Dial Plans, Dial Peers, and Digit Manipulation” chapter in this configuration guide for additional information about dial-peer characteristics and configuring dial peers.

Step 4 Configure number expansion. Use the **num-exp** command to configure number expansion if your telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. Refer to the “Configuring Digit Manipulation Features” section of the “Configuring Dial Plans, Dial Peers, and Digit Manipulation” chapter of this guide for information about number expansion.

Step 5 Optimize dial peer and network interface configurations. You can use VoIP dial peers to define characteristics such as codec, voice activity detection (VAD), and additional QoS parameters (when RSVP is configured). If you have configured RSVP, use either the **req-qos** or **acc-qos** command to configure QoS parameters. Use the **codec** command to configure specific voice coder rates. Use the **vad** command to disable voice activation detection and the transmission of silence packets. Refer to the “Configuring Dial Plan Options for VoIP Dial Peers” section of the “Configuring Dial Plans, Dial Peers, and Digit Manipulation” chapter in this guide for additional information about optimizing dial-peer characteristics.

Step 6 Configure voice ports. In general, voice-port commands define the characteristics associated with a particular voice-port signaling type. The following voice signaling types are supported:

- FXO—Foreign Exchange Office interface
- FXS—The Foreign Exchange Station interface
- E&M—The “ear and mouth” interface (also called the “earth and magnet interface, or the “recEive and transMit” interface)

Under most circumstances, the default voice-port command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice-port values configured, depending on the specifications of the devices in your telephony network. For information about configuring voice ports, refer to the “Configuring Voice Ports” chapter in this guide.

Configuring VoIP over Frame Relay

You must consider certain factors when configuring VoIP to ensure that it runs smoothly over Frame Relay. A public Frame Relay cloud provides no guarantees for QoS. For real-time traffic to be sent in a timely manner, the data rate must not exceed the committed information rate (CIR) or packets may be dropped. In addition, Frame Relay traffic shaping and RSVP are mutually exclusive. Remembering this is particularly important if multiple data link connection identifiers (DLCIs) are carried on a single interface.

For Frame Relay links with slow output rates (less than or equal to 64 kbps) in which data and voice are being sent over the same permanent virtual circuit (PVC), we recommend the following solutions:

- Separate DLCIs for voice and data—By providing a separate subinterface for voice and data, you can use the appropriate QoS tool for each line. For example, with each DLCI using 32 kbps of a 64-kbps line, you could do the following:
 - Apply adaptive traffic shaping to both DLCIs.
 - Use RSVP or IP Precedence to prioritize voice traffic.
 - Use compressed RTP to minimize voice packet size.
 - Use weighted fair queueing to manage voice traffic.
- Lower the maximum transmission unit (MTU) size—Voice packets are generally small. With a lower MTU size (for example, 300 bytes), large data packets can be broken up into smaller data packets that can more easily be interwoven with voice packets.



Note Some applications do not support a smaller MTU size. If you decide to lower the MTU size, use the **ip mtu** command; this command affects only IP traffic.



Note Lowering the MTU size affects data throughput speed.

- CIR equal to line rate—Make sure that the data rate does not exceed the CIR. One way you can make sure that the data rate does not exceed the CIR is through generic traffic shaping. For example, you could do the following:
 - Use IP precedence to prioritize voice traffic.
 - Use compressed RTP to minimize voice packet header size.
- Traffic shaping—Use adaptive traffic shaping to throttle back the output rate based on the backward explicit congestion notification (BECN). If the feedback from the switch is ignored, both data and voice packets might be discarded. Because the Frame Relay switch does not distinguish between voice and data packets, voice packets could be discarded, resulting in a deterioration of voice quality. For example, you could do the following:
 - Use compressed RTP, reduced MTU size, and adaptive traffic shaping based on BECN to hold the data rate to the CIR.
 - Use generic traffic shaping to obtain a low interpacket wait time. For example, set Bc to 4000 to obtain an interpacket wait of 125 ms.



Note

We recommend FRF.12 fragmentation setup rules for VoIP connections over Frame Relay. For more information, refer to the “Configuring Voice over Frame Relay” chapter.

VoIP Configuration Examples

This section contains the following configuration examples:

- [VoIP over Frame Relay Configuration Example, page 18](#)
- [VoIP for the Cisco 3600 Series Configuration Examples, page 19](#)
- [VoIP for the Cisco AS5300 Configuration Example, page 26](#)
- [VoIP for the Cisco AS5800 Configuration Example, page 29](#)

VoIP over Frame Relay Configuration Example

For Frame Relay, it is customary to configure a main interface and one subinterface per permanent virtual circuit (PVC). The following example configures a Frame Relay main interface and a subinterface so that voice and data traffic can be successfully transported:

```
interface Serial0/0
 ip mtu 300
 no ip address
 encapsulation frame-relay
 no ip route-cache
 no ip mroute-cache
 fair-queue 64 256 1000
 frame-relay ip rtp header-compression

interface Serial0/0.1 point-to-point
 ip mtu 300
 ip address 40.0.0.7 255.0.0.0
 no ip route-cache
 no ip mroute-cache
 bandwidth 64
 traffic-shape rate 32000 4000 4000
 frame-relay interface-dlci 16
 frame-relay ip rtp header-compression
```

In this configuration example, the main interface has been configured as follows:

- Maximum Transmission Unit (MTU) size of IP packets is 300 bytes.
- No IP address is associated with this serial interface. The IP address must be assigned for the subinterface.
- Encapsulation method is Frame Relay.
- Fair queuing is enabled.
- IP RTP header compression is enabled.

The subinterface has been configured as follows:

- MTU size is inherited from the main interface.
- IP address for the subinterface is specified.
- Bandwidth is set to 64 kbps.
- Generic traffic shaping is enabled with 32 kbps CIR where Bc = 4000 bits and Be = 4000 bits.
- Frame Relay DLCI number is specified.
- IP RTP header compression is enabled.

**Note**

When traffic bursts over the CIR, the output rate is held at the speed configured for the CIR (for example, traffic will not go beyond 32 kbps if CIR is set to 32 kbps).

For more information about Frame Relay, refer to the “Configuring Frame Relay” chapter in the *Cisco IOS Wide-Area Networking Configuration Guide*.

VoIP for the Cisco 3600 Series Configuration Examples

The actual VoIP configuration procedure you complete depends on the topology of your voice network. The following configuration examples are a starting point. Of course, these configuration examples must be customized to reflect your network topology.

Configuration examples are supplied for the following sections:

- [FXS-to-FXS Connection Using RSVP, page 19](#)
- [Linking PBX Users with E&M Trunk Lines, page 22](#)
- [PSTN Gateway Access Using FXO Connection, page 24](#)
- [PSTN Gateway Access Using FXO Connection \(PLAR Mode\), page 25](#)

FXS-to-FXS Connection Using RSVP

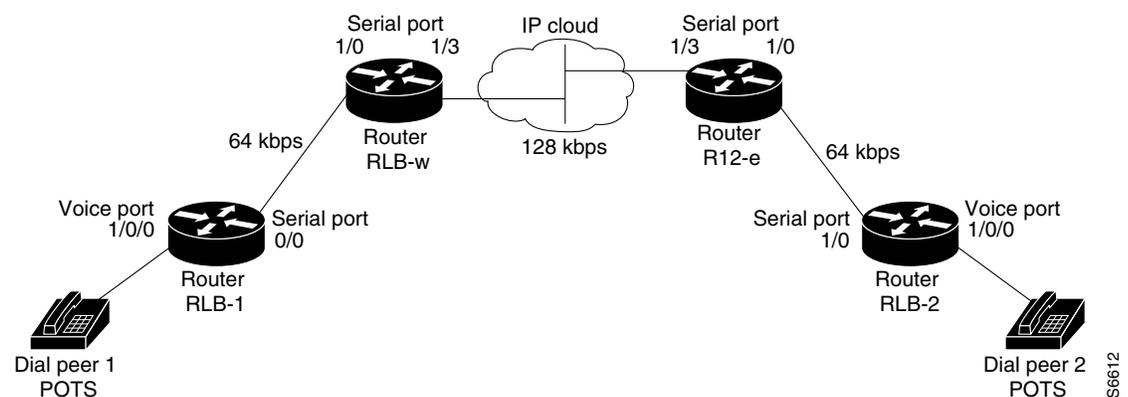
The following example shows how to configure VoIP for simple FXS-to-FXS connections.

In this example, a very small company of two offices has decided to integrate VoIP into its existing IP network. One basic telephony device is connected to Router RLB-1; therefore Router RLB-1 is configured for one POTS dial peer and one VoIP dial peer. Router RLB-w and Router R12-e establish the WAN connection between the two offices. Because one POTS telephony device is connected to Router RLB-2, it is also configured for only one POTS peer and one VoIP peer.

**Note**

In this example, only the calling end (Router RLB-1) is requesting RSVP. [Figure 4](#) illustrates the topology of this FXS-to-FXS connection example.

Figure 4 FXS-to-FXS Connection Example



Router RLB-1

```
hostname rlb-1

! Create voip dial peer 10
dial-peer voice 10 voip

! Define its associated telephone number and IP address
destination-pattern +4155554000
session target ipv4:40.0.0.1

! Request RSVP
req-qos guaranteed-delay

! Create pots dial peer 1
dial-peer voice 1 pots

! Define its associated telephone number and voice port
destination-pattern +4085554000
port 1/0/0

! Configure serial interface 0/0
interface Serial0/0
ip address 10.0.0.1 255.0.0.0
no ip mroute-cache

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25

! Enable RSVP on this interface
ip rsvp bandwidth 48 48
fair-queue 64 256 36
clockrate 64000

router igrp 888
network 10.0.0.0
network 20.0.0.0
network 40.0.0.0
```

Router RLB-w

```
hostname rlb-w

! Configure serial interface 1/0
interface Serial1/0
ip address 10.0.0.2 255.0.0.0

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25

! Enable RSVP on this interface
ip rsvp bandwidth 96 96
fair-queue 64 256 3

! Configure serial interface 1/3
interface Serial1/3
ip address 20.0.0.1 255.0.0.0

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25
```

```
! Enable RSVP on this interface
ip rsvp bandwidth 96 96
fair-queue 64 256 3

! Configure IGRP
router igrp 888
network 10.0.0.0
network 20.0.0.0
network 40.0.0.0
```

Router R12-e

```
hostname r12-e

! Configure serial interface 1/0
interface Serial1/0
ip address 40.0.0.2 25.0.0.0

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25

! Enable RSVP on this interface
ip rsvp bandwidth 96 96
fair-queue 64 256 3

! Configure serial interface 1/3
interface Serial1/3
ip address 20.0.0.2 255.0.0.0

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25

! Enable RSVP on this interface
ip rsvp bandwidth 96 96
fair-queue 64 256 3
clockrate 128000

! Configure IGRP
router igrp 888
network 10.0.0.0
network 20.0.0.0
network 40.0.0.0
```

Router RLB-2

```
hostname r1b-2

! Create pots dial peer 2
dial-peer voice 2 pots

! Define its associated telephone number and voice port
destination-pattern +4155554000
port 1/0/0

! Create voip dial peer 20
dial-peer voice 20 voip
!Define its associated telephone number and IP address
destination-pattern +4085554000
session target ipv4:10.0.0.1

! Configure serial interface 0/0
interface Serial0/0
```

```

ip address 40.0.0.1 255.0.0.0
no ip mroute-cache

! Configure RTP header compression
ip rtp header-compression
ip rtp compression-connections 25

! Enable RSVP on this interface
ip rsvp bandwidth 96 96
fair-queue 64 256 3
clockrate 64000

! Configure IGRP
router igrp 888
network 10.0.0.0
network 20.0.0.0
network 40.0.0.0

```

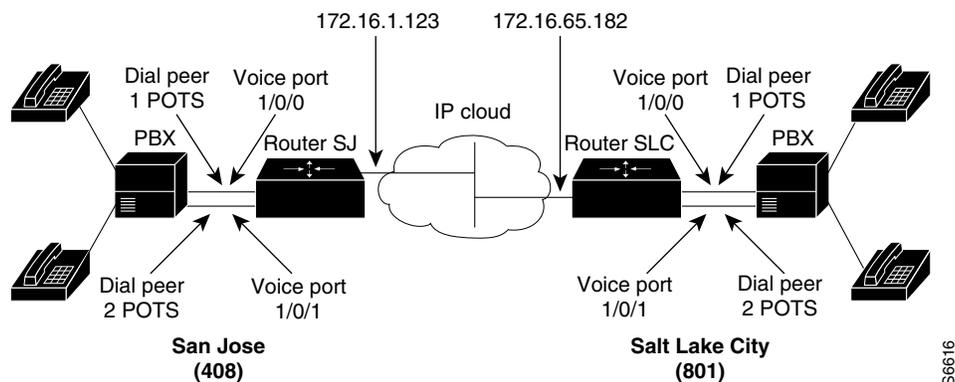
Linking PBX Users with E&M Trunk Lines

The following example shows how to configure VoIP to link PBX users with E&M trunk lines.

In this example, a company wants to connect two offices: one in San Jose, California, and the other in Salt Lake City, Utah. Each office has an internal telephone network using a PBX that is connected to the voice network by an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II with 4-wire operation and Immediate Start signaling. Each E&M interface connects to the router using two voice interface connections. Users in San Jose dial 8569 and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial 4527 and then the extension number to reach a destination in San Jose.

Figure 5 illustrates the topology of this connection example.

Figure 5 Linking PBX Users with E&M Trunk Lines Example



Note

This example assumes that the company already has working IP connection between its two remote offices.

Router SJ

```

hostname sanjose

!Configure pots dial peer 1
dial-peer voice 1 pots

```

```
destination-pattern 555....
port 1/0/0

!Configure pots dial peer 2
dial-peer voice 2 pots
destination-pattern 555....
port 1/0/1

!Configure voip dial peer 3
dial-peer voice 3 voip
destination-pattern 119....
session target ipv4:172.16.65.182

!Configure the E&M interface
voice-port 1/0/0
signal immediate
operation 4-wire
type 2

voice-port 1/0/1
signal immediate
operation 4-wire
type 2

!Configure the serial interface
interface serial 0/0
description serial interface type dce (provides clock)
clock rate 2000000
ip address 172.16.1.123
no shutdown
```

Router SLC

```
hostname saltlake

!Configure pots dial peer 1
dial-peer voice 1 pots
destination-pattern 119....
port 1/0/0

!Configure pots dial peer 2
dial-peer voice 2 pots
destination-pattern 119....
port 1/0/1

!Configure voip dial peer 3
dial-peer voice 3 voip
destination-pattern 555....
session target ipv4:172.16.1.123

!Configure the E&M interface
voice-port 1/0/0
signal immediate
operation 4-wire
type 2

voice-port 1/0/0
signal immediate
operation 4-wire
type 2

!Configure the serial interface
interface serial 0/0
description serial interface type dte
```

```
ip address 172.16.65.182
no shutdown
```

**Note**

PBXs should be configured to pass all DTMF signals to the router. We recommend that you do not configure store and forward tone.

**Note**

If you change the gain or the telephony port, make sure that the telephony port still accepts DTMF signals.

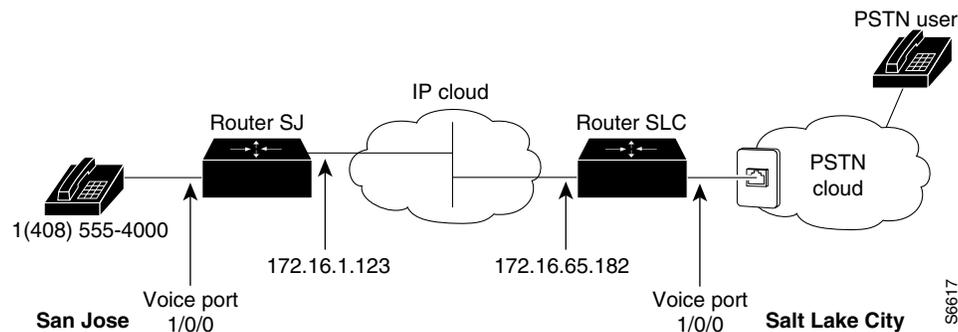
PSTN Gateway Access Using FXO Connection

The following example shows how to configure VoIP to link users with the PSTN gateway using an FXO connection.

In this example, users connected to Router SJ in San Jose, California, can reach PSTN users in Salt Lake City, Utah, via Router SLC. Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure 6 illustrates the topology of this connection example.

Figure 6 PSTN Gateway Access Using FXO Connection Example

**Note**

This example assumes that the company already has a working IP connection between its two remote offices.

Router SJ

```
! Configure pots dial peer 1
dial-peer voice 1 pots
destination-pattern +14085554000
port 1/0/0
```

```
! Configure voip dial peer 2
dial-peer voice 2 voip
destination-pattern 9.....
session target ipv4:172.16.65.182
```

```
! Configure the serial interface
interface serial 0/0
clock rate 2000000
```

```
ip address 172.16.1.123
no shutdown
```

Router SLC

```
! Configure pots dial peer 1
dial-peer voice 1 pots
destination-pattern 9.....
port 1/0/0
```

```
! Configure voip dial peer 2
dial-peer voice 2 voip
destination-pattern +14085554000
session target ipv4:172.16.1.123
```

```
! Configure serial interface
interface serial 0/0
ip address 172.16.65.182
no shutdown
```

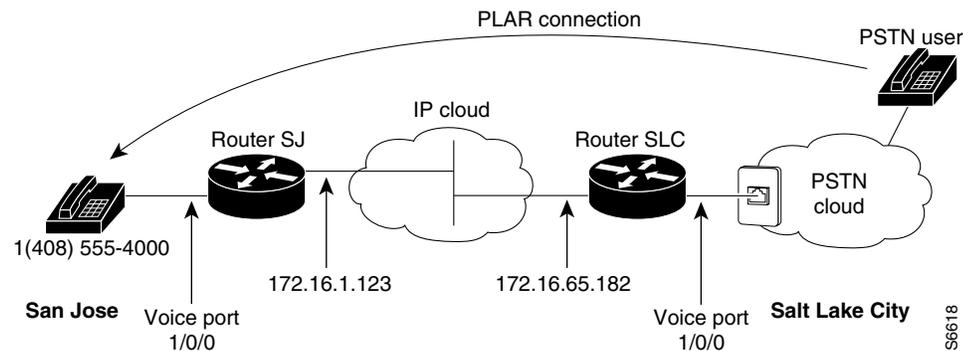
PSTN Gateway Access Using FXO Connection (PLAR Mode)

The following example shows how to configure VoIP to link users with the PSTN gateway using an FXO connection in private line auto-ringdown (PLAR) mode.

In this example, PSTN users in Salt Lake City, Utah, can dial a local number and establish a private-line connection in a remote location. As in the preceding example, Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure 7 illustrates the topology of this connection example.

Figure 7 PSTN Gateway Access Using FXO Connection (PLAR Mode)



Note

This example assumes that the company already has a working IP connection between its two remote offices.

Router SJ

```
! Configure pots dial peer 1
dial-peer voice 1 pots
destination-pattern +14085554000
port 1/0/0
```

```

! Configure voip dial peer 2
dial-peer voice 2 voip
  destination-pattern 9.....
  session target ipv4:172.16.65.182

! Configure the serial interface
interface serial 0/0
  clock rate 2000000
  ip address 172.16.1.123
  no shutdown

```

Router SLC

```

! Configure pots dial peer 1
dial-peer voice 1 pots
  destination-pattern 9.....
  port 1/0/0

! Configure voip dial peer 2
dial-peer voice 2 voip
  destination-pattern +14085554000
  session target ipv4:172.16.1.123

! Configure the voice-port
voice-port 1/0/0
  connection plar 14085554000

! Configure the serial interface
interface serial 0/0
  ip address 172.16.65.182
  no shutdown

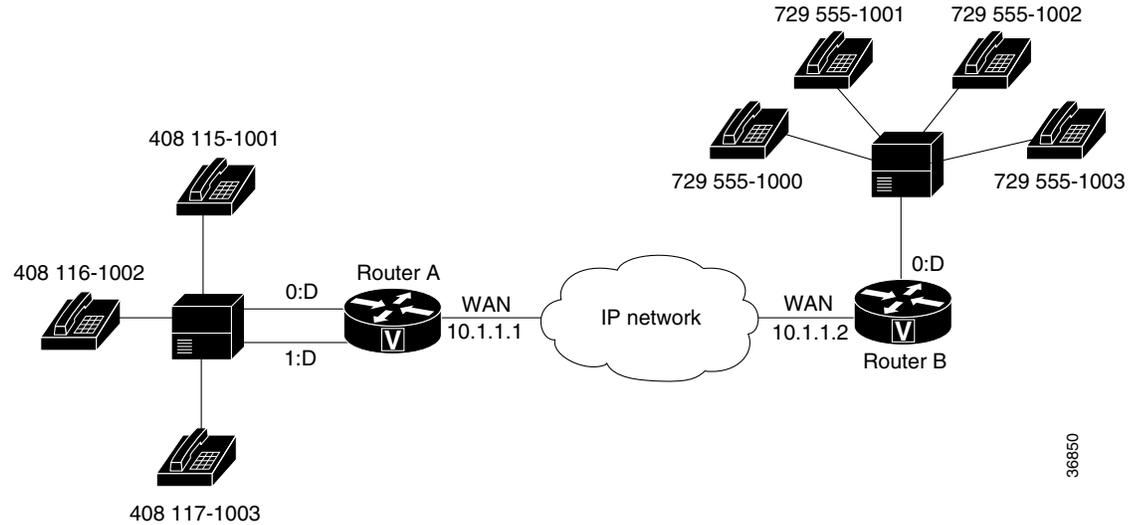
```

VoIP for the Cisco AS5300 Configuration Example

This configuration example should give you a starting point in your configuration process. The actual VoIP configuration procedure you complete depends on the topology of your voice network. These configuration examples must be customized to reflect your network topology.

Linking PBX Users to a T1 ISDN PRI Interface

This example describes how to configure VoIP to link PBX users with T1 channels configured for ISDN PRI signaling. In this example, the company has already established a working IP connection between its two remote offices, one in San Jose, California, and the other in Research Triangle Park (RTP), North Carolina. [Figure 8](#) illustrates the topology of this example.

Figure 8 Linking PBX Users to a T1 ISDN PRI Interface Example

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Each office has an internal telephone network using a PBX that is connected to the voice network by T1 interfaces. The San Jose office, located to the left of the IP cloud, has two T1 connections; the RTP office, located to the right of the IP cloud, has only one. Both offices are using PRI signaling for the T1 connections.

To reach a destination in RTP, callers in San Jose pick up the handset, hear a primary dial tone, and dial 9, 411, and the destination extension number. To reach a destination in San Jose, callers in RTP pick up the handset, hear a primary dial tone, and dial 4. After dialing 4, callers hear a secondary dial tone. They then dial 555 and the extension number.

Configuration for San Jose Access Server

The first part of this configuration example defines dial-in access, including configuring the T1 lines and the ISDN D-channel parameters:

```
hostname sanjose
!
! Define the telephone company's switch type
isdn switch-type primary-5ess
!
! Configure T1 PRI for line 1
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure T1 PRI for line 2
controller T1 1
 framing esf
 clock source line secondary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure the ISDN D channel for each ISDN PRI line
! Serial interface 0:23 is the D channel for controller T1 0
!
interface Serial0:23
 isdn incoming-voice modem
```

```
!
! Serial interface 1:23 is the D channel for controller T1 1
interface Serial1:23
 isdn incoming-voice modem
```

The next part of this example configures number expansion:

```
! Configure number expansion.
num-exp 555.... 1408555....
num-exp 4115... 17294115...
```

The next part of this example configures the POTS and VoIP dial peers:

```
! Configure POTS dial peer 1 using the first T1
dial-peer voice 1 pots
 prefix 6
 dest-pat 1408555....
 port 0:D
!
! Configure POTS dial-peer 2 using the first T1
dial-peer voice 2 pots
 prefix 7
 dest-pat 1408555....
 port 0:D
!
! Configure POTS dial-peer 3 using the second T1
dial-peer voice 3 pots
 prefix 5
 dest-pat 1408555....
 port 1:D
!
! Configure VoIP dial-peer 4
dial-peer voice 4 voip
 dest-pat 17294115...
 session-target ipv4:10.1.1.2
```

Configuration for RTP Access Server

The first part of this configuration example defines dial-in access, including configuring the T1 line and the ISDN D-channel parameters:

```
hostname rtp

! Define the telephone company's switch type
isdn switch-type primary-5ess

! Configure T1 PRI for line 1
controller T1 0
 framing esf
 clock source line primary
 linecode b8zs
 pri-group timeslots 1-24
!
! Configure the ISDN D channel for ISDN PRI line 1
! Serial interface 0:23 is the D channel for controller T1 0
interface Serial0:23
 ip address 7.1.1.10 255.255.255.0
 encapsulation ppp
 isdn incoming-voice modem
 dialer-group 1
 ppp authentication chap
```

The next part of this example configures number expansion:

```
! Configure number expansion.
num-exp 555.... 1408555....
num-exp 4115... 17294115...
```

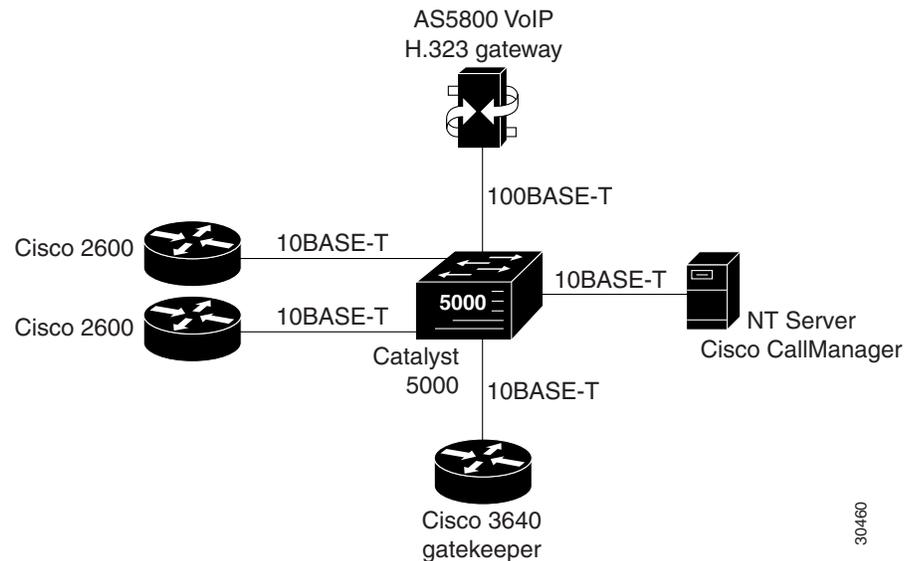
The next part of this configuration example defines the POTS and VoIP peers:

```
! Configure POTS dial-peer 1
dial-peer voice 1 pots
  dest-pat 17294115...
  port 0:D
!
! Configure VoIP dial-peer 5
dial-peer voice 4 voip
  dest-pat 1408555....
  session-target ipv4:10.1.1.1
```

VoIP for the Cisco AS5800 Configuration Example

The following configuration example shows an abbreviated configuration using a Cisco 2600 router and a Cisco AS5800 universal access server as gateways and a Cisco 3600 router as a gatekeeper. [Figure 9](#) shows the network diagram for this particular scenario.

Figure 9 Cisco AS5800 Universal Access Server Acting As a Gateway



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Configuring the Cisco 3640 As a Gatekeeper

The following example shows how to configure a Cisco 3640 router as a gatekeeper:

```
! Configure the Ethernet interface to be used at the gatekeeper interface.
interface Ethernet0/1
  ip address 172.30.00.00 255.255.255.0
  no ip directed-broadcast
  no logging event link-status
  no keepalive
```

```

!
! Configure the gatekeeper interface and enable the interface.
gatekeeper
  zone local gk3.gg-dn1 gg-dn1 173.50.00.00
  zone prefix gk3.gg-dn1 21*
  gw-type-prefix 9#* gw ipaddr 173.60.0.0 1720
  gw-type-prefix 6#* gw ipaddr 173.60.0.199 1720
  no use-proxy gk3.gg-dn1 default inbound-to terminal
  no shutdown
!

```

Configuring the Cisco 2600 As a Gateway

The following example shows how to configure a Cisco 2600 series router as a gateway:

```

! Configure POTS and VoIP dial peers.
dial-peer voice 88 voip
  destination-pattern 11111
  tech-prefix 9#
  session ras
!
dial-peer voice 11 pots
  incoming called-number 11111
  destination-pattern 6#12345
port 1/1/1
prefix 12345
!
! Configure the gateway interface.
interface Ethernet0/0
ip address 173.60.0.199 255.255.255.0
no ip directed-broadcast
no ip mroute-cache
no logging event link-status
no keepalive
no cdp enabled
h323-gateway voip interface
h323-gateway voip id gk3.gg-dn1 ipaddr 173.30.0.0 1719
h323-gateway voip h323-id gw6@gg-dn1
h323-gateway voip tech-prefix 6#
!

```

Configuring the Cisco AS5800 as a Gateway

The following example shows how to configure the Cisco AS5800 universal access server as a gateway:

```

! Configure the T1 controller. (This configuration is for a T3 card.)
controller T1 1/0/0:1
  framing esf
  linecode b8zs
  pri-group timeslots 1-24
!
! Configure POTS and VoIP dial peers.
dial-peer voice 11111 pots
  incoming called-number 12345
  destination-pattern 9#11111
  direct-inward-dial
  port 1/0/0:1:D
  prefix 11111
!
dial-peer voice 12345 voip
  destination-pattern 12345
  tech-prefix 6#

```

```
session target ras
!
! Enable gateway functionality.
gateway
!
! Enable Cisco Express Forwarding.
ip cef
!
! Configure and enable the gateway interface.
interface FastEthernet0/3/0
 ip address 173.60.0.0.255.255.255.0
 no ip directed-broadcast
 no keepalive
 full-duplex
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id gk3.gg-dn1 ipaddr 173.30.0.0 1719
 h323-gateway voip h323-id gw3@gg-dn1
 h323-gateway voip tech-prefix 9#
!
! Configure the serial interface.(This configuration is for a T3 serial interface.)
interface Serial1/0/0:1:23
 no ip address
 no ip directed-broadcast
 ip mroute-cache
 isdn switch-type primary-5ess
 isdn incoming-voice modem
 no cdp enable
```

