CVOICE

Cisco Voice Over IP

Version 4.2

Student Guide

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Table of Contents

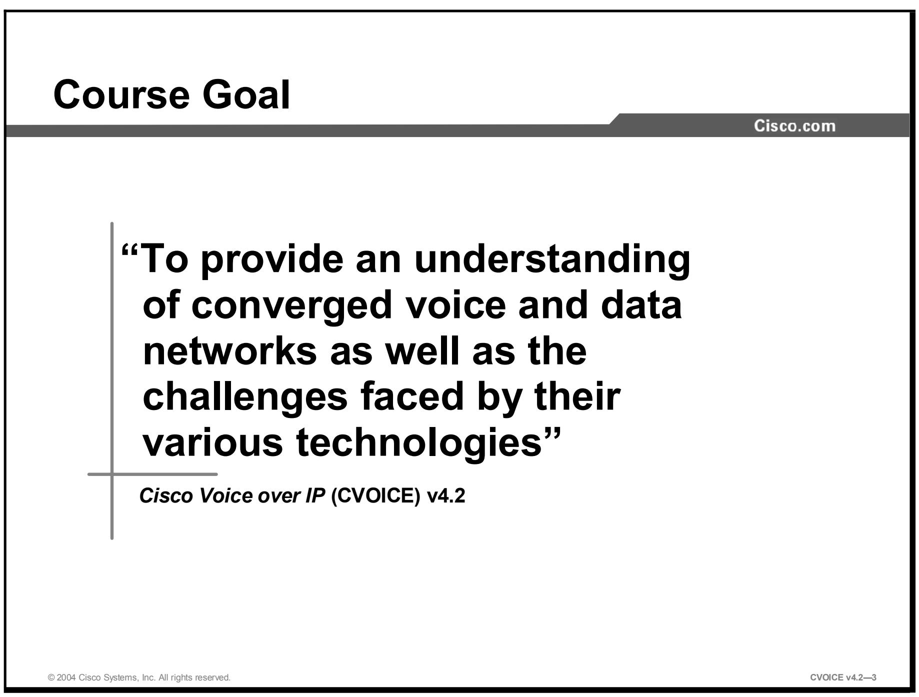
Course Introduction

Overview

The *Cisco Voice over IP* (CVOICE) course provides an understanding of converged voice and data networks as well as the challenges faced by their various technologies. The course presents Cisco Systems solutions and implementation considerations to address those challenges.

Course Goal and Objectives

This section describes the course goal and objectives.



Upon completing this course, you will be able to meet these objectives:

- Describe the similarities and differences between traditional PSTN voice networks and IP telephony solutions
- Explain the processes and standards for voice digitization, compression, digital signaling, and fax transport as they relate to VoIP networks
- Configure voice interfaces on Cisco voice-enabled equipment for connection to traditional, nonpacketized telephony equipment
- Configure the call flows for POTS, VoIP, and default dial peers
- Describe the fundamentals of VoIP and identify challenges and solutions regarding its implementation
- Compare centralized and decentralized call control and signaling protocols
- Describe specific voice quality issues and the QoS solutions used to solve them

Course Outline

The outline lists the modules included in this course.

Course Outline

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- **Introduction to Packet Voice Technologies**
- **Analog and Digital Voice Connections**
- **Configuring Voice Interfaces**
- **Voice Dial Plans**
- **Introduction to VoIP**
- **VoIP Signaling and Call Control**
- **Improving and Maintaining Voice Quality**

CVOICE v4.2—4

Cisco Certifications

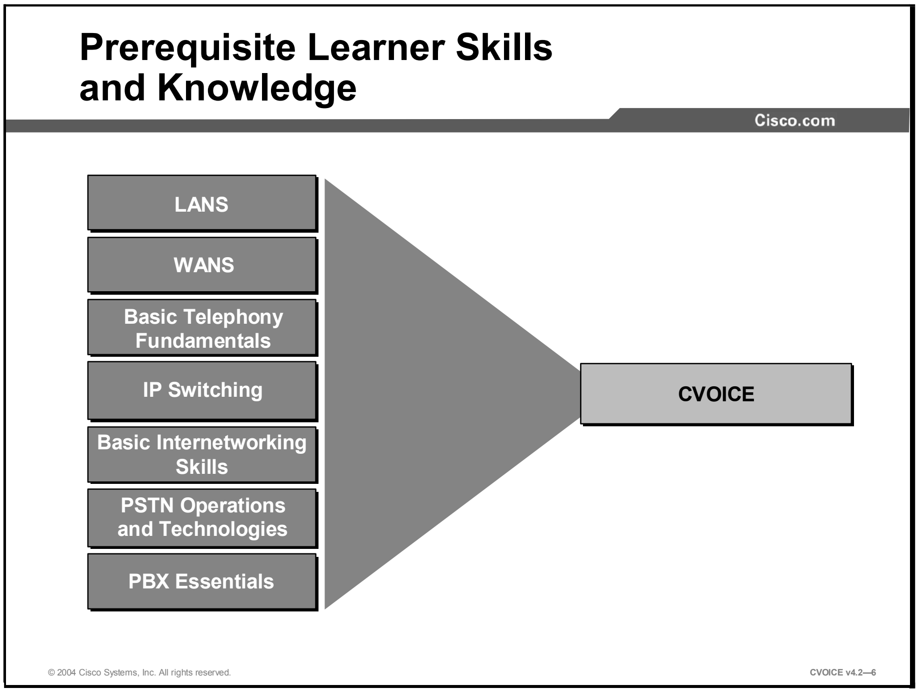
This topic lists the certification requirements of this course.

Cisco provides three levels of general career certifications for IT professionals with several different tracks to meet individual needs. Cisco also provides a variety of Cisco Qualified Specialist (CQS) certifications, which enable learners to demonstrate knowledge in specific technologies, solutions, or job roles. In contrast to general certifications, each CQS certification is focused on a designated area such as cable communications, voice, or security. All CQS certifications are customized to meet current market needs. They may also have special focused prerequisite requirements.

There are many paths to Cisco certification, but only one requirement—passing one or more exams demonstrating knowledge and skill. For details, go to http://www.cisco.com/go/certifications.

Learner Skills and Knowledge

This topic lists the course prerequisites.

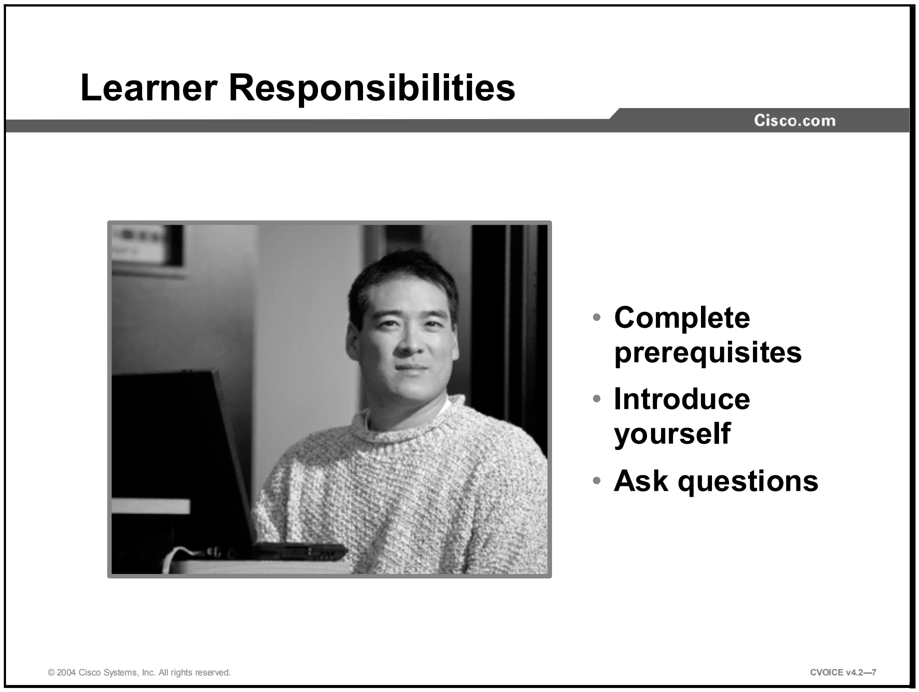


To benefit fully from this course, you must have these prerequisite skills and knowledge:

- A working knowledge of LANs, WANs, and IP switching and routing
- Basic internetworking skills taught in the *Interconnecting Cisco Network Devices* (ICND) course, or its equivalent
- Knowledge of traditional public switched telephone network (PSTN) operations and voice fundamentals

Learner Responsibilities

This topic discusses the responsibilities of the learners.



To take full advantage of the information that is presented in this course, you must have completed the prerequisite requirements.

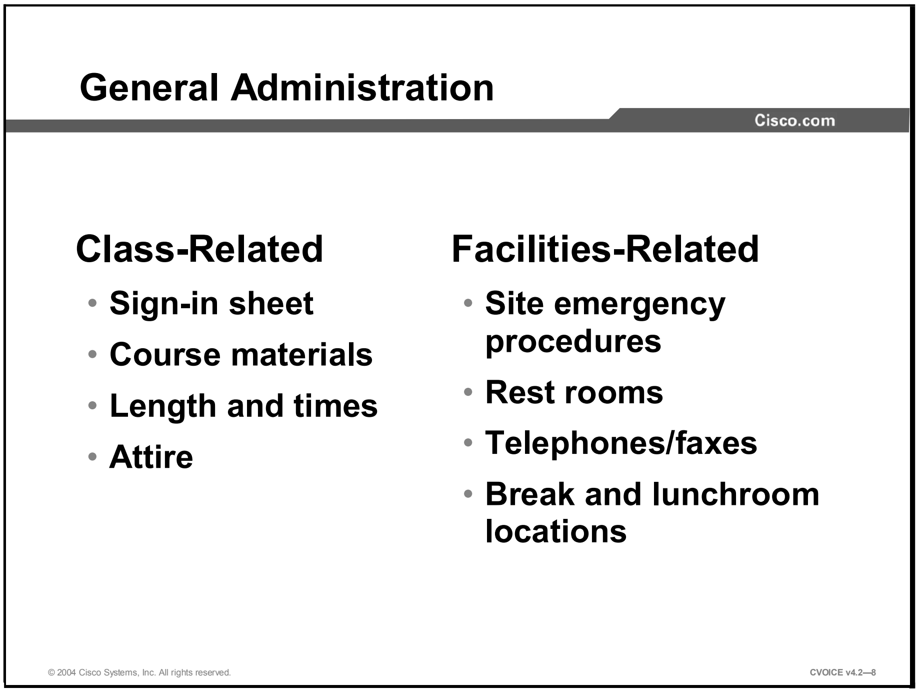
In class, you are expected to participate in all lesson exercises and assessments.

In addition, you are encouraged to ask any questions that are relevant to the course materials.

If you have pertinent information or questions concerning future Cisco product releases and product features, please discuss these topics during breaks or after class. The instructor will answer your questions or direct you to an appropriate information source.

General Administration

This topic lists the administrative issues for the course.



The instructor will discuss these administrative issues:

- Sign-in process
- Starting and anticipated ending times of each class day
- Class breaks and lunch facilities
- Appropriate attire during class
- Materials that you can expect to receive during class
- What to do in the event of an emergency
- Location of the rest rooms
- How to send and receive telephone and fax messages

Course Flow Diagram

This topic covers the suggested flow of the course materials.

Course Flow Diagram

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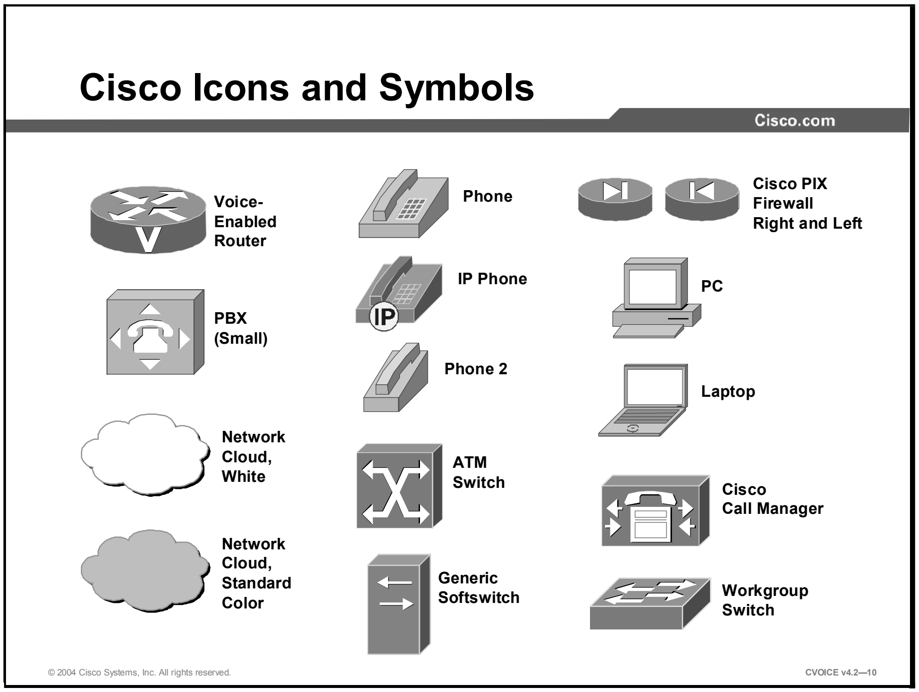
	Day 1	Day 2	Day 3	Day 4	Day 5
AM	Course Introduction; Intro to Packet Voice Technologies	Analog and Digital Voice Connections (Cont.); Configuring Voice Interfaces	Voice Dial Plans (Cont.)	Introduction to VoIP (Cont.); VoIP Signaling and Call Control	VoIP Signaling and Call Control (Cont.); Improving and Maintaining Voice Quality
	Lunch				
PM	Analog and Digital Voice Connections	Configuring Voice Interfaces (Cont.); Voice Dial Plans	Introduction to VoIP	VoIP Signaling and Call Control (Cont.)	Improving and Maintaining Voice Quality (Cont.)

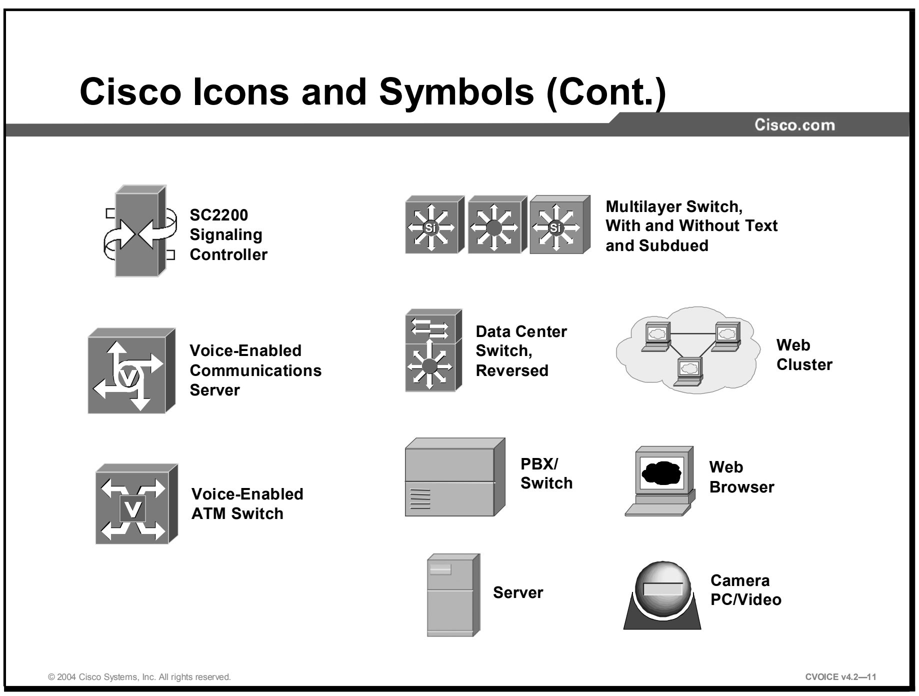
CVOICE v4.2—9

The schedule reflects the recommended structure for this course. This structure allows enough time for the instructor to present the course information and for you to work through the laboratory exercises. The exact timing of the subject materials and labs depends on the pace of your specific class.

Icons and Symbols

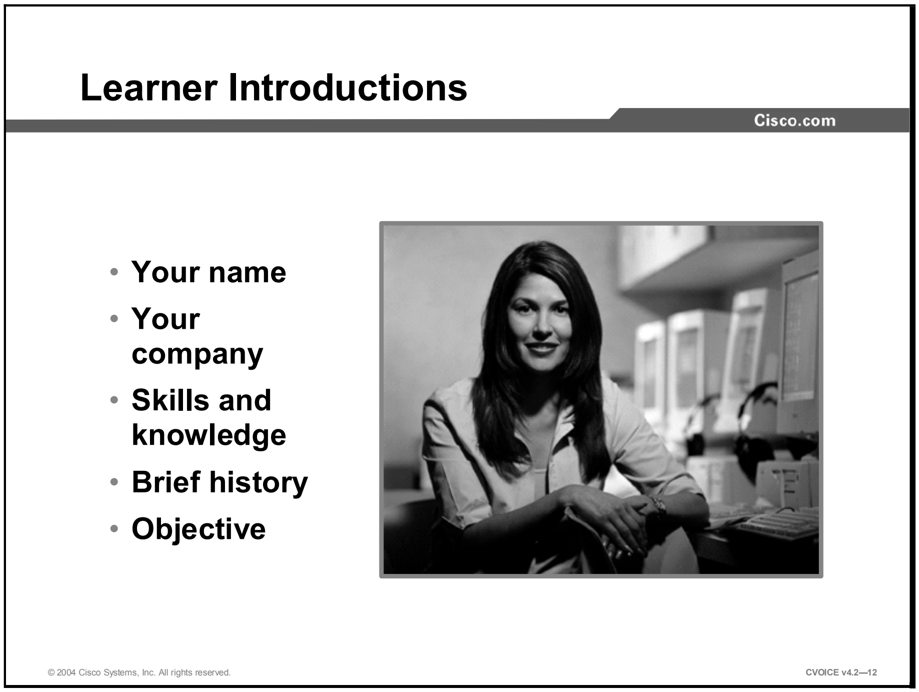
This topic shows the Cisco icons and symbols used in this course.





Learner Introductions

This is the point in the course where you introduce yourself.



Prepare to share the following information:

- Your name
- Your company
- If you have most or all of the prerequisite skills
- A profile of your experience
- What you would like to learn from this course

Module 1

Introduction to Packet Voice Technologies

Overview

Voice over IP (VoIP) is forecasted to have explosive growth in the coming months and years. Many corporate environments have migrated, are actively migrating, or are researching the process of migrating to VoIP. Some long-distance providers are using VoIP to carry voice traffic, particularly on international calls.

Migration is a process that involves gradually phasing out old components and replacing them with new ones. Many terms have been used to describe the technologies and applications for transporting voice in a converged packet network environment. When designing a converged network, it is necessary to clearly define all requirements and understand the various options that are available.

An important first step in designing a converged network is to understand the traditional telephony network and how it interfaces with voice components. You must know, from the start, how legacy voice equipment is connected and its possible migration paths.

The next step toward a good design is being knowledgeable about the components available for Packet Telephony Networks. You should be aware of the difference between voice and data flow within the network and the tools for controlling voice calls. Network requirements vary according to the location size. Knowing the difference between campus, enterprise, and service provider environments is crucial in choosing the right components and technologies.

This module provides an overview of the basic telephony functions and devices, including PBXs, switching functions, call signaling, and multiplexing techniques. It also reviews the basic components of the Packet Telephony Network and identifies the different requirements in campus, enterprise, and service provider environments. Together, these concepts and techniques provide a solid introduction to the VoIP arena.

Module Objectives

Upon completing this module, you will be able to describe the similarities and differences between traditional public switched telephone network (PSTN) voice networks and IP telephony solutions.

Module Objectives

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- Identify the components, processes, and features of traditional telephony networks that provide end-to-end call functionality
- Describe two methods of call control used on voice and data networks and provide one protocol example for each
- List five components or capabilities that are required to provide integrated voice and data services in campus LAN, enterprise, and service provider environments

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Module Outline

The outline lists the components of this module.

Module Outline

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- Understanding Traditional Telephony
- Understanding Packetized Telephony Networks
- Understanding IP Telephony Applications

 CVOICE v4.2—1-3

Understanding Traditional Telephony

Overview

This lesson describes the components that make up a traditional telephony network and the process of passing calls through the network.

Relevance

In traditional telephony networks, many components and processes are transparent to the customer. As you move from traditional telephony networks to converged voice and data networks, you must manage new components and processes to ensure seamless end-to-end call handling. To maintain acceptable service levels, you must understand which devices must now be supported and the processes that are necessary to ensure end-to-end call functionality.

Objectives

Upon completing this lesson, you will be able to identify the components, processes, and features of traditional telephony networks that provide end-to-end call functionality. This includes being able to meet these objectives:

- Describe the components and functionality of traditional telephony networks
- Explain how CO switches process telephone calls
- Identify types of private switching systems used in traditional telephony networks and list the main features of each
- Describe the three types of signaling in traditional telephony networks and identify how each is used
- Describe two methods used to multiplex voice in traditional telephony networks

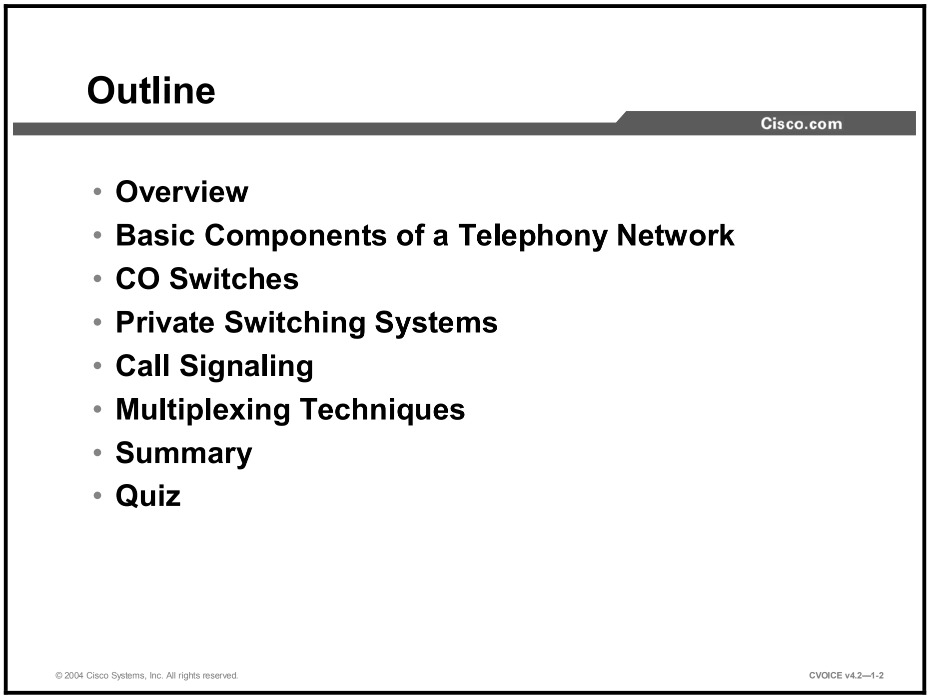
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- General knowledge of telephony technology, including customer premises equipment (CPE) and the PSTN

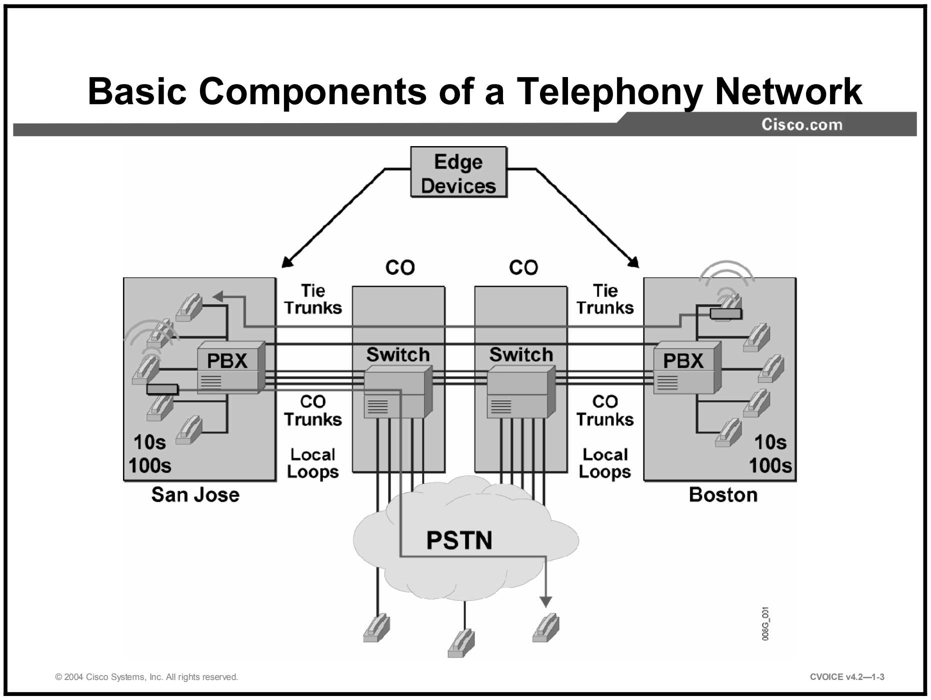
Outline

The outline lists the topics included in this lesson.



Basic Components of a Telephony Network

This topic introduces the components of traditional telephony networks.



A number of components must be in place for an end-to-end call to succeed. These components are shown in the figure and include the following:

- Edge devices
- Local loops
- Private or central office (CO) switches
- Trunks

Edge Devices

The two types of edge devices that are used in a telephony network include:

- **Analog telephones:** Analog telephones are most common in home, small office/home office (SOHO), and small business environments. Direct connection to the PSTN is usually made by using analog telephones. Proprietary analog telephones are occasionally used in conjunction with a PBX. These telephones provide additional functions such as speakerphone, volume control, PBX message-waiting indicator, call on hold, and personalized ringing.
- **Digital telephones:** Digital telephones contain hardware to convert analog voice into a digitized stream. Larger corporate environments with PBXs generally use digital telephones. Digital telephones are typically proprietary, meaning that they work with the PBX or key system of that vendor only.

Local Loops

A local loop is the interface to the telephone company network. Typically, it is a single pair of wires that carry a single conversation. A home or small business may have multiple local loops.

Private or CO Switches

The CO switch terminates the local loop and handles signaling, digit collection, call routing, call setup, and call teardown.

A PBX switch is a privately owned switch located at the customer site. A PBX typically interfaces with other components to provide additional services, such as voice mail.

Trunks

The primary function of a trunk is to provide the path between two switches. There are several common trunk types, including:

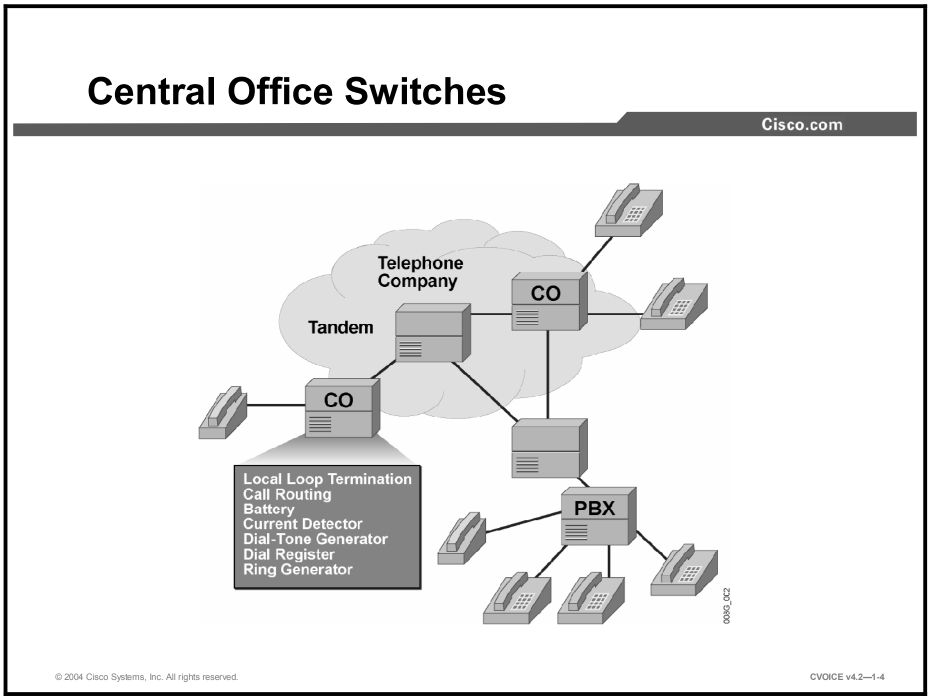
- **Tie trunk:** A dedicated circuit that connects PBXs directly
- **CO trunk:** A direct connection between a local CO and a PBX
- **Interoffice trunk:** A circuit that connects two local telephone company COs

Example: Telephony Components

The telephone installed in your home is considered an edge device because it terminates the service provided by your local telephone company. PBXs or key systems installed in a business would also be considered edge devices. The local loop is the pair of wires that come to your house to provide residential telephone service. Trunks are the interconnections between telephone switches. They can be between private switches or telephone company switches.

CO Switches

This topic describes how CO switches function and make switching decisions.



The figure shows a typical CO switch environment. The CO switch terminates the local loop and makes the initial call-routing decision.

The call-routing function forwards the call to one of the following:

- Another end-user telephone, if it is connected to the same CO
- Another CO switch
- A tandem switch

The CO switch makes the telephone work with the following components:

- **Battery:** The battery is the source of power to both the circuit and the telephone. It determines the status of the circuit. When the handset is lifted to let current flow, the telephone company provides the source that powers the circuit and the telephone. Because the telephone company powers the telephone from the CO, electrical power outages should not affect the basic telephone.

Note Some telephones on the market offer additional features that require a supplementary power source that the subscriber supplies; for example, cordless telephones. Some cordless telephones may lose function during a power outage.

- **Current detector:** The current detector monitors the status of a circuit by detecting whether it is open or closed. The table here describes current flow in a typical telephone.

Current Flow in a Typical Telephone

Handset	Circuit	Current Flow
On cradle	On hook/open circuit	No
Off cradle	Off hook/closed circuit	Yes

- **Dial-tone generator:** When the digit register is ready, the dial-tone generator produces a dial tone to acknowledge the request for service.
- **Dial register:** The digit register receives the dialed digits.
- **Ring generator:** When the switch detects a call for a specific subscriber, the ring generator alerts the called party by sending a ring signal to that subscriber.

You must configure a PBX connection to a CO switch that matches the signaling of the CO switch. This configuration ensures that the switch and the PBX can detect on hook, off hook, and dialed digits coming from either direction.

CO Switching Systems

Switching systems provide three primary functions:

- Call setup, routing, and teardown
- Call supervision
- Customer ID and telephone numbers

CO switches switch calls between locally terminated telephones. If a call recipient is not locally connected, the CO switch decides where to send the call based on its call-routing table. The call then travels over a trunk to another CO or to an intermediate switch that may belong to an interexchange carrier (IXC). Although intermediate switches do not provide dial tone, they act as hubs to connect other switches and provide interswitch call routing.

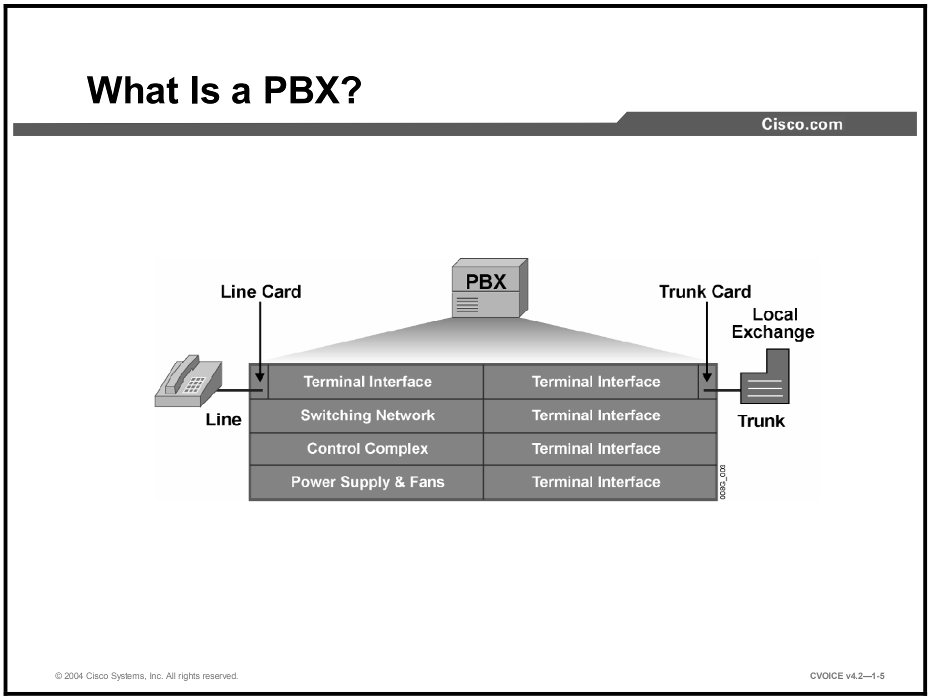
PSTN calls are traditionally circuit-switched, which guarantees end-to-end path and resources. Therefore, as the PSTN sends a call from one switch to another, the same resource is associated with the call until the call is terminated.

Example: CO Switches

CO switches provide local service to your residential telephone. The CO switch provides dial tone, indicating that the switch is ready to receive digits. When you dial your phone, the CO switch receives the digits, then routes your call. The call routing may involve more than one switch as the call progresses through the network.

Private Switching Systems

In a corporate environment, where large numbers of staff need access to each other and the outside, individual telephone lines are not economically viable. This topic explores PBX and key telephone system functionality in environments today.



A PBX is a smaller, privately owned version of the CO switches used by telephone companies.

Most businesses have a PBX telephone system, a key telephone system, or Centrex service. Large offices with more than 50 telephones or handsets choose a PBX to connect users, both in-house and to the PSTN.

PBXs come in a variety of sizes, from 20 to 20,000 stations. The selection of a PBX is important to most companies because a PBX has a typical life span of seven to ten years.

All PBXs offer a standard, basic set of calling features. Optional software provides additional capabilities.

The figure illustrates the internal components of a PBX. It connects to telephone handsets using line cards and to the local exchange using trunk cards.

A PBX has three major components:

- **Terminal interface:** The terminal interface provides the connection between terminals and PBX features that reside in the control complex. Terminals can include telephone handsets, trunks, and lines. Common PBX features include dial tone and ringing.
- **Switching network:** The switching network provides the transmission path between two or more terminals in a conversation. For example, two telephones within an office communicate over the switching network.
- **Control complex:** The control complex provides the logic, memory, and processing for call setup, call supervision, and call disconnection.

Example: PBX Installations

PBX switches are installed in large business campuses to relieve the public telephone company switches from having to switch local calls. When you call a coworker locally in your office campus, the PBX switches the call locally instead of having to rely on the public CO switch. The existence of PBX switches also limits the number of trunks needed to connect to the telephone company's CO switch. With a PBX installed, every office desktop telephone does not need its own trunk to the CO switch. Rather, the trunks are shared among all users.

What Is a Key System?

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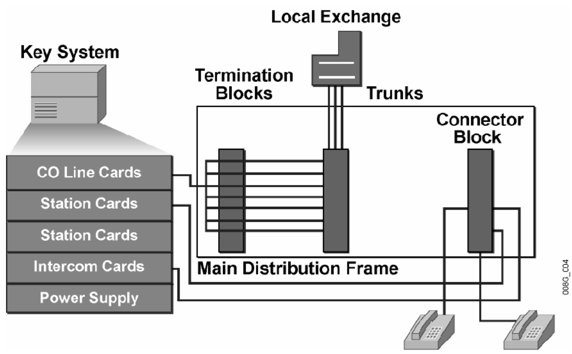

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Small organizations and branch offices often use a key telephone system, because a PBX offers functionality and extra features that they may not require. A key system offers small businesses distributed answering from any telephone, unlike the central answering position required for a PBX.

Today, key telephone systems are either analog or digital and are microprocessor-based. Key systems are typically used in offices with 30 to 40 users, but can be scaled to support over 100 users.

A key system has three major components:

- **Key service unit:** A key service unit (KSU) holds the system switching components, power supply, intercom, line and station cards, and the system logic.
- **System software:** System software provides the operating system and calling-feature software.
- **Telephones (instruments or handsets):** Telephones allow the user to choose a free line and dial out, usually by pressing a button on the telephone.

Comparing Key Systems with PBXs

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	PBX	Key System
Technology	Primarily digital	Analog or digital
Switch Functionality	Similar to the CO switch	Not a switch
Typical Installation	Large company site (typically more than 50 users)	Small company or branch office (typically 50 or fewer users)
Method for Accessing Outside Trunks	Dial 9 or other access number to access outside line	Press a button to access outside line

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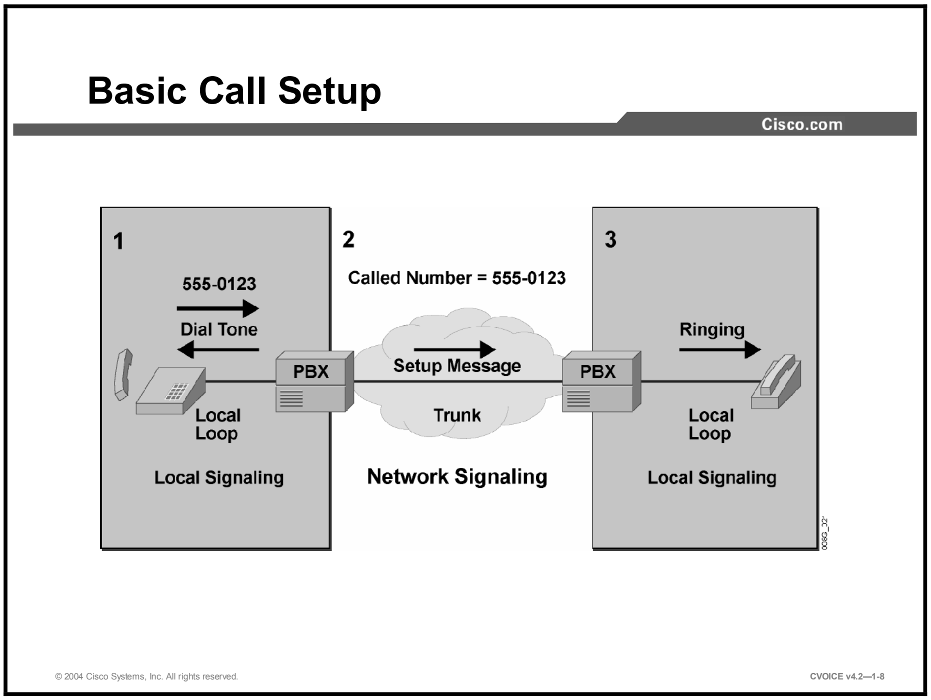
Larger companies use proprietary telephone networks with PBXs. In a key telephone system, each telephone has multiple lines that allow users to access outside lines to their CO. When a call comes into the company, a line or a key lights up on the telephone and indicates that a particular line is in use. Users can call another extension or let another person know where to pick up a call by using an intercom function, such as an overhead paging system or speakerphone.

Key telephone system functionality has evolved over time to include a class called hybrid telephone systems. The hybrid system adds many features that were previously available only in PBXs. There is no single definition of the functions and features that are classified as a hybrid system because all vendors provide a mix that they believe gives them a competitive advantage.

The main difference between a key telephone system and a hybrid telephone system is whether a single-line telephone can access a single CO local loop or trunk (key telephone system) only, or whether the single-line telephone can access a pool of CO local loops or trunks (hybrid telephone system).

Call Signaling

Call signaling, in its most basic form, is the capacity of a user to communicate a need for service to a network. The call-signaling process requires the ability to detect a request for service and termination of service, send addressing information, and provide progress reports to the initiating party. This functionality corresponds to the three call-signaling types discussed in this topic: supervisory, address, and informational signaling.

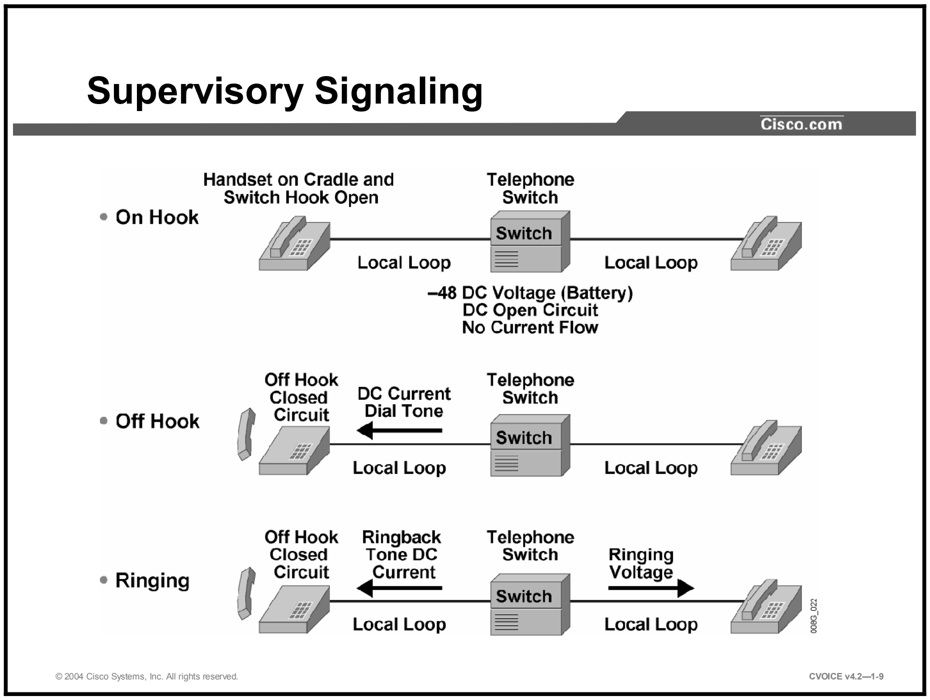


The figure shows the three major steps in an end-to-end call. These steps include:

1. **Local signaling — originating side:** The user signals the switch by going off hook and sending dialed digits through the local loop.

2. **Network signaling:** The switch makes a routing decision and signals the next, or terminating, switch through the use of setup messages sent across a trunk.

3. **Local signaling — terminating side:** The terminating switch signals the call recipient by sending ringing voltage through the local loop to the recipient telephone.



A subscriber and telephone company notify each other of call status with audible tones and an exchange of electrical current. This exchange of information is called supervisory signaling.

There are three different types of supervisory signaling:

- **On hook:** When the handset rests on the cradle, the circuit is on hook. The switch prevents current from flowing through the telephone. Regardless of the signaling type, a circuit goes on hook when the handset is placed on the telephone cradle and the switch hook is toggled to an open state. This prevents the current from flowing through the telephone. Only the ringer is active when the telephone is in this position.
- **Off hook:** When the handset is removed from the telephone cradle, the circuit is off hook. The switch hook toggles to a closed state, causing circuit current to flow through the electrical loop. The current notifies the telephone company equipment that someone is requesting to place a telephone call. When the telephone network senses the off-hook connection by the flow of current, it provides a signal in the form of a dial tone to indicate that it is ready.
- **Ringing:** When a subscriber makes a call, the telephone sends voltage to the ringer to notify the other subscriber of an inbound call. The telephone company also sends a ringback tone to the caller alerting the caller that it is sending ringing voltage to the recipient telephone. Although the ringback tone sounds similar to ringing, it is a call-progress tone and not part of supervisory signaling.

Note The ringing tone in the United States is 2 seconds of tone followed by 4 seconds of silence. Europe uses a double ring followed by 2 seconds of silence.

Address Signaling

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- Tone telephone
  - DTMF dialing
- Rotary telephone
  - Pulse dialing

CVOICE v4.2—1-10

There are two types of telephones: a rotary-dial telephone and a push-button (tone) telephone. These telephones use two different types of address signaling to notify the telephone company where a subscriber is calling:

- **Dual tone multifrequency:** Each button on the keypad of a touch-tone pad or push-button telephone is associated with a set of high and low frequencies. On the keypad, each row of keys is identified by a low-frequency tone and each column is associated with a high-frequency tone. The combination of both tones notifies the telephone company of the number being called, thus the term “dual tone multifrequency” (DTMF).
- **Pulse:** The large numeric dial-wheel on a rotary-dial telephone spins to send digits to place a call. These digits must be produced at a specific rate and within a certain level of tolerance. Each pulse consists of a “break” and a “make,” which are achieved by opening and closing the local loop circuit. The break segment is the time during which the circuit is open. The make segment is the time during which the circuit is closed. The break-and-make cycle must correspond to a ratio of 60 percent break to 40 percent make.

  A governor inside the dial controls the rate at which the digits are pulsed; for example, when a subscriber calls someone by dialing a digit on the rotary dial, a spring winds. When the dial is released, the spring rotates the dial back to its original position. While the spring rotates the dial back to its original position, a cam-driven switch opens and closes the connection to the telephone company. The number of consecutive opens and closes, or breaks and makes, represents the dialed digit.

Informational Signaling

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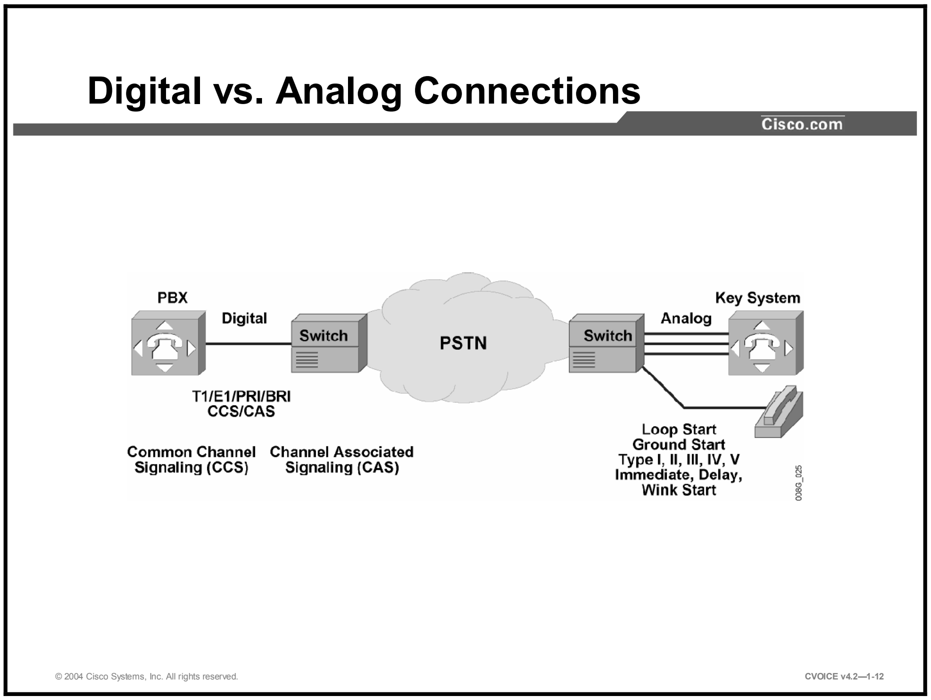
Tone	Frequency (Hz)	On Time (Sec)	Off Time (Sec)
Dial	350 + 440	Continuous	Continuous
Busy	480 + 620	0.5	0.5
Ringback, line	440 + 480	2	4
Ringback, PBX	440 + 480	1	3
Congestion (toll)	480 + 620	0.2	0.3
Reorder (local)	480 + 620	0.3	0.2
Receiver off hook	(1400 + 2060 + 2450 + 2600)	0.1	0.1
No such number	200 to 400	Continuous	Continuous
Confirmation tone		Freq. Mod. 1 kHz	Freq. Mod. 1 kHz

CVOICE v4.2—1-11

Tone combinations indicate call progress and are used to notify subscribers of call status. Each combination of tones represents a different event in the call process. These events include the following:

- **Dial tone:** Indicates that the telephone company is ready to receive digits from the user telephone
- **Busy:** Indicates that a call cannot be completed because the telephone at the remote end is already in use
- **Ringback (normal or PBX):** Indicates that the telephone company is attempting to complete a call on behalf of a subscriber
- **Congestion:** Indicates that congestion in the long-distance telephone network is preventing a telephone call from being processed
- **Reorder tone:** Indicates that all the local telephone circuits are busy, thus preventing a telephone call from being processed
- **Receiver off hook:** Indicates that a receiver has been off hook for an extended period of time without placing a call
- **No such number:** Indicates that a subscriber has placed a call to a nonexistent number
- **Confirmation tone:** Indicates that the telephone company is attempting to complete a call



Supervisory, address, and informational signaling must be carried across both analog and digital connections. Depending on your connection to the network, you must configure specific signaling to match the type of signaling required by the service provider.

Digital PBX connections to the network are common in many countries. They may be a T1 or E1 line carrying channel associated signaling (CAS) or a PRI using common channel signaling (CCS).

CAS is a signaling method that allows passing on-hook or off-hook status by setting bits that are associated with each specific voice channel. These bits are carried in band for T1 and out of band for E1.

An ISDN connection uses the D channel as the common channel to carry signaling messages for all other channels. CCS carries the signaling out of band, meaning that the signaling and the voice path do not share the same channel.

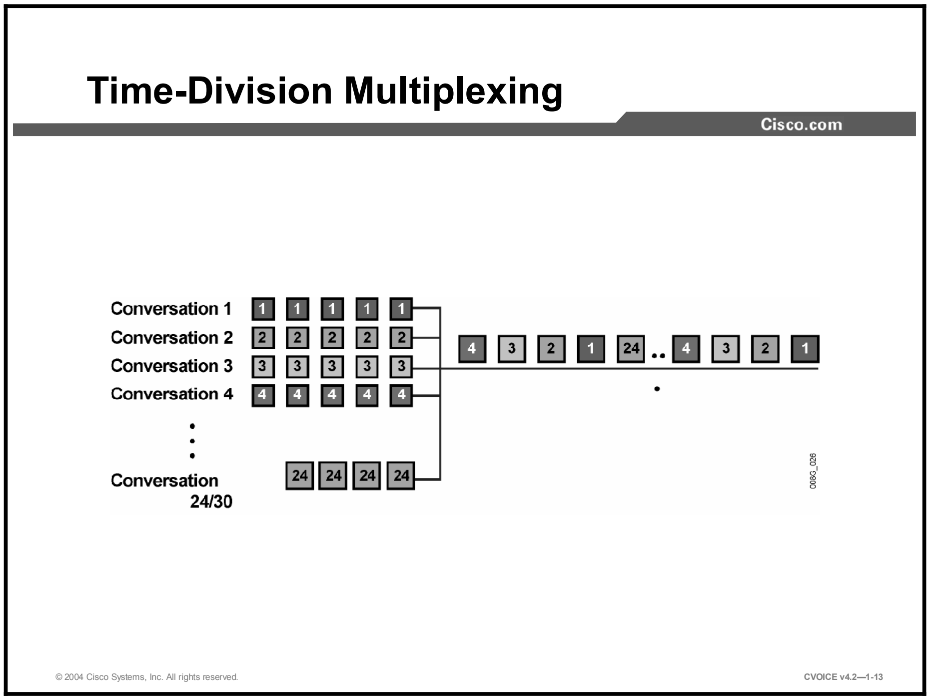
Analog interfaces require configuration of a specific signaling type to match the provider requirement. For interfaces that connect to the PSTN or to a telephone or similar edge device, the signaling is configured for either loop start or ground start. For analog trunk interfaces that connect two PBXs to each other, or a PBX to a CO switch, the signaling is either Wink Start, immediate start, or delay start with the signaling type set to 1, 2, 3, 4, or 5.

Example: Call Signaling at Home

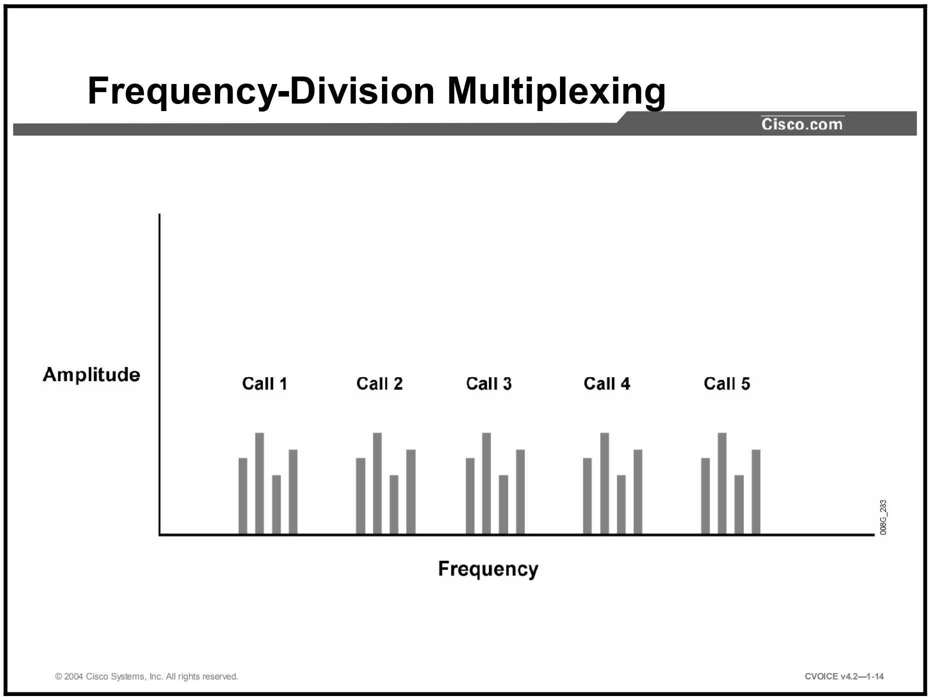
A call placed from your residential telephone uses all three types of call signaling. When you lift the handset, a switch in your telephone closes to start current flow and notifies the telephone company that you want to make a call (supervisory signaling). The telephone company then sends dial tone to indicate that it is ready to receive your dialed digits (informational signaling). You then dial your digits by pressing the number on the keypad (address signaling).

Multiplexing Techniques

A two-wire analog local loop typically carries one call at a time. To make better use of wiring facilities, different multiplexing techniques have been implemented to enable two-wire or four-wire connections to carry multiple conversations at the same time. This topic discusses two of these multiplexing techniques.



Time-division multiplexing (TDM) is used extensively in telephony networks to carry multiple conversations concurrently across a four-wire path. TDM involves simultaneously transmitting multiple separate voice signals over one communications medium by quickly interleaving pieces of each signal, one after another. Information from each data channel is allocated bandwidth based on preassigned timeslots, regardless of whether there is data to transmit.



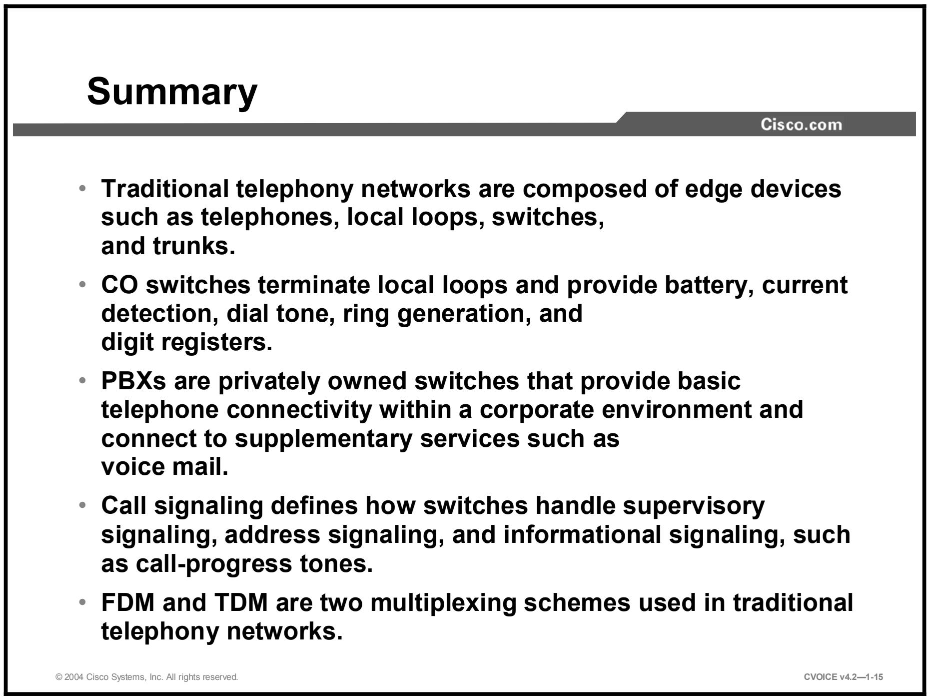
Frequency-division multiplexing (FDM) involves carrying multiple voice signals by allocating an individual frequency range to each call. FDM is typically used in analog connections, although its functionality is similar to that of TDM in digital connections. FDM is used in cable or digital subscriber line (DSL) connections to allow the simultaneous use of multiple channels over the same wire.

Example: Multiplexing Television Channels

If you have cable television service at your home, the television channels are all carried (and multiplexed) over a single pair of wires. This includes both the audio signals and the video signals. Your set-top cable tuner then determines which channel is sent to your television by way of selecting the channel you want to watch. All of the channels are present on the cable wires all of the time, but you tune your selected channel using the set-top tuner.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to this resource:

- *Voice Network Signaling and Control*
  http://www.cisco.com/warp/public/788/signalling/net_signal_control.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Match the component of a telephony network with the function it performs.

A) edge device

B) local loop

C) private or CO switch

D) trunk

_____ 1. handles signaling, call routing, call setup, and call teardown

_____ 2. provides a path between two switches

_____ 3. connects to the PSTN

_____ 4. interfaces to the telephone company network

Q2) Which type of trunk connects two local telephone company COs?

A) tie trunk

B) CO trunk

C) interoffice trunk

D) OPX trunk

Q3) Which two of the following CO switch components are used to make a telephone? (Choose two.)

A) tandem switch

B) current detector

C) terminal interface

D) control complex

E) digit register

Q4) At what point is a dial tone generated?

A) A user dials a number.

B) The flow of current is interrupted.

C) DTMF tones are detected.

D) The digit register is ready.

Q5) What is the difference between a PBX and a key system?

A) Key systems handle more calls.

B) Key systems are digital only.

C) Key systems are not switches.

D) Key systems are analog only.

Q6) Which three of the following are components of a key system? (Choose three.)

A) terminal interface

B) key service unit

C) telephones

D) control complex

E) switching network

F) system software

Q7) Match the type of signaling to its description.

A) supervisory signaling

B) informational signaling

C) address signaling

_____ 1. signaling that uses either pulse or DTMF

_____ 2. signaling that provides call-progress indicators to the initiator of a call

_____ 3. signaling that monitors on-hook/off-hook transitions and provides call notification

Q8) How does CAS pass signaling?

A) in the same channel as voice for T1

B) in the D channel

C) using frequency-division multiplexing

D) using DTMF frequencies

Q9) Which multiplexing process is used by TDM?

A) Timeslots are assigned to different channels, regardless of whether there is data to transmit.

B) Timeslots are assigned to different channels depending on which traffic has a higher priority at that time.

C) All timeslots are used by the channel that is transmitting data at that time.

D) Timeslots are assigned to the channel that starts transmitting first and are used by the other channels as timeslots become available.

Q10) DSL is an example of which style of multiplexing?

A) frequency-division

B) phase-division

C) time-division

D) statistical time-division

Quiz Answer Key

Q1) 1-C, 2-D, 3-A, 4-B

Relates to: Basic Components of a Telephony Network

Q2) C

Relates to: Basic Components of a Telephony Network

Q3) B, E

Relates to: CO Switches

Q4) D

Relates to: CO Switches

Q5) C

Relates to: Private Switching Systems

Q6) B, C, F

Relates to: Private Switching Systems

Q7) 1-C, 2-B, 3-A

Relates to: Call Signaling

Q8) A

Relates to: Call Signaling

Q9) A

Relates to: Multiplexing Techniques

Q10) A

Relates to: Multiplexing Techniques

Understanding Packetized Telephony Networks

Overview

This lesson investigates the driving forces behind converging networks, the components of Packet Telephony Networks, the traffic characteristics of packet telephony, and the control of call volume in the network.

Relevance

The increased efficiency of packet networks and the ability to statistically multiplex voice traffic with data packets allows companies to maximize their return on investment (ROI) in data network infrastructures. Multiplexing voice traffic with data traffic reduces the number of costly circuits dedicated to servicing voice applications.

As demand for voice services expands, it is important to understand the different requirements of voice and data traffic. Previously, voice and data networks were separate and could not impact each other. Today, it is necessary to determine the protocols that are available to control voice calls and ensure that data flows are not negatively impacted.

Objectives

Upon completing this lesson, you will be able to describe two methods of call control used on voice and data networks and provide one protocol example for each. This includes being able to meet these objectives:

- List five benefits of Packet Telephony Networks compared to circuit-switched networks
- Briefly describe three mechanisms of call control that are used in packet telephony
- Compare the gateway functions and signaling processes in centralized and distributed call control models
- List seven basic components of a packet voice network
- Identify a solution for transmitting real-time traffic, such as VoIP, on a best-effort delivery network such as IP

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- General knowledge of telephony technology, including CPE and the PSTN
- General knowledge of networking terminology and IP network concepts

Outline

The outline lists the topics included in this lesson.

Outline

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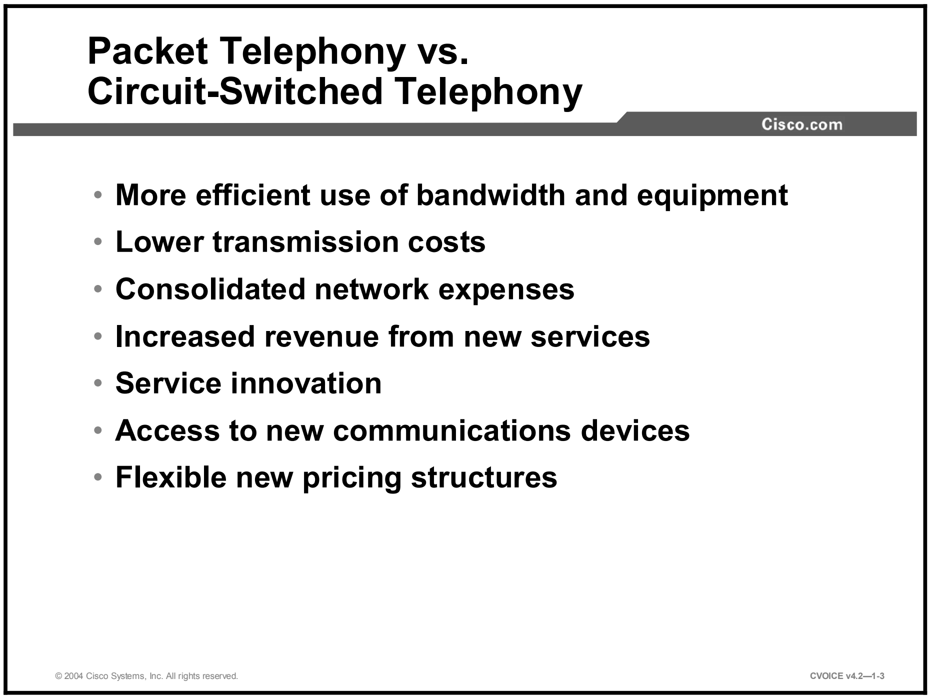
- Overview
- Benefits of Packet Telephony Networks
- Call Control
- Distributed vs. Centralized Call Control
- Packet Telephony Components
- Best-Effort Delivery of Real-Time Traffic
- Summary
- Quiz

 CVOICE v4.2—1-2

Benefits of Packet Telephony Networks

Traditionally, the potential savings on long-distance costs was the driving force behind the migration to converged voice and data networks. The cost of long-distance calls has dropped in recent years, and other factors have come to the forefront as benefits of converged networks. This topic describes some of these benefits.



The benefits of packet telephony versus circuit-switched telephony are as follows:

- **More efficient use of bandwidth and equipment:** Traditional telephony networks use a 64-kbps channel for every voice call. Packet telephony shares bandwidth among multiple logical connections and offloads traffic volume from existing voice switches.
- **Lower costs for telephony network transmission:** A substantial amount of equipment is needed to combine 64-kbps channels into high-speed links for transport across the network. Packet telephony statistically multiplexes voice traffic alongside data traffic. This consolidation represents substantial savings on capital equipment and operations costs.
- **Consolidated voice and data network expenses:** Data networks that function as separate networks to voice networks become major traffic carriers. The underlying voice networks are converted to utilize the packet-switched architecture to create a single integrated communications network with a common switching and transmission system. The benefit is significant cost savings on network equipment and operations.
- **Increased revenues from new services:** Packet telephony enables new integrated services, such as broadcast-quality audio, unified messaging, and real-time voice and data collaboration. These services increase employee productivity and profit margins well above those of basic voice services. In addition, these services enable companies and service providers to differentiate themselves and improve their market position.

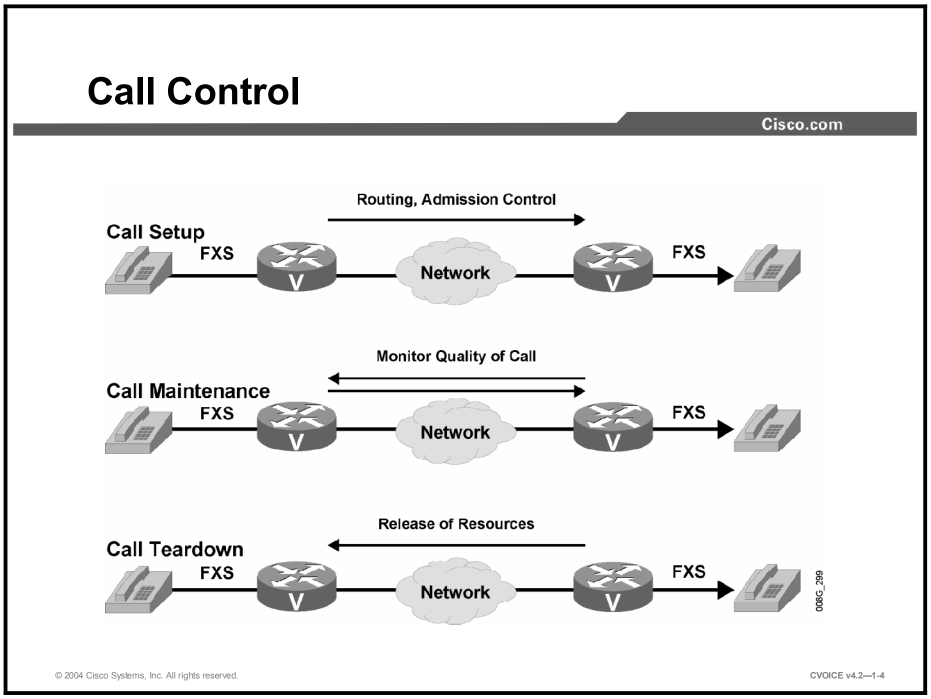
- **Greater innovation in services:** Unified communications use the IP infrastructure to consolidate communication methods that were previously independent; for example, fax, voice mail, e-mail, wireline telephones, cellular telephones, and the web. The IP infrastructure provides users with a common method to access messages and initiate real-time communications—independent of time, location, or device.
- **Access to new communications devices:** Packet technology can reach devices that are largely inaccessible to the TDM infrastructures of today. Examples of such devices are computers, wireless devices, household appliances, personal digital assistants, and cable set-top boxes. Intelligent access to such devices enables companies and service providers to increase the volume of communications they deliver, the breadth of services they offer, and the number of subscribers they serve. Packet technology, therefore, enables companies to market new devices, including videophones, multimedia terminals, and advanced IP Phones.
- **Flexible new pricing structures:** Companies and service providers with packet-switched networks can transform their service and pricing models. Because network bandwidth can be dynamically allocated, network usage no longer needs to be measured in minutes or distance. Dynamic allocation gives service providers the flexibility to meet the needs of their customers in ways that bring them the greatest benefits.

Although packet technology has clear benefits, you should carefully consider the following points before migrating to this technology:

- ROI, when based on the new system features, can be difficult to prove.
- Generally, voice and data staffs do not speak the same language.
- Current voice telephony components have not yet fully depreciated.
- Potential upgrade costs will override potential savings benefits.
- The long-distance carrier may be unwilling to offer price breaks based on digital technology.

Call Control

Call control allows users to establish, maintain, and disconnect a voice flow across a network. This topic describes call control services.

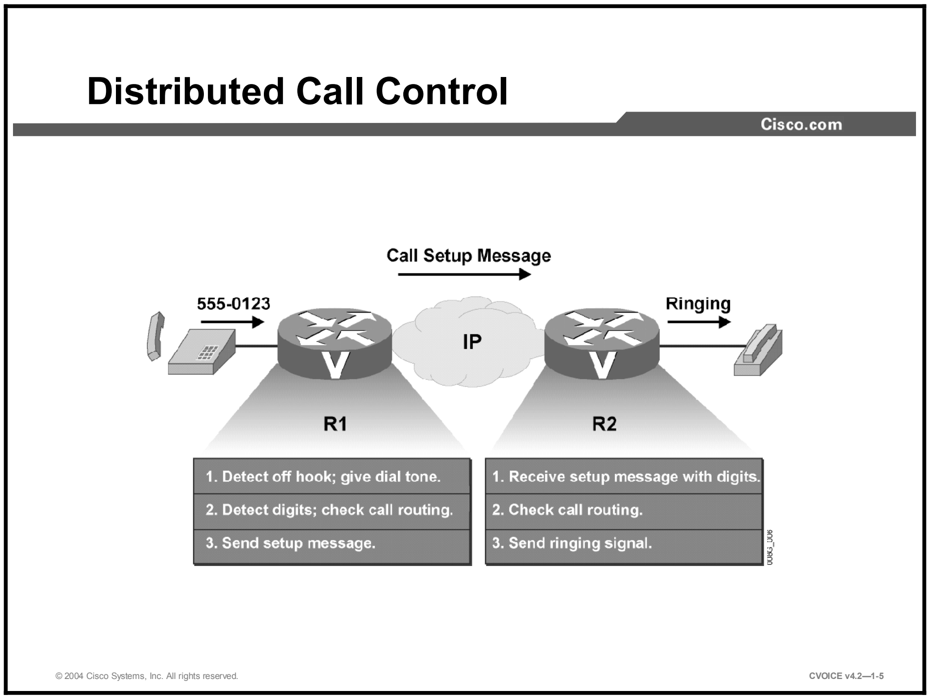


Although different protocols address call control in different ways, they all provide a common set of services. The following are basic components of call control:

- **Call setup:** Checks call-routing configuration to determine the destination of a call. The configuration specifies the bandwidth requirements for the call. When the bandwidth requirements are known, Call Admission Control (CAC) determines if sufficient bandwidth is available to support the call. If bandwidth is available, call setup generates a setup message and sends it to the destination. If bandwidth is not available, call setup notifies the initiator by presenting a busy signal. Different call control protocols, such as H.323, Media Gateway Control Protocol (MGCP), and session initiation protocol (SIP), define different sets of messages to be exchanged during setup.
- **Call maintenance:** Tracks packet count, packet loss, and interarrival jitter or delay when the call is set up. Information passes to the voice-enabled devices to determine if connection quality is good or if it has deteriorated to the point where the call should be dropped.
- **Call teardown:** Notifies voice-enabled devices to free resources and make them available for the next call when either side terminates a call.

Distributed vs. Centralized Call Control

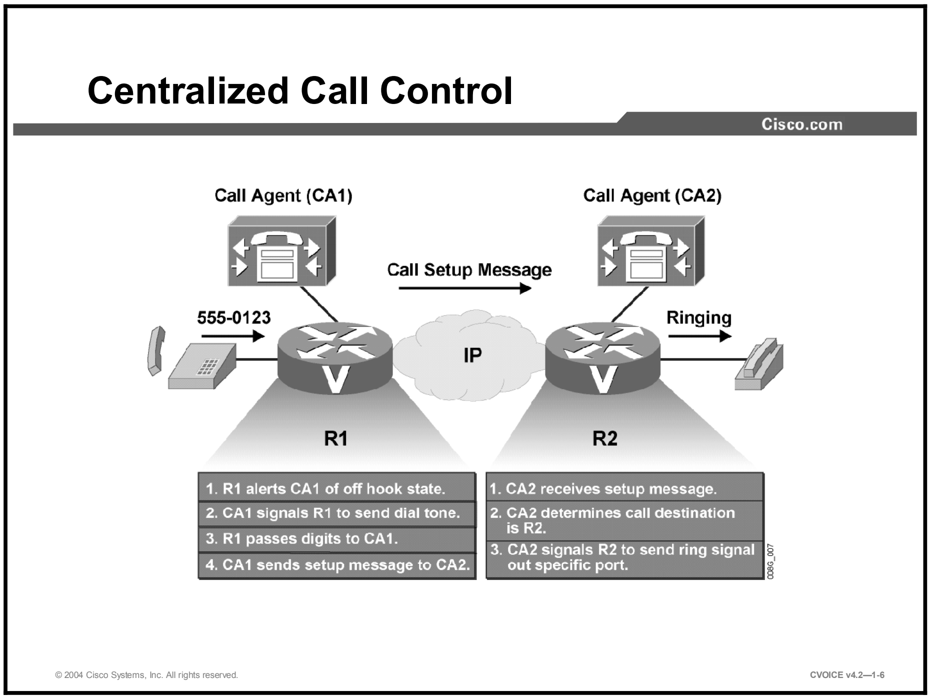
This topic compares distributed and centralized call control.



This figure shows an environment where call control is handled by multiple components in the network. Distributed call control is possible where the voice-capable device is configured to support call control directly. This is the case with a voice gateway when protocols, such as H.323 or SIP, are enabled on the device.

Distributed call control enables the gateway to perform the following procedure:

1. Recognize the request for service
2. Process dialed digits
3. Route the call
4. Supervise the call
5. Terminate the call



Centralized call control allows an external device (call agent) to handle the signaling and call processing, leaving the gateway to translate audio signals into voice packets after call setup. The call agent is responsible for all aspects of signaling, thus instructing the gateways to send specific signals at specific times.

When the call is set up:

- The voice path runs directly between the two gateways and does not involve the call agent.
- When either side terminates the call, the call agent signals the gateways to release resources and wait for another call.

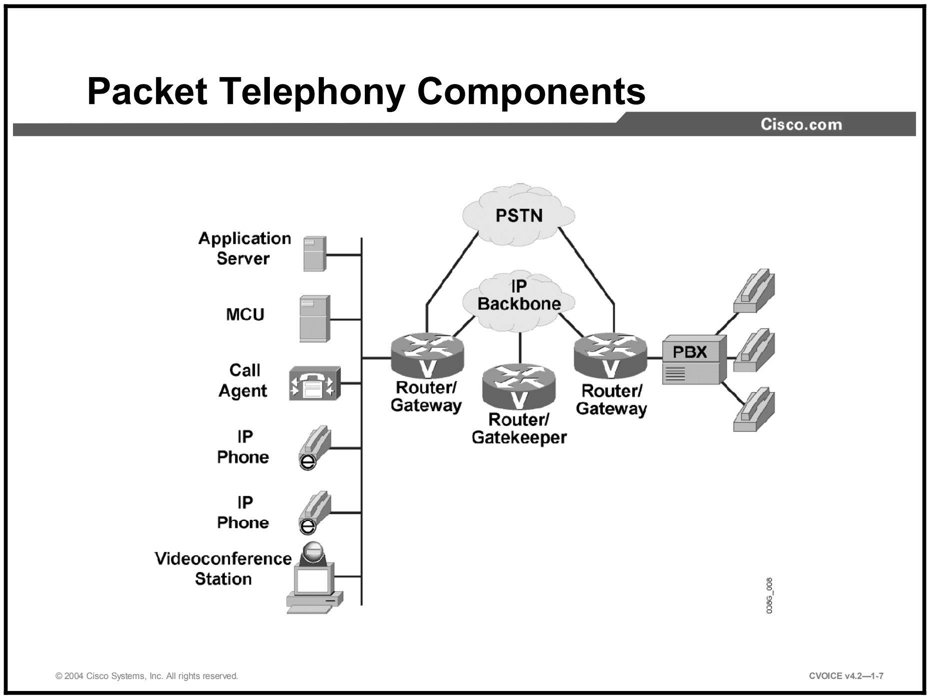
The use of centralized call control devices is beneficial in several ways:

- It centralizes the configuration for call routing and CAC. In a large voice environment, centralization can be extremely beneficial.
- The call agent is the only device that needs the intelligence to understand and participate in call control functions. These call control functions enable the customer to purchase less expensive voice-gateway devices and point to a single device to handle call control.

MGCP is one example of a centralized call control model.

Packet Telephony Components

This topic introduces the basic components of a packet voice network.



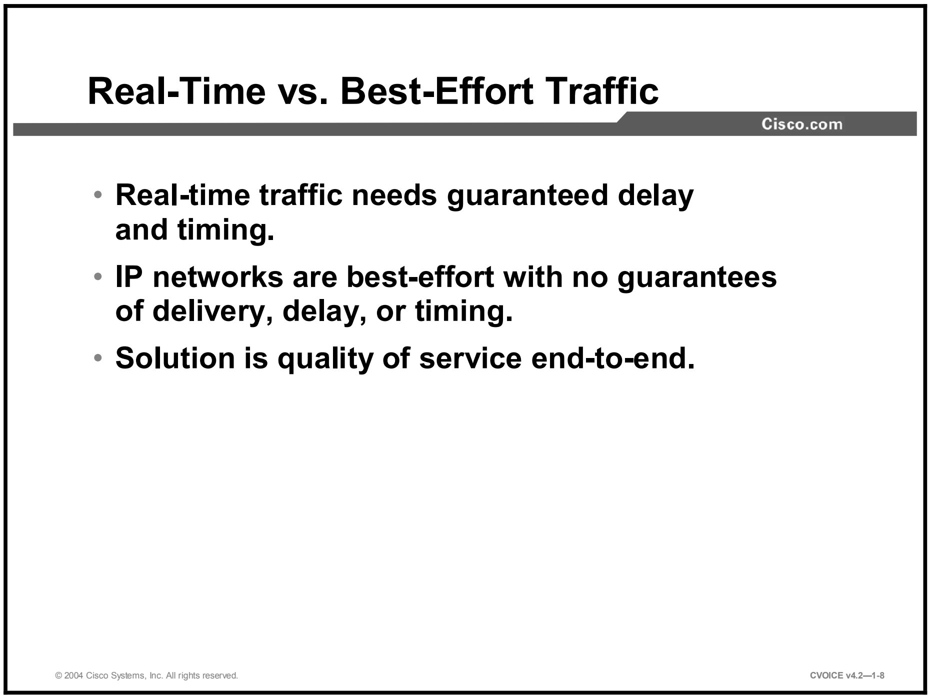
The basic components of a packet voice network include the following:

- **IP Phones:** Provide IP voice to the desktop.
- **Gatekeeper:** Provides CAC, bandwidth control and management, and address translation.
- **Gateway:** Provides translation between VoIP and non-VoIP networks, such as the PSTN. It also provides physical access for local analog and digital voice devices, such as telephones, fax machines, key sets, and PBXs.
- **Multipoint control unit (MCU):** Provides real-time connectivity for participants in multiple locations to attend the same videoconference or meeting.
- **Call agent:** Provides call control for IP Phones, CAC, bandwidth control and management, and address translation.
- **Application servers:** Provide services such as voice mail, unified messaging, and Cisco CallManager Attendant Console.
- **Videoconference station:** Provides access for end-user participation in videoconferencing. The videoconference station contains a video capture device for video input and a microphone for audio input. The user can view video streams and hear the audio that originates at a remote user station.

Other components, such as software voice applications, interactive voice response (IVR) systems, and softphones, provide additional services to meet the needs of enterprise sites.

Best-Effort Delivery of Real-Time Traffic

Voice and data can share the same medium; however, their traffic characteristics differ widely: voice is real-time traffic and data is typically sent as best-effort traffic. This topic compares real-time requirements versus best-effort delivery.



Traditional telephony networks were designed for real-time voice transmission, and therefore cater to the need for a constant voice flow over the connection. Resources are reserved end to end on a per-call basis and are not released until the call is terminated. These resources guarantee that voice flows in an orderly manner. Good voice quality depends on the capacity of the network to deliver voice with guaranteed delay and timing—the requirement for delivery of real-time traffic.

Traditional data networks were designed for best-effort packet transmission. Packet Telephony Networks transmit with no guarantee of delivery, delay, or timing. Data handling is effective in this scenario because upper-layer protocols, such as TCP, provide for reliable—although untimely—packet transmission. TCP trades delay for reliability. Data can typically tolerate a certain amount of delay and is not affected by interpacket jitter.

A well-engineered, end-to-end network is required when converging delay-sensitive traffic, such as VoIP, with best-effort data traffic. Fine-tuning the network to adequately support VoIP involves a series of protocols and features to improve quality of service (QoS). Because the IP network is, by default, best-effort, steps must be taken to ensure proper behavior of both the real-time and best-effort traffic. Packet Telephony Networks succeed, in large part, based on the QoS parameters that are implemented networkwide.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Benefits of a packet voice network include more efficient use of bandwidth, lower costs, innovative services, and increased revenue opportunities.
- Call control is the ability to establish, maintain, and disconnect a voice flow across the network. The basic components of call control are call setup, call maintenance, and call teardown.
- Call control is handled in either a distributed model, where telephony signaling responsibility is spread across many components, or a centralized model, where signaling responsibility rests on one device in the network, called a call agent.

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Summary (Cont.)

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- Basic components of a packet voice network include IP Phones, gateways, gatekeepers, MCUs, call agents, application servers, and videoconference stations.
- Adding real-time traffic such as voice to a best-effort delivery network such as IP requires that QoS measures be implemented throughout the entire network.

 CVOICE v4.2—1-10

References

For additional information, refer to this resource:

- *Extending Enterprise Productivity with Converged IP-Based Cisco Solutions* http://www.cisco.com/warp/public/cc/so/neso/vvda/avvid/avid_pl.htm

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which three of the following are benefits of packet voice networks? (Choose three.)

- A) consolidated voice and data network expenses
- B) guaranteed delivery of digital voice
- C) greater innovation in services
- D) more efficient use of bandwidth and equipment
- E) reduced complexity of converged network design
- F) reduced network protocol support by converting all applications to IP

Q2) What is the main drawback of migrating to packet technology?

- A) Upgrade cost may override potential cost savings.
- B) New devices may not be compatible with the existing PBX.
- C) Unified communications may overload the network.
- D) Voice transmission cost may increase.

Q3) When discussing call control, what function does call setup use to ensure that there is enough bandwidth to place a call?

- A) call routing
- B) call maintenance
- C) call supervision
- D) Call Admission Control

Q4) Which two functions are performed by the call setup component of call control? (Choose two.)

- A) monitors the quality of the connection
- B) determines the destination of the call
- C) tracks packet count and packet loss
- D) notifies voice-enabled devices to make resources available for next call
- E) sends the call initiator a busy signal if sufficient bandwidth is not available

Q5) Which two protocols are examples of distributed call control? (Choose two.)

- A) MGCP
- B) H.323
- C) SIP
- D) Megaco

Q6) In the centralized call control model, signaling is performed by which component?

A) gateway
B) gatekeeper
C) MCU
D) call agent

Q7) Which two features are associated with a gatekeeper? (Choose two.)

A) CAC
B) access to the PSTN
C) billing and accounting
D) address translation
E) call routing

Q8) Match the component of a packet voice network to its function.

A) gateway
B) MCU
C) IP Phone
D) call agent
E) application server
F) videoconference station

_____ 1. translates between the IP network and the PSTN

_____ 2. provides access for end-user participation in videoconferencing

_____ 3. brings IP voice to the desktop

_____ 4. provides real-time connectivity for videoconferencing

_____ 5. provides services such as voice mail and unified messaging

_____ 6. performs call control on behalf of IP Phones

Q9) Which two of the following are required by real-time traffic? (Choose two.)

A) low delay
B) fast converging routing protocol
C) consistent decibel level
D) consistent timing

Q10) Traditional data networks were designed for which characteristic?

A) real-time packet transmission

B) guaranteed delivery

C) guaranteed timing

D) best-effort packet delivery

Quiz Answer Key

Q1) A, C, D

Relates to: Benefits of Packet Telephony Networks

Q2) A

Relates to: Benefits of Packet Telephony Networks

Q3) D

Relates to: Call Control

Q4) B, E

Relates to: Call Control

Q5) B, C

Relates to: Distributed vs. Centralized Call Control

Q6) D

Relates to: Distributed vs. Centralized Call Control

Q7) A, D

Relates to: Packet Telephony Components

Q8) 1-A, 2-F, 3-C, 4-B, 5-E, 6-D

Relates to: Packet Telephony Components

Q9) A, D

Relates to: Best-Effort Delivery of Real-Time Traffic

Q10) D

Relates to: Best-Effort Delivery of Real-Time Traffic

Understanding IP Telephony Applications

Overview

This lesson provides an overview of analog, digital, and Ethernet voice interfaces. It demonstrates campus, enterprise, and service provider voice network topologies.

Relevance

As customers migrate their voice networks, they face a myriad of choices regarding interface types, components, and topologies. A good network design incorporates solutions for current requirements and room for future growth. It is important to understand how voice interfaces with a network, and how the components fit together to provide service in any environment.

Objectives

Upon completing this lesson, you will be able to list five components or capabilities that are required to provide integrated voice and data services in campus LAN, enterprise, and service provider environments. This includes being able to meet these objectives:

- Describe the three basic analog interfaces and the telephony devices that they connect
- Describe the three basic digital interfaces and the type of signaling that each interface supports
- Describe the three physical connectivity options for IP Phones and explain the functioning of the Cisco IP Phone
- List five basic components of an integrated voice and data campus LAN network
- Explain the difference between distributed and centralized enterprise environments and list five main components of each enterprise network type
- List four capabilities that allow service provider networks to offer more efficient, less expensive alternatives to the PSTN

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- General knowledge of telephony technology, including CPE and the PSTN

Outline

The outline lists the topics included in this lesson.

Outline

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- Overview
- Analog Interfaces
- Digital Interfaces
- IP Phones
- Campus LAN Environment
- Enterprise Environment
- Service Provider Environment
- Summary
- Quiz

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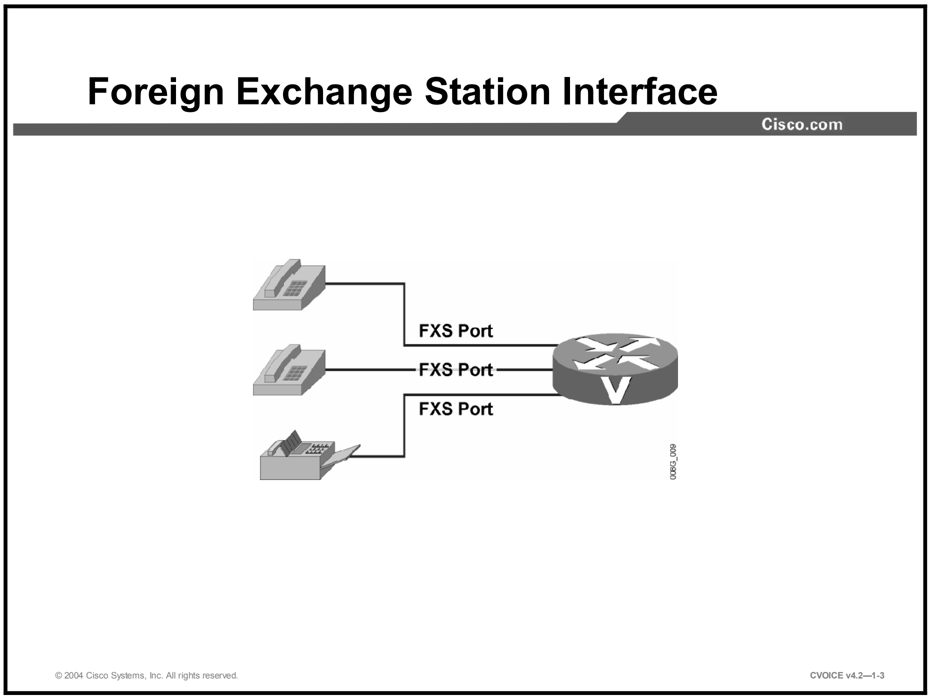
Analog Interfaces

This topic defines three analog interfaces: Foreign Exchange Station (FXS), Foreign Exchange Office (FXO), and ear and mouth (E&M). It also discusses how each of these interfaces is used.



This figure depicts an FXS interface. The FXS interface provides a direct connection to an analog telephone, a fax machine, or a similar device. From a telephone perspective, the FXS interface functions like a switch; therefore, it must supply line power, ring voltage, and dial tone.

The FXS interface contains the coder-decoder (codec), which converts the spoken analog voice wave into a digital format for processing by the voice-enabled device.

Foreign Exchange Office Interface

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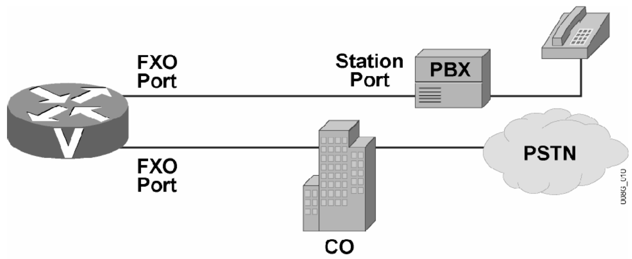

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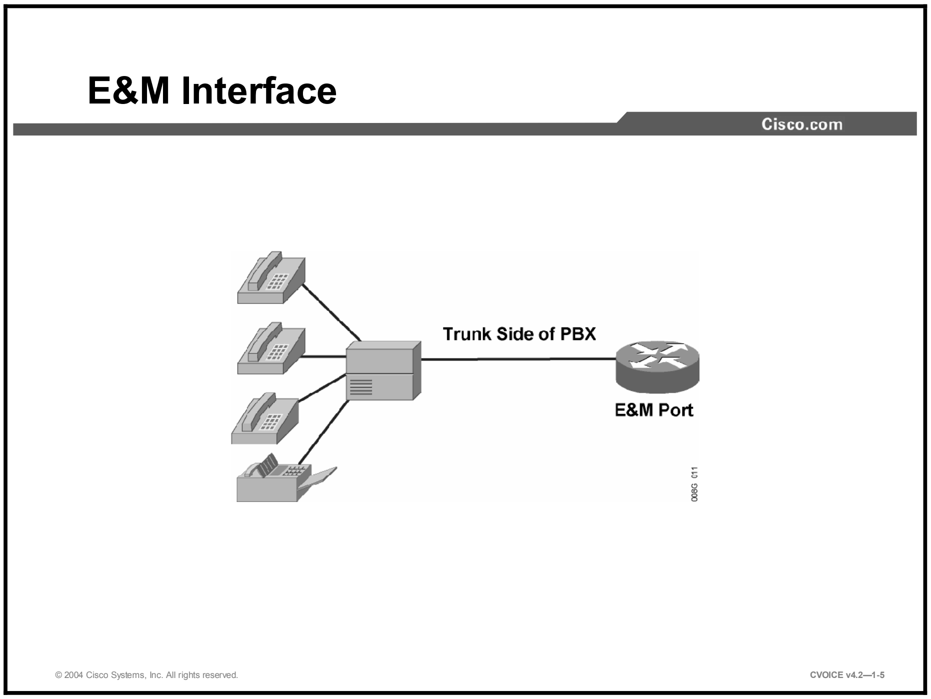
This figure depicts an FXO interface. The FXO interface allows an analog connection to be directed at the CO of a PSTN or to a station interface on a PBX. The switch recognizes the FXO interface as a telephone because the interface plugs directly into the line side of the switch. The FXO interface provides either pulse or DTMF digits for outbound dialing.

In PSTN terminology, an FXO-to-FXS connection is also referred to as a foreign exchange (FX) trunk. An FX trunk is a CO trunk that has access to a distant CO. Because this connection is FXS at one end and FXO at the other end, it acts as a long-distance extension of a local telephone line. In this instance, a local user can pick up the telephone and get a dial tone from a foreign city. Users in the foreign city can dial a local number and have the call connect to the user in the local city.

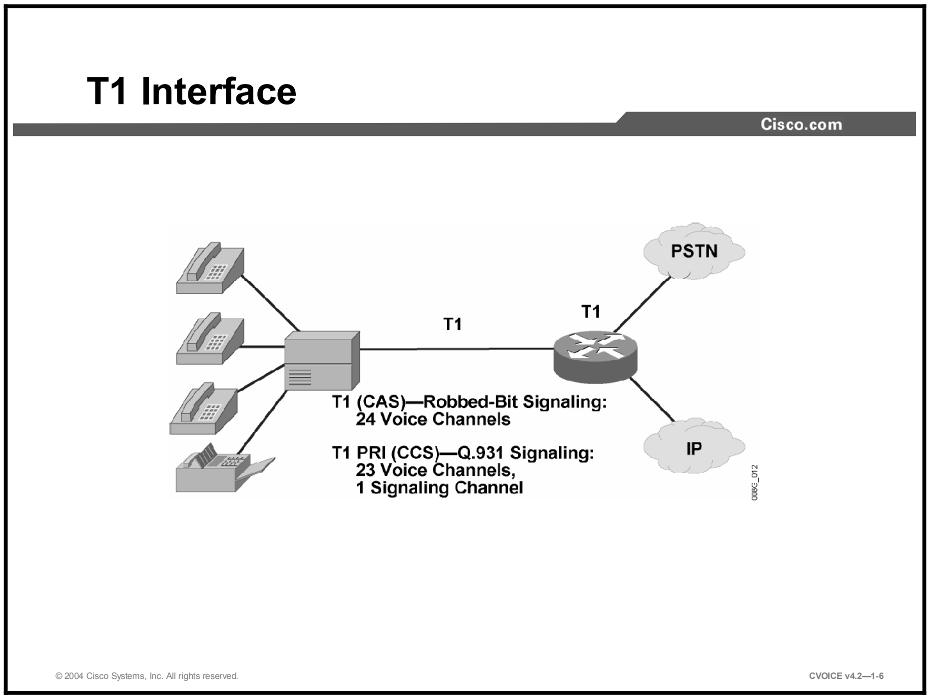


This figure depicts an E&M interface. The E&M interface provides signaling for analog trunking. Analog trunk circuits connect automated systems (PBXs) and networks (COs). E&M signaling is also referred to as "ear and mouth," but its origin comes from the term "Earth and Magneto." Earth represents the electrical ground and magneto represents the electromagnet used to generate tone.

E&M signaling defines a trunk-circuit side and a signaling-unit side for each connection, similar to the DCE and DTE reference types. The PBX is usually the trunk-circuit side and the telco, CO, channel bank, or Cisco voice-enabled platform is the signaling-unit side.

Digital Interfaces

This topic describes the three basic digital voice interfaces: T1, E1, and BRI.



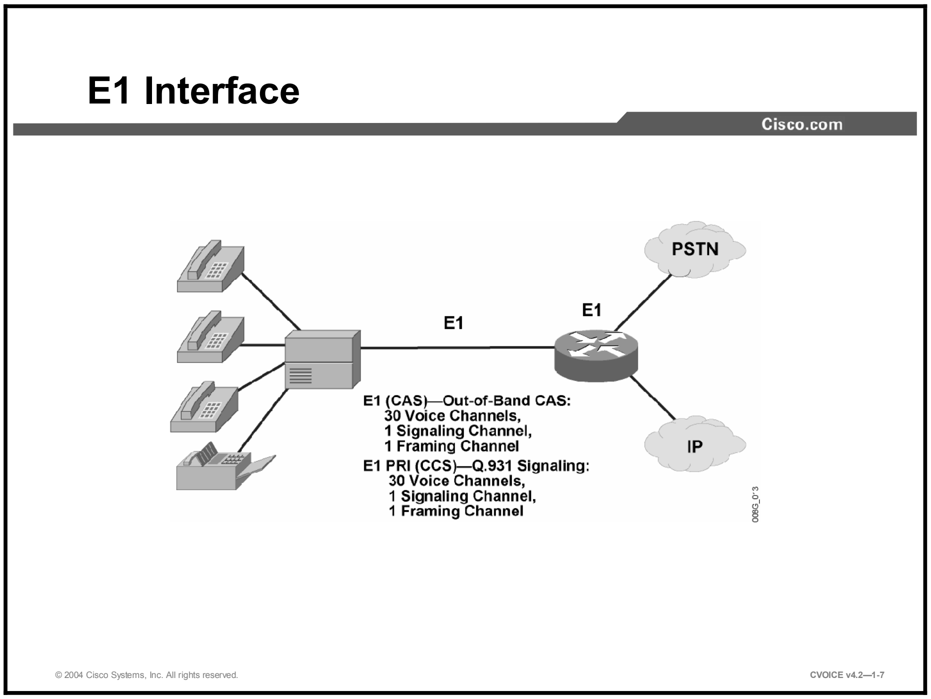
This figure depicts a T1 interface. In a corporate environment with a large volume of voice traffic, connections to the PSTN and to PBXs are primarily digital.

A T1 interface is a form of digital connection that can simultaneously carry up to 24 conversations using two-wire pairs. When a T1 link operates in full-duplex mode, one wire pair sends and the other wire pair receives. The 24 channels are grouped together to form a frame. The frames are then grouped together into Super Frames (groups of 12 frames) or into Extended Superframes (groups of 24 frames).

The T1 interface carries either CAS or CCS. When a T1 interface uses CAS, the signaling robs a sampling bit for each channel to convey in band. When a T1 interface uses CCS, Q.931 signaling is used on a single channel, typically the last channel.

To configure CAS you must:

- Specify the type of signaling that the robbed bits carry; for example, E&M Wink Start. This signaling must match the PSTN requirements or the PBX configuration. This is considered in-band signaling because the signal shares the same channel as the voice.
- Configure the interface for PRI signaling. This level of configuration makes it possible to use channels 1 to 23 (called B channels) for voice traffic. Channel 24 (called the D channel) carries the Q.931 call control signaling for call setup, maintenance, and teardown. This type of signaling is considered out-of-band signaling because the Q.931 messages are sent in the D channel only.



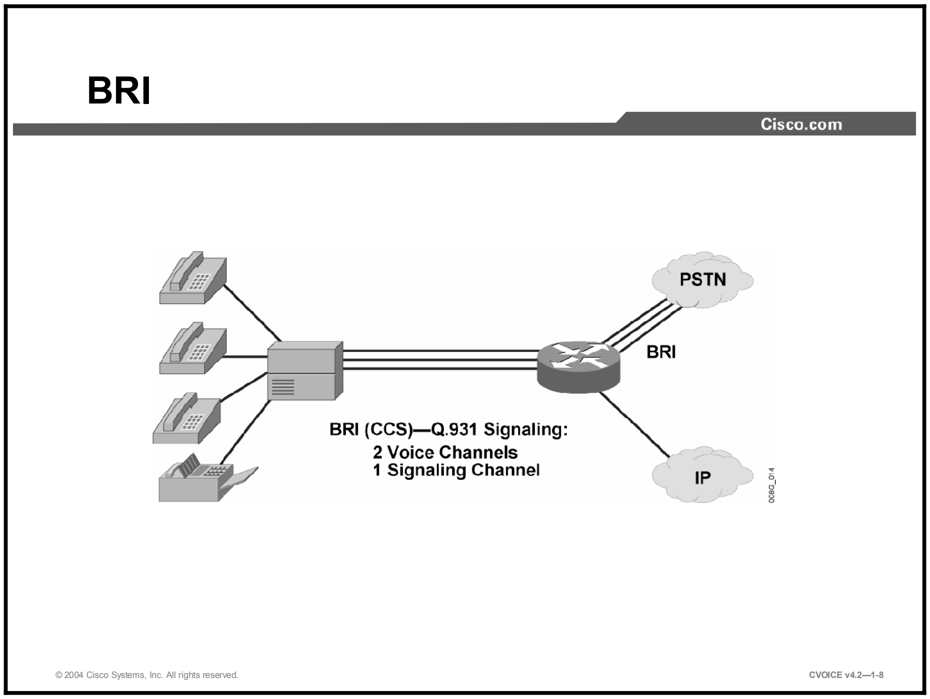
This figure depicts an E1 interface. An E1 interface has 32 channels and simultaneously carries up to 30 conversations. The other two channels are used for framing and signaling. The 32 channels are grouped to form a frame. The frames are then grouped together into multiframes (groups of 16 frames). Europe and Mexico use the E1 interface.

Although you can configure the E1 interface for either CAS or CCS, the most common usage is CCS.

When an E1 interface uses CAS, signaling travels out of band in the signaling channel but follows a strict association between the signal carried in the signaling channel and the channel to which the signaling is being applied. The signaling channel is channel 16.

In the first frame, channel 16 carries 4 bits of signaling for channel 1 and 4 bits of signaling for channel 17. In the second frame, channel 16 carries 4 bits of signaling for channel 2 and 4 bits for channel 18, and so on. This process makes it out-of-band CAS.

When an E1 interface uses CCS, Q.931 signaling is used on a single channel, typically channel 17. When configuring for CCS, configure the interface for PRI signaling. When E1 is configured for CCS, channel 16 carries Q.931 signaling messages only.

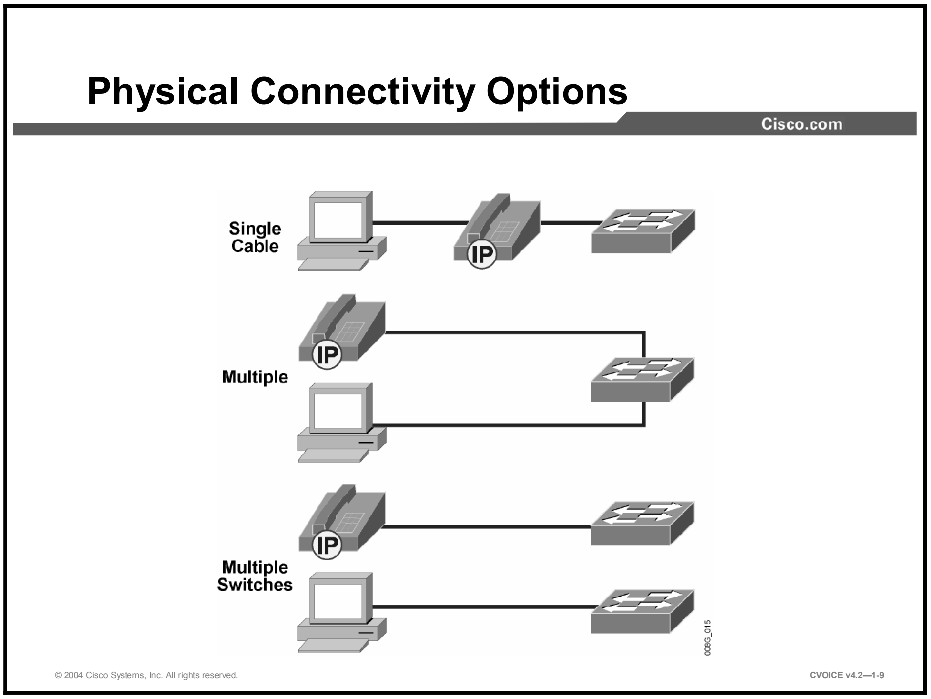


This figure depicts a Basic Rate Interface (BRI). You can use a BRI to connect the PBX voice into the network. Used primarily in Europe for PBX connectivity, BRI provides a 16-kbps D channel for signaling and two 64-kbps B channels for voice. BRI uses Q.931 signaling in the D channel for call signaling.

Note	Cisco Systems does not officially support ISDN telephones.

IP Phones

This topic describes scenarios for desktop telephone connections and the IP Phone built-in switch ports.



This figure depicts physical connection options for IP Phones. The IP Phone connects to the network through a Category 5 or better cable that has RJ-45 connectors. The power-enabled switch port or an external power supply provides power to an IP Phone. The IP Phone functions like other IP-capable devices sending IP packets to the IP network. Because these packets are carrying voice, you must consider both logical and physical configuration issues.

At the physical connection level, there are three options for connecting the IP Phone:

- **Single cable:** A single cable connects the telephone and the PC to the switch. Most enterprises install IP Phones on their networks using a single cable for both the telephone and a PC. Reasons for using a single cable include ease of installation and cost savings on cabling infrastructure and wiring-closet switch ports.
- **Multiple cables:** Separate cables connect the telephone and the PC to the switch. Users often connect the IP Phone and PC using separate cables. This connection creates a physical separation between the voice and data networks.
- **Multiple switches:** Separate cables connect the telephone and the PC to separate switches. With this option, IP Phones are connected to separate switches in the wiring closet. By using this approach, you can avoid the cost of upgrading the current data switches and keep the voice and data networks completely separate.

Multiple switches are used to do the following:

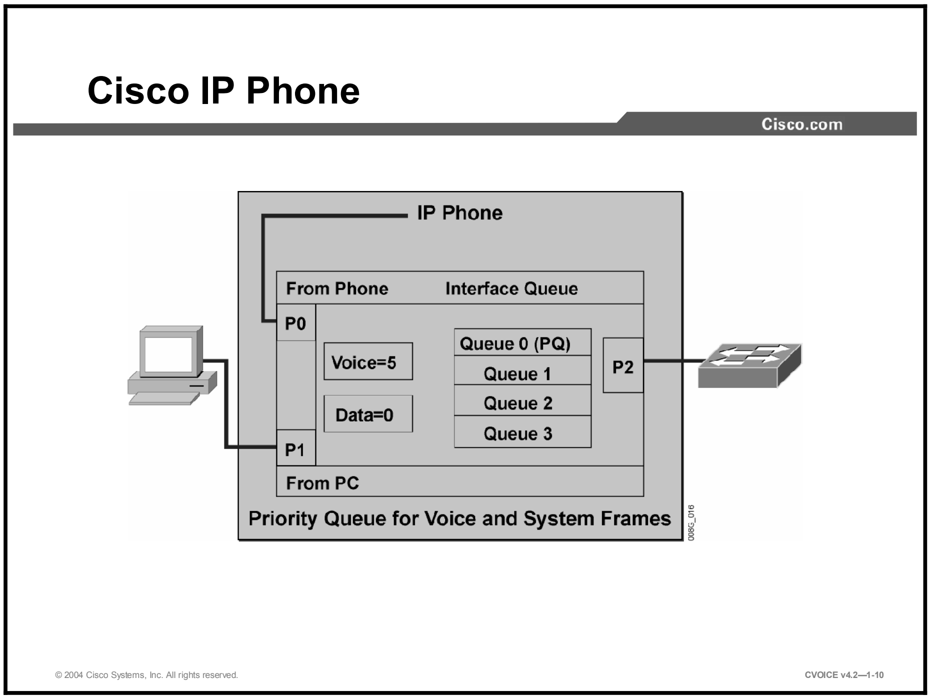
- Provide inline power to IP Phones without having to upgrade the data infrastructure
- Limit the number of switches that need an uninterruptible power supply (UPS)
- Reduce the amount of Cisco IOS Catalyst software upgrades needed in the network
- Limit the spanning-tree configuration in the wiring-closet switches

The physical configuration for connecting an IP Phone must address the following issues:

- Speed and duplex settings
- Inline power settings

The logical configuration for connecting an IP Phone must address the following issues:

- IP addressing
- VLAN assignment
- Spanning tree
- Classification and queuing



The basic function of a Cisco IP Phone depends on a three-port 10/100 switch. Port P0 is an internal port that connects the voice electronics in the telephone. Port P1 connects a daisy-chained PC. Port P2 uplinks to the Ethernet switch in the wiring closet.

Each port contains four queues with a single threshold. One of these queues is a high-priority queue used for system frames. By default, voice frames are classified for processing in the high-priority queue, and data frames are classified for processing in the low-priority queue.

The internal Ethernet switch on the Cisco IP Phone switches incoming traffic to either the access port or the network port.

If a computer is connected to the port P1, data packets traveling to and from the computer, and to and from the phone, share the same physical link to the access layer switch connected to port P2, and to the same port on the access layer switch. This shared physical link has the following implications for the VLAN network configuration:

- Current VLANs may be configured on an IP subnet basis. However, additional IP addresses may not be available for assigning the telephone to the same subnet as the other devices that are connected to the same port.
- Data traffic that is supporting phones on the VLAN may reduce the quality of VoIP traffic.

You can resolve these issues by isolating the voice traffic on a separate VLAN for each of the ports connected to a telephone. The switch port configured for connecting a telephone would have separate VLANs configured to carry the following types of traffic:

- Voice traffic to and from the IP Phone (auxiliary VLAN)
- Data traffic to and from the PC connected to the switch through the IP Phone access port (native VLAN)

Isolating the telephones on a separate auxiliary VLAN increases voice-traffic quality and allows a large number of telephones to be added to an existing network that has a shortage of IP addresses.

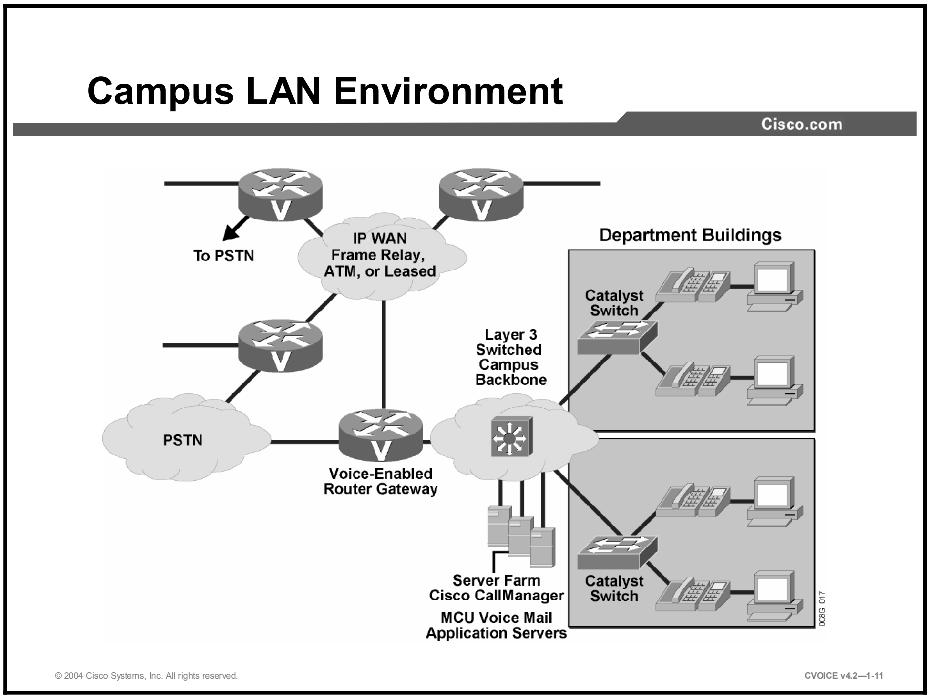
Note	For more information, refer to the documentation included with the Cisco Catalyst switch.

Example: IP Phone Installations

Cisco IP Phones deployed in an office environment attach to Ethernet switches. The IP Phone uses the existing cable infrastructure, or the infrastructure is updated to allow one connection for the phone and one for the desktop PC. The connections from the phone and the PC may lead to the same switch or to different switches. In either case, the IP Phone has the capability to prioritize voice frames.

Campus LAN Environment

This topic describes the components used in campus IP telephony installations.



Campus LAN environments have grown tremendously in the past several years due to the demand for networked resources, instant business communication, and Voice over Data applications.

Components for integrated voice and data campus networks include the following:

- **IP Phone:** Provides IP voice to the desktop.
- **Cisco CallManager cluster:** Acts as the centralized call-processor for all VoIP devices in the system. The Cisco CallManager cluster is also used for configuring telephones, routing plans for the gateways, and other features. Clustering provides scalability and redundancy.
- **Gateway:** Connects the IP voice to external voice networks, such as the PSTN, or voice devices, such as IVR systems or fax machines. The gateway performs important functions, such as transcoding digital and analog voice, and may provide conferencing resources.
- **MCU:** Provides conferencing capabilities.
- **Application server:** Provides additional services, such as voice mail and unified messaging.

When you are designing the campus infrastructure for voice, you must consider the following key issues:

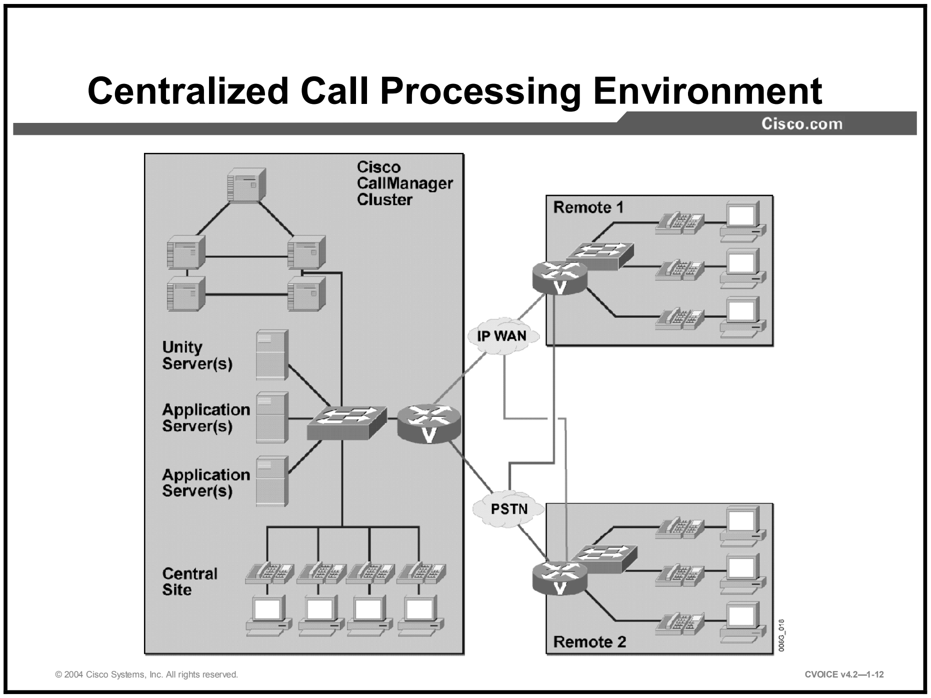
- Robust, fault-tolerant, highly available network design
- Ability to power IP Phones
- Redundant power supply for network components
- Ease of IP addressing
- QoS enhancements

Example: Campus Environments

Cisco Systems' internal telephone network in San Jose can be considered a campus LAN environment. All desktop phones connect to Ethernet switches and are controlled by CallManager applications. CallManager also controls the gateways and other application servers, such as the Unity server.

Enterprise Environment

Enterprise networks grow and evolve as company services and locations change and expand. Heavy reliance on information processing and universal access to corporate information has driven network designs to provide reliable access, redundancy, reachability, and manageability. These same principles apply to designing corporation-wide voice access in the enterprise environment. This topic describes both the centralized and distributed call processing solutions for enterprise environments.

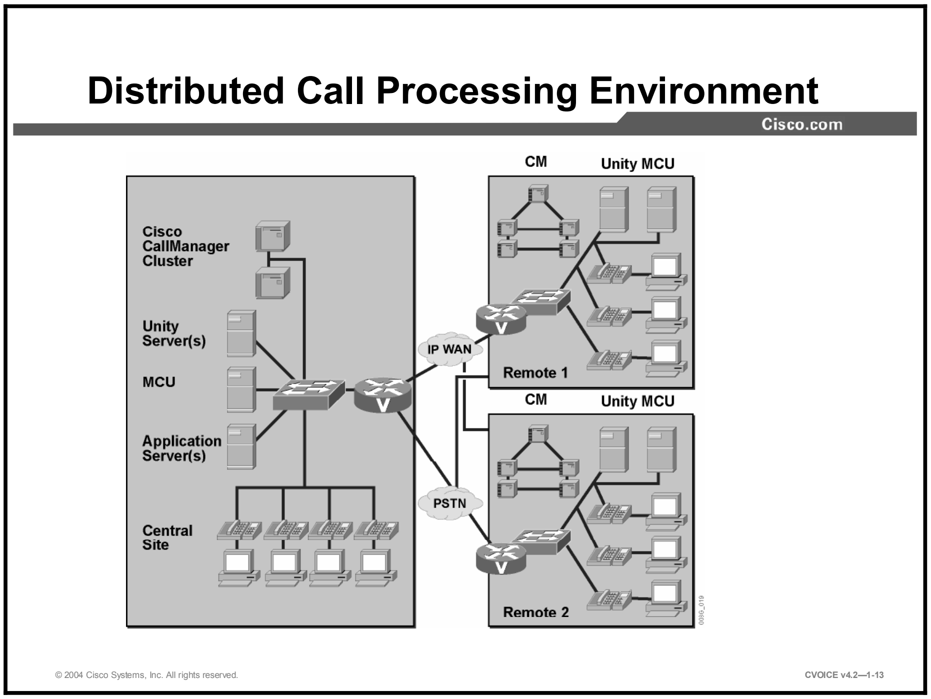


This figure depicts an enterprise centralized call processing environment. Centralized voice networks provide enterprise-wide voice access for calls and voice services controlled from a central site. In this environment, the central site provisions all voice services, such as Cisco CallManager, voice mail, and unified messaging. IP Phones at remote sites connect to Cisco CallManager through the IP WAN for call processing.

Components for centralized voice enterprise networks include the following:

- **IP Phone:** Provides IP voice to the desktop.
- **Cisco CallManager cluster (central site only):** Acts as the central management console for all VoIP devices in the system and is used for configuring telephones, routing plans for the gateways, and all other features. Clustering provides scalability and redundancy.
- **Gateway (all sites):** Connects the IP voice to external voice networks, such as the PSTN, or to voice devices, such as IVR systems or fax machines. The gateway performs important functions, such as transcoding digital and analog voice, and may provide conferencing resources.
- **MCU (central site only):** Provides conferencing capabilities.
- **Application server (central site only):** Provides additional services, such as voice mail and unified messaging.

- **Unity server:** Provides a central repository for unified messaging, such as voice mail, e-mail, and fax.
- **Survivable Remote Site Telephony (SRST) based on Cisco IOS software (remote sites only):** Provides local call-processing capabilities in the event of a WAN outage.
- **IP WAN:** Serves as the primary voice path between sites, with the PSTN as the secondary voice path.



This figure depicts an enterprise distributed call processing environment. Distributed voice networks place voice components at each site and utilize the WAN for intersite calls only.

Components for distributed voice enterprise networks include the following:

- **IP Phone:** Provides IP voice to the desktop.
- **Cisco CallManager cluster:** Acts as the central management console for all VoIP devices at each site and is used for configuring telephones, routing plans for the gateways, and all other features. Clustering provides scalability and redundancy.
- **Gateway:** Connects the IP voice to external voice networks, such as the PSTN, or voice devices, such as IVR systems or fax machines. The gateway performs important functions, such as transcoding digital and analog voice, and may provide conferencing resources.
- **MCU:** Provides real-time videoconferencing capabilities.
- **Application server:** Provides additional services, such as voice mail and unified messaging.
- **IP WAN:** Serves as the primary voice path between sites, with the PSTN as the secondary voice path.

Modern enterprise network applications include:

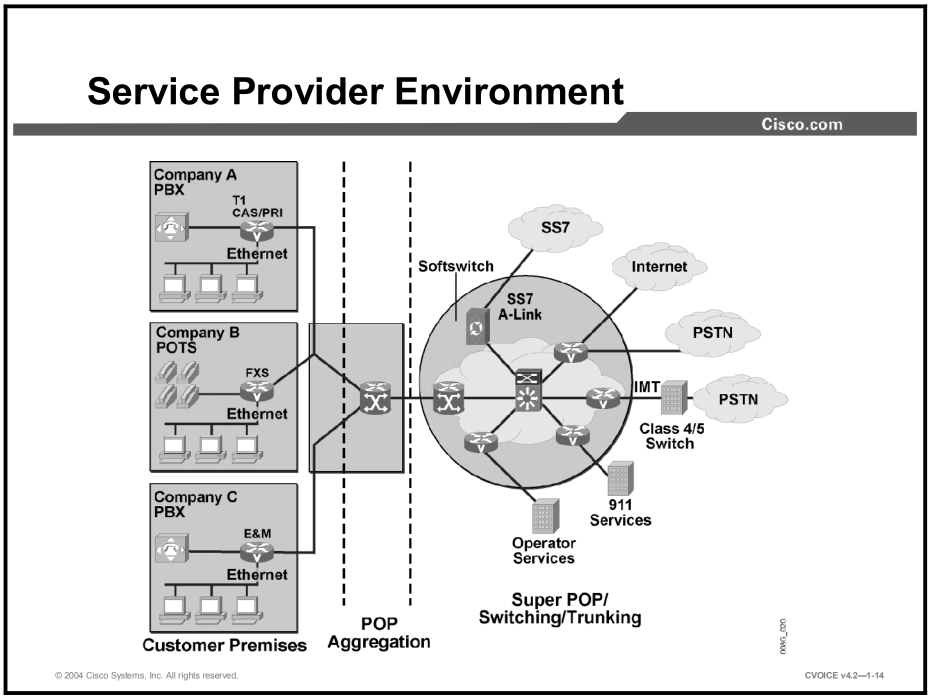
- E-business
- E-learning
- Customer care
- Unified messaging
- Videoconferencing
- Voice calls placed from web pages

Example: Enterprise Environments

The enterprises can be considered to be either centralized or distributed call processing environments. In the centralized call processing environment, all of the components of the voice system are controlled by a single centralized call agent, such as CallManager, regardless of their physical location. In a distributed call processing environment, the components of the voice network at each location can act independently.

Service Provider Environment

Service provider requirements add another level of complexity to the voice environment. This topic describes basic service provider requirements.



This figure depicts a service provider environment. To be competitive, service providers must provide their business customers with more efficient, less expensive alternatives to the PSTN for voice and data services.

Requirements in the service provider arena include:

- **Carrier class performance:** Voice gateways must provide service that minimizes latency and controls jitter. This level of performance allows customers to maintain voice quality as they migrate from circuit-switched voice to IP-based services.
- **Scalability:** Design must accommodate rapid growth to enable service providers to grow with their customer base. An important aspect of scalability is the automation, configuration, and administration of IP networks and gateways for seamless expansion.
- **Comprehensive call records supporting flexible service pricing:** This is the ability to extract IP session and transaction information from multiple network devices and from all layers of the network—in real time—to produce detailed billing records.
- **Signaling System 7 (SS7) interconnect capabilities:** Tariffs favor interconnection using SS7 signaling, because Inter-Machine Trunks (IMTs) are less expensive than ISDN-based facilities. This financial benefit equates to lower monthly expenses, the reduced cost of goods that are sold, and higher margins for service providers.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- FXS interface connects analog edge devices, such as telephone or fax.
- FXO interface connects to the PSTN or station side of a switch.
- E&M interface is an analog trunk that connects PBXs and switches.
- T1 is a digital interface that provides up to 24 channels for voice.
- E1 is a digital interface that provides 30 channels for voice.
- BRI is a digital interface that provides two channels for voice.

 CVOICE v4.2—1-15

Summary (Cont.)

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- IP Phones use a Category 5 or better cable with RJ-45 connectors to attach to the switch.
- Campus voice components include Cisco CallManager, IP Phones, gateways, and application servers, such as voice mail and unified messaging.
- Enterprise voice network design allows for a distributed or centralized call-processing model.
- Service provider environments require carrier-level performance, scalability, comprehensive call records, and SS7 connection capabilities.

 CVOICE v4.2—1-16

References

For additional information, refer to these resources:

- *Design Guides* http://www.cisco.com/en/US/netsol/ns340/ns394/ns147/ns17/networking_solutions_design__guidances_list.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) In E&M signaling, to what does the "E" refer?

A) electromagnet

B) ring

C) electrical ground

D) tip

E) erlang

Q2) To what does the FXS interface provide a direct connection?

A) CO

B) E&M trunk

C) telephone

D) gatekeeper

Q3) An E1 interface can carry up to how many simultaneous conversations?

A) 16

B) 24

C) 30

D) 32

Q4) What type of signaling is used when a T1 interface uses CCS?

A) robbed-bit

B) Megaco

C) Q.931

D) H.323

Q5) What are three types of physical connectivity options for an IP Phone? (Choose three.)

A) single cable IP Phone to hub

B) single cable PC to IP Phone to switch

C) dual cable IP Phone to switch and PC to switch

D) dual cable PC to IP phone, IP Phone to switch

E) multiswitch, PC to switch A for data, IP Phone to switch B for voice

Q6) How many queues does each port on a Cisco IP Phone contain?

A) 2

B) 3

C) 4

D) 12

Q7) In a campus LAN environment, what does the multipoint control unit provide?

A) central call-processing capabilities

B) configuration capabilities

C) conferencing capabilities

D) extended-calling capabilities

E) call-pickup capabilities

Q8) Which two of the following key issues must be considered when designing the campus infrastructure for voice? (Choose two.)

A) ability to power IP Phones

B) processing power of the gatekeeper

C) size of DRAM in the routers

D) ease of IP addressing

Q9) Which two models are used for designing enterprise VoIP networks? (Choose two.)

A) service provider

B) carrier class

C) distributed

D) centralized

E) remote site

Q10) What is the function of the SRTS component of a centralized voice enterprise network?

A) connects IP Phones on remote sites to Cisco CallManager

B) provides local call-processing capabilities in case of a WAN outage

C) configures routing plans for the gateways

D) serves as the primary voice path between sites

Q11) To be competitive, service providers must provide their business customers with what kind of alternatives to the PSTN for voice and data services?

A) more extensive, less complicated

B) more exhaustive, less limited

C) more digital, less analog

D) more efficient, less expensive

Q12) In service provider environments, what must the network design accommodate?

A) support for TDM

B) the automation of management and configuration of IP networks

C) support for SIP, H.323, and MGCP

D) a full Class A IP-address block

Quiz Answer Key

Q1) C

Relates to: Analog Interfaces

Q2) C

Relates to: Analog Interfaces

Q3) C

Relates to: Digital Interfaces

Q4) C

Relates to: Digital Interfaces

Q5) B, C, and E

Relates to: IP Phones

Q6) C

Relates to: IP Phones

Q7) C

Relates to: Campus LAN Environment

Q8) A, D

Relates to: Campus LAN Environment

Q9) C, D

Relates to: Enterprise Environment

Q10) B

Relates to: Enterprise Environment

Q11) D

Relates to: Service Provider Environment

Q12) B

Relates to: Service Provider Environment

Module 2

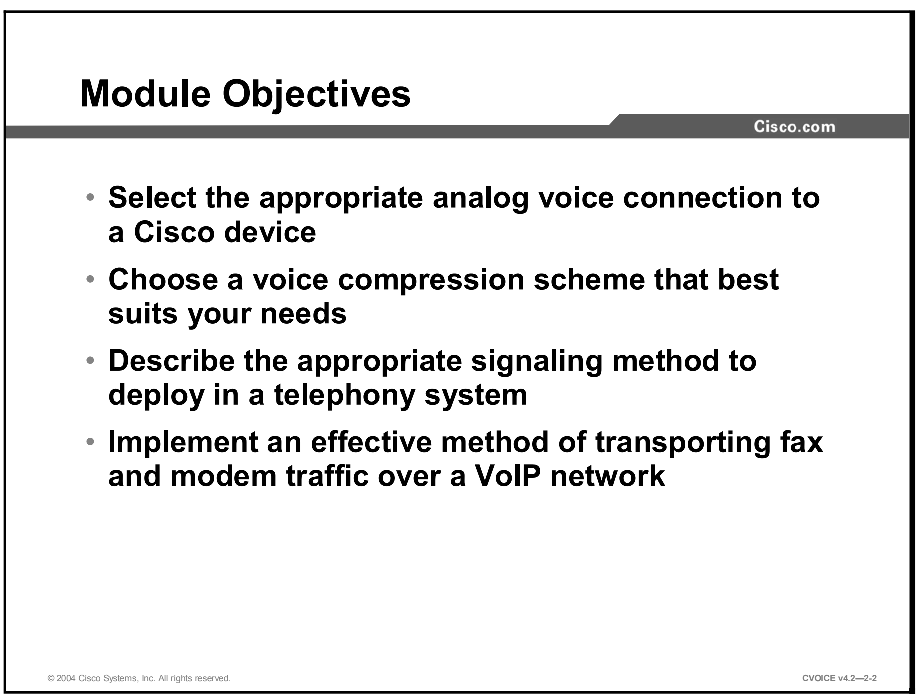
Analog and Digital Voice Connections

Overview

Cisco voice devices must support a wide variety of connection types. This module describes the various analog and digital connections, introduces quality issues, describes common compression schemes, and concludes with a description of fax-over-IP voice networks.

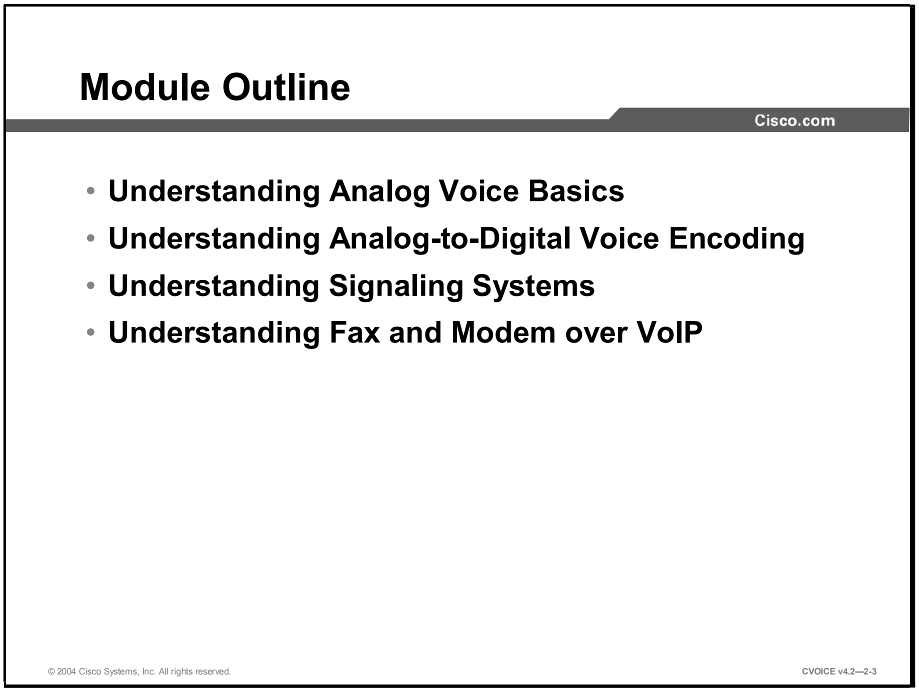
Module Objectives

Upon completing this module, you will be able to explain the processes and standards for voice digitization, compression, digital signaling, and fax transport as they relate to Voice over IP (VoIP) networks.



Module Outline

The outline lists the components of this module.



Understanding Analog Voice Basics

Overview

Interfacing Cisco Systems equipment with traditional analog telephony devices requires an understanding of the various interfaces used in the industry. This lesson explains the types of analog connections that are used and their susceptibility to line quality problems.

Relevance

To correctly choose and implement an analog voice connection to a Cisco device, you must understand the types of interfaces that are used in the industry and their line quality effects.

Objectives

Upon completing this lesson, you will be able to select the appropriate analog voice connection to a Cisco device. This includes being able to meet these objectives:

- Identify the components of local-loop connections
- Identify the signaling used on local loops
- Identify on-hook, off-hook, and ringing supervisory signaling
- Identify pulse dialing and dual tone multifrequency
- List the call-progress indicators and their functions
- State the purpose and types of trunk connections
- Identify the signaling used on trunks
- Identify the five types of wiring schemes used in E&M signaling
- Describe Wink-Start, immediate-start, and delay-start signaling
- Describe impairments that interfere with line quality
- Identify two methods that are used in the industry to reduce the problem of echo

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with telephony station equipment, such as handsets and fax machines
- Familiarity with telephony switching equipment, such as key systems and PBXs

Outline

The outline lists the topics included in this lesson.

Outline

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- **Overview**
- **Local-Loop Connections**
- **Types of Local-Loop Signaling**
- **Supervisory Signaling**
- **Address Signaling**
- **Informational Signaling**
- **Trunk Connections**
- **Types of Trunk Signaling**
- **E&M Signaling Types**
- **Trunk Signal Types Used by E&M**
- **Line Quality**
- **Management of Echo**
- **Summary**
- **Quiz**
- **Lab**

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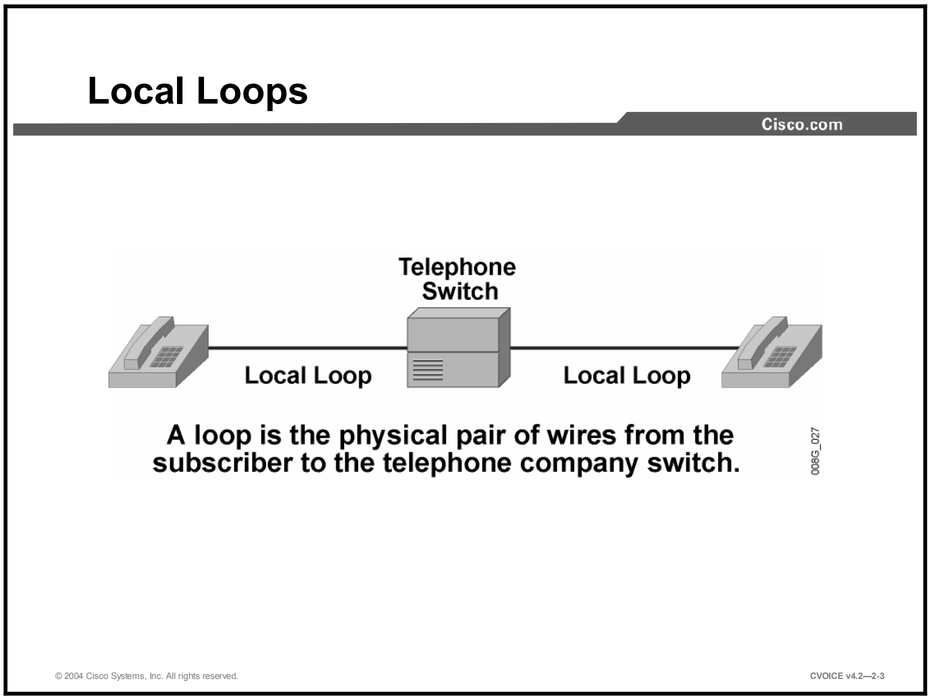
Local-Loop Connections

This topic describes the parts of a traditional telephony local-loop connection between a telephone subscriber and the telephone company.



A subscriber home telephone connects to the telephone company central office (CO) via an electrical communication path called a local loop, as depicted in the figure. The loop consists of a pair of twisted wires—one is called *tip,* the other is called *ring*.

In most arrangements, the ring wire ties to the negative side of a power source, called the *battery,* while the tip wire connects to the ground. When you take your telephone off hook, current flows around the loop, allowing dial tone to reach your handset. Your local loop, along with all others in your neighborhood, connects to the CO in a cable bundle, either buried underground or strung on poles.

Example: Residential Telephone Service

Your home telephone service is provided to you from your service provider by way of two wires. Your home telephone controls whether or not the service on these wires is activated via the switch hook inside the telephone.

Types of Local-Loop Signaling

This topic explains local-loop signaling and lists some of the signaling types.

Types of Local-Loop Signaling

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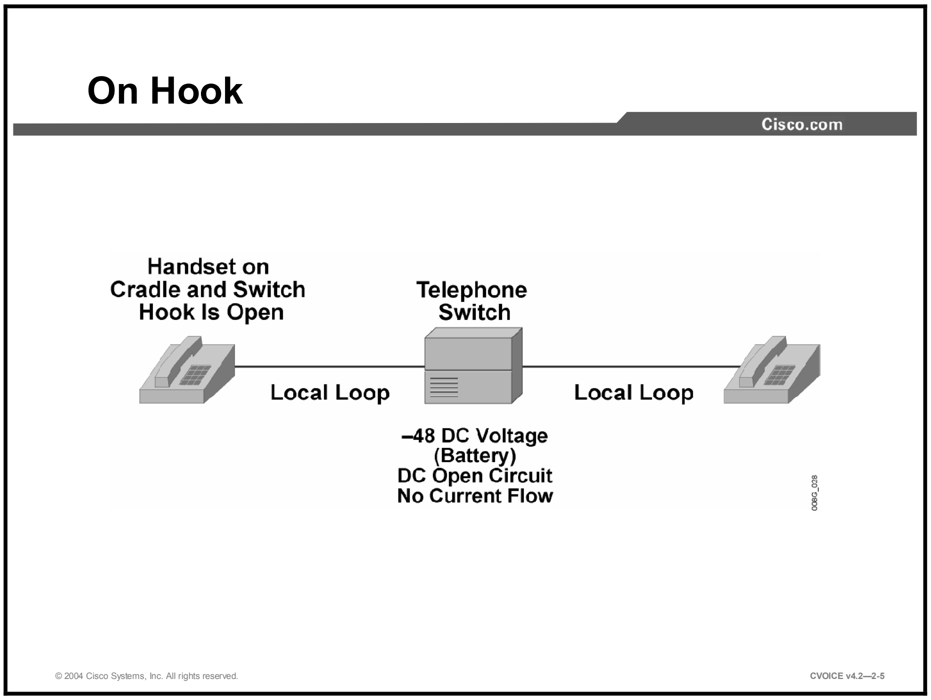
- **Supervisory signaling**
- **Address signaling**
- **Informational signaling**

 CVOICE v4.2—2-4

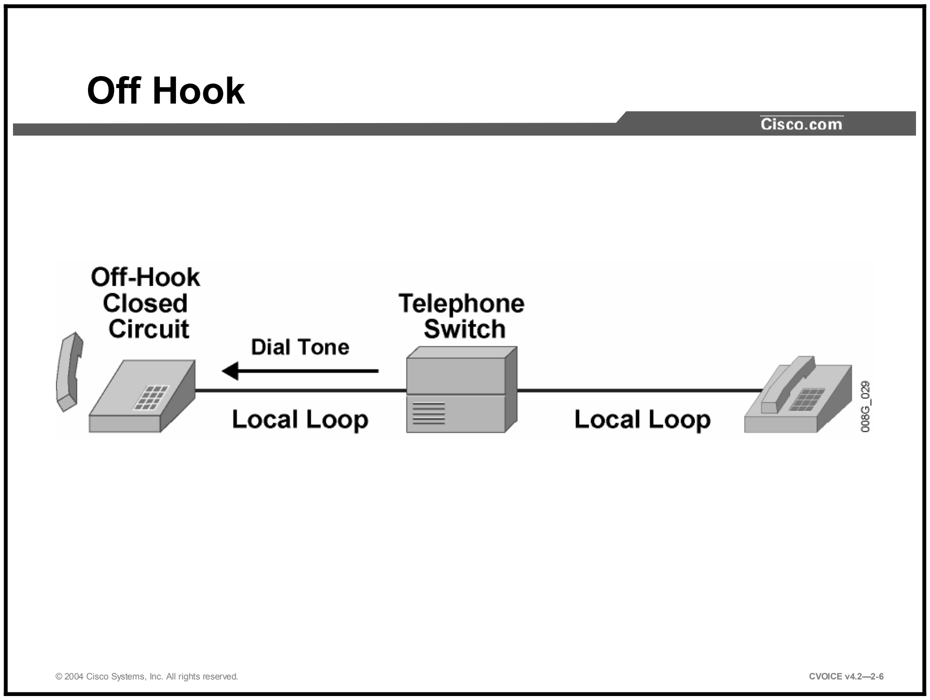
A subscriber and telephone company notify each other of the call status through audible tones and an exchange of electrical current. This exchange of information is called local-loop signaling. Local-loop signaling consists of supervisory signaling, address signaling, and informational signaling, each of which has their own characteristics and purpose. The three types of local-loop signaling appear on the local loop and serve to prompt the subscriber and the switch into a certain action.

Supervisory Signaling

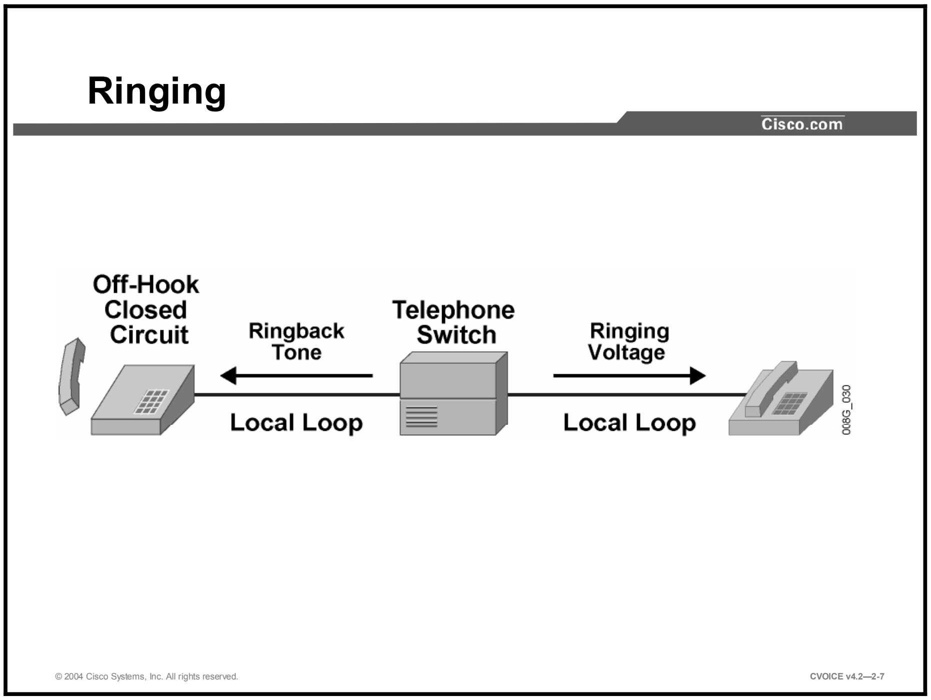
This topic describes on-hook, off-hook, and ringing supervisory signaling. Supervisory signaling serves to initiate the interaction between the subscriber and the attached switch.



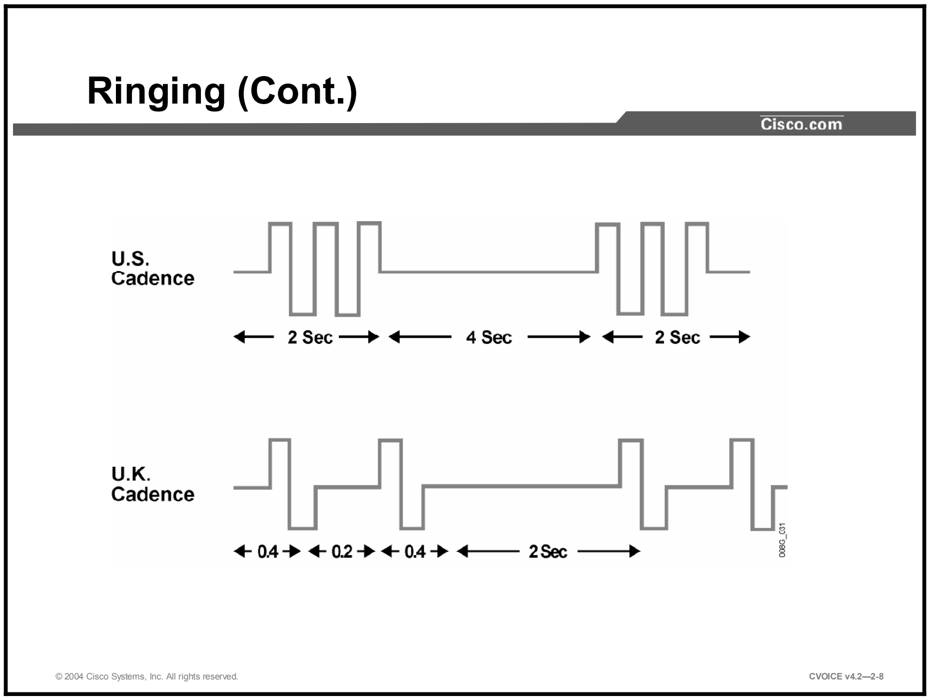
Resting the handset on the telephone cradle opens the switch hook and prevents the circuit current from flowing through the telephone. Regardless of the signaling type, a circuit goes on hook when the handset is placed on the telephone cradle and the switch hook is toggled to an open state. When the telephone is in this position, only the ringer is active.



To place a call, a subscriber must lift the handset from the telephone cradle. Removing the handset from the cradle places the circuit off hook. The switch hook is then toggled to a closed state, causing circuit current to flow through the electrical loop. The current notifies the telephone company that someone is requesting to place a telephone call. When the telephone network senses the off-hook connection by the flow of current, it provides a signal in the form of the dial tone to indicate that it is ready.



When a subscriber makes a call, the telephone sends voltage to the ringer to notify the other subscriber of an inbound call. The telephone company also sends a ringback tone to the caller, alerting the caller that it is sending ringing voltage to the recipient telephone. Although ringback tone is not the same as ringing voltage, it sounds similar.



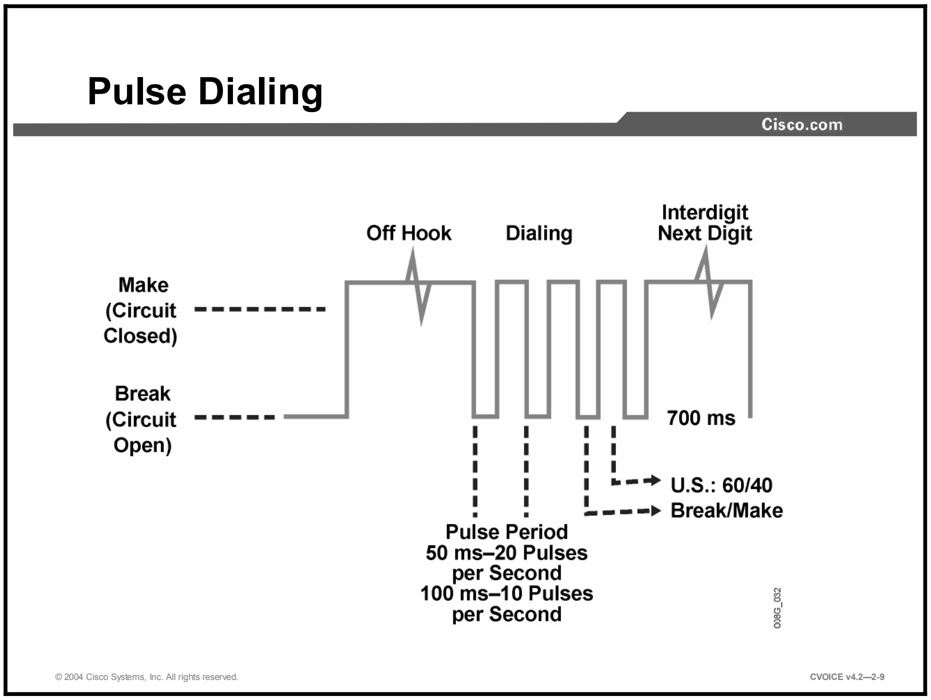
As depicted in the figure, the ringing supervisory tone in the United States is 2 seconds of tone followed by 4 seconds of silence. The United Kingdom uses a double ring of 0.4 seconds separated by 0.2 seconds of silence, followed by 2 seconds of silence.

Example: Ringing Cadences

The pattern of the ring signal, or ring cadence, varies around the world. In the United States, the ring signal, sent by the local service provider, is 2 seconds of ring followed by 4 seconds of silence. Your home telephone rings with this cadence when you have an incoming call.

Address Signaling

This topic describes pulse dialing and dual-tone multifrequency (DTMF) signaling.

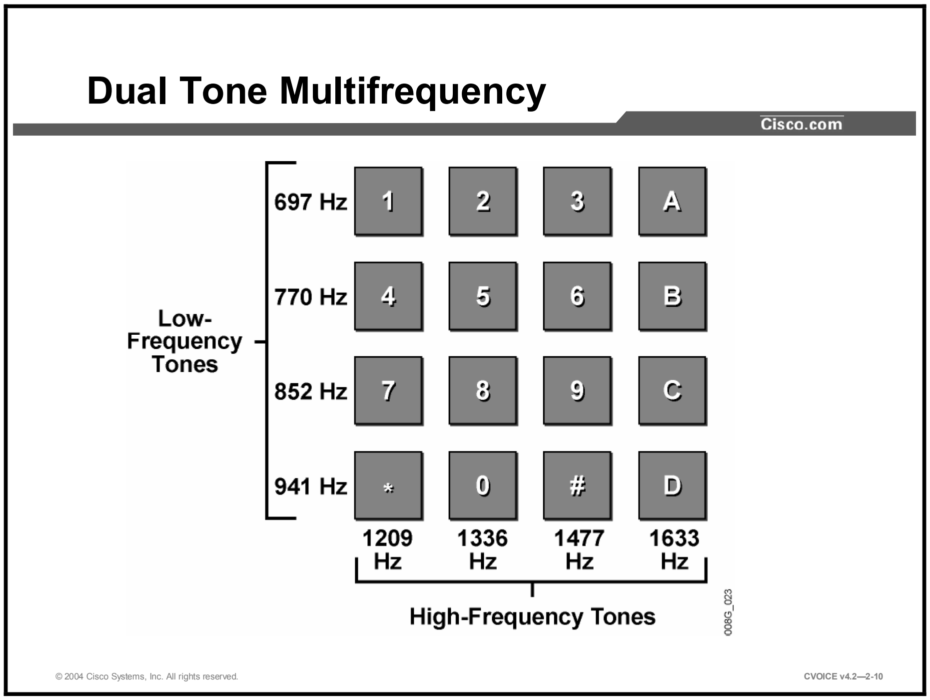


Although somewhat outdated, rotary-dial telephones are still in use and easily recognized by their large numeric dial-wheel. When placing a call, the subscriber spins the large numeric dial-wheel to send digits. These digits must be produced at a specific rate and within a certain level of tolerance. Each pulse consists of a "break" and a "make." The break segment is the time that the circuit is open. The make segment is the time during which the circuit is closed. In the United States, the break-and-make cycle must correspond to a ratio of 60 percent break to 40 percent make.

A governor inside the dial controls the rate at which the digits are pulsed.

The dial pulse signaling process occurs as follows:

1. When a subscriber calls someone by dialing a digit on the rotary dial, a spring winds.
2. When the dial is released, the spring rotates the dial back to its original position.
3. While the spring rotates the dial back to its original position, a cam-driven switch opens and closes the connection to the telephone company. The number of consecutive opens and closes—or breaks and makes—represents the dialed digit.



Users who have a touch-tone pad or a push-button telephone must push the keypad buttons to place a call. Each button on the keypad is associated with a set of high and low frequencies. Each row of keys on the keypad is identified by a low-frequency tone; each column of keys on the keypad is identified by a high-frequency tone. The combination of both tones notifies the telephone company of the number being called, hence the term dual tone multifrequency (DTMF).

The figure illustrates the combination of tones generated for each button on the keypad.

Informational Signaling

This topic lists the call-progress indicators and describes their functions.

Informational Signaling with Call-Progress Indicators

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Tone	Frequency (Hz)	On Time (Sec) - Off Time (Sec)	
Dial	350 + 440	Continuous	
Busy	480 + 620	0.5	0.5
Ringback, line	440 + 480	2	4
Ringback, PBX	440 + 480	1	3
Congestion (toll)	480 + 620	0.2	0.3
Reorder (local)	480 + 620	0.3	0.2
Receiver off hook	1400 + 2060 + 2450 + 2600	0.1	0.1
No such number	200 + 400	Continuous	
Confirmation tone		Freq. Mod 1 kHz	

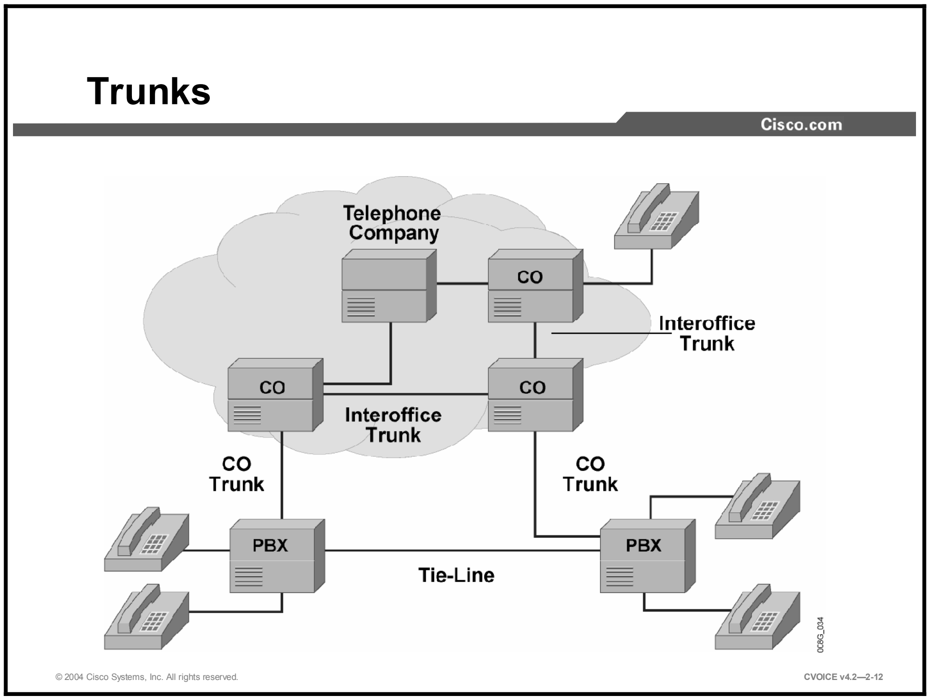
CVOICE v4.2—2-11

Call-progress indicators in the form of tone combinations are used to notify subscribers of call status. Each combination of tones represents a different event in the call process, as follows:

- **Dial tone:** Indicates that the telephone company is ready to receive digits from the user telephone. The Cisco routers provide dial tone as a method of showing that the hardware is installed. In a PBX or key telephone system, the dial tone indicates that the system is ready to receive digits.
- **Busy tone:** Indicates that a call cannot be completed because the telephone at the remote end is already in use.
- **Ringback (normal or PBX):** Indicates that the telephone company is attempting to complete a call on behalf of a subscriber.
- **Congestion:** Indicates that congestion in the long-distance telephone network is preventing a telephone call from being processed. The congestion tone is sometimes known as the all-circuits-busy tone.
- **Reorder tone:** Indicates that all of the local telephone circuits are busy, thus preventing a telephone call from being processed. The reorder tone is known to the user as fast-busy, and is familiar to anyone who operates a telephone from a PBX.
- **Receiver off hook:** Indicates that the receiver has been off hook for an extended period without placing a call.
- **No such number:** Indicates that a subscriber placed a call to a nonexistent number.
- **Confirmation tone:** Indicates that the telephone company is working on completing the call.

Trunk Connections

This topic describes the different types of trunks on a voice network and how they operate.



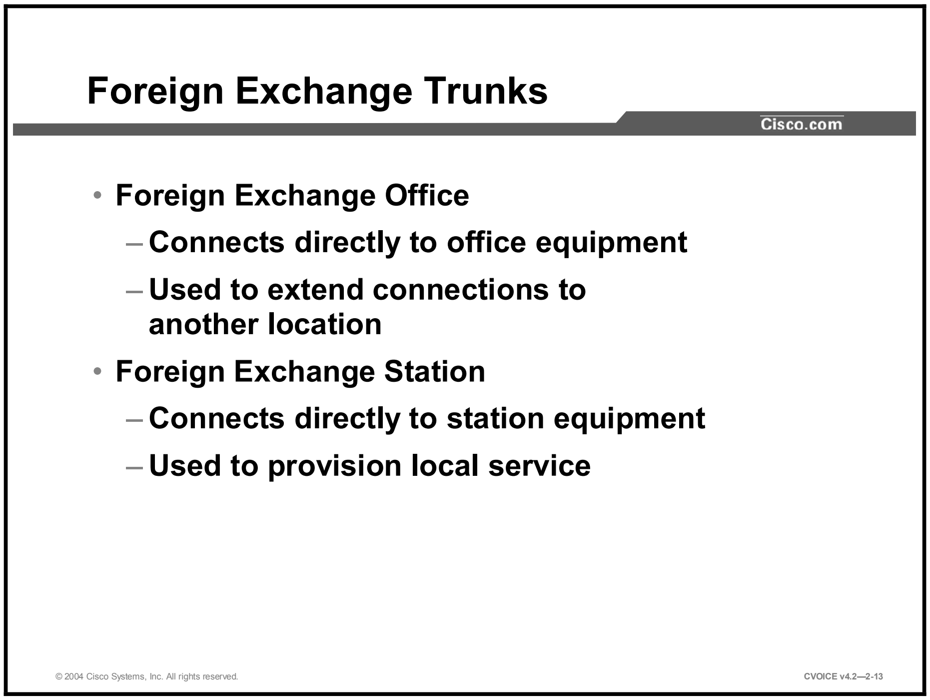
Before a telephone call terminates at its final destination, it is routed through multiple switches. When a switch receives a call, it determines whether the destination telephone number is within a local switch or if the call needs to go through another switch to a remote destination. Trunks connect the telephone company and PBX switches.

The primary function of the trunk is to provide the path between switches. The switch must route the call to the correct trunk or telephone line. Although many different subscribers share a trunk, only one subscriber uses it at any given time. As telephone calls end, they release trunks and make them available to the switch for subsequent calls. There can be several trunks between two switches.

The following are examples of the more common trunk types:

- **Private trunk lines:** Companies with multiple PBXs often connect them with tie trunk lines. Generally, tie trunk lines serve as dedicated circuits that connect PBXs. On a monthly basis, subscribers lease trunks from the telephone company to avoid the expense of using telephone lines on a per-call basis. These types of connections, known as tie-lines, typically use special interfaces called recEive and transMit, or ear and mouth (E&M), interfaces.
- **CO trunks:** A CO trunk is a direct connection between a PBX and the local CO that routes calls; for example, the connection from a private office network to the public switched telephone network (PSTN). When users dial **9**, they are connecting through their PBX to the CO trunk to access the PSTN. CO trunks typically use Foreign Exchange Office (FXO) interfaces. Certain specialized CO trunks are frequently used on the telephony network. A direct inward dialing (DID) trunk, for example, allows outside callers to reach specific internal destinations without having to be connected via an operator.

- **Interoffice trunks:** An interoffice trunk is a circuit that connects two local telephone company COs.



Foreign exchange (FX) trunks are interfaces that are connected to switches that support connection to either office equipment or station equipment. Office equipment includes other switches (to extend the connection) and Cisco devices. Station equipment includes telephones, fax machines, and modems.

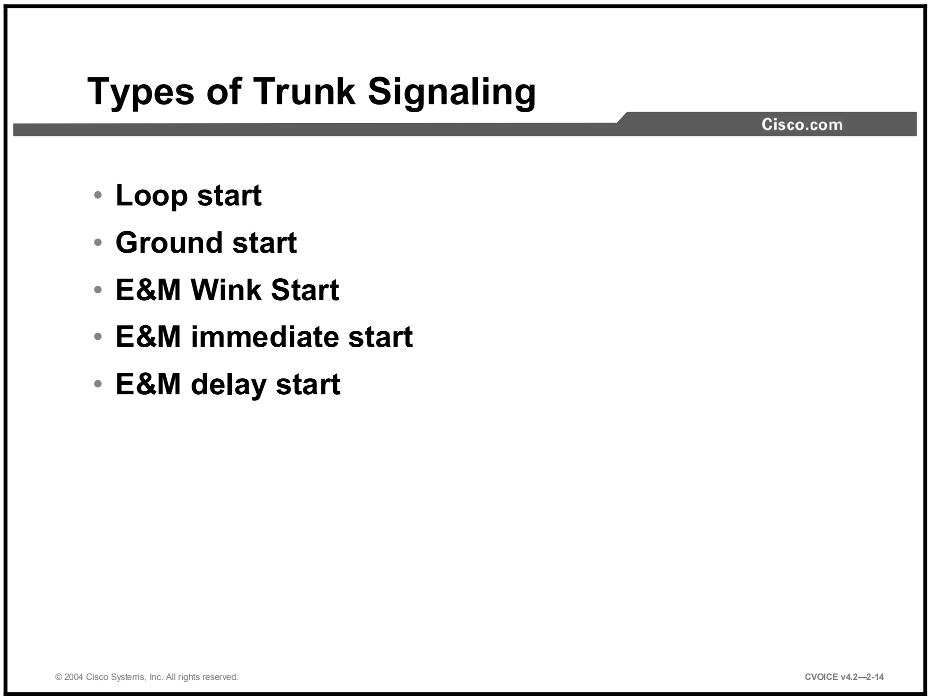
- **Foreign Exchange Office (FXO) interfaces:** An FXO interface connects a PBX to another switch or Cisco device. The purpose of an FXO interface is to extend the telephony connection to a remote site; for example, if a user on a corporate PBX wanted a telephone installed at home instead of in the local office where the PBX is located, an FXO interface would be used. The FXO interface would connect to a Cisco voice router, which would serve to extend the connection to the user home. This connection is an Off-Premises eXtension (OPX).
- **Foreign Exchange Station (FXS) interfaces:** An FXS interface connects station equipment: telephones, fax machines, and modems. A telephone connected directly to a switch or Cisco device requires an FXS interface. Because a home telephone connects directly to the telephone company CO switch, an FXS interface is used.

Example: Foreign Exchange Interfaces

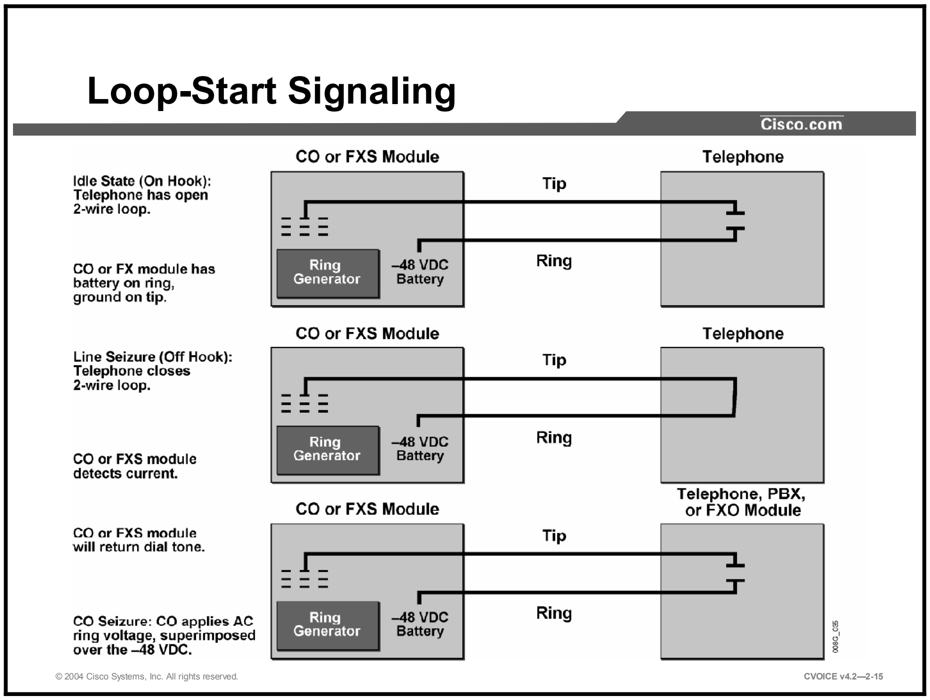
The service provided by local telephone companies for residential phones uses a foreign exchange interface—specifically FXS. This service is provided on two wires. The service is considered a station-side connection because the interface terminates with a telephone.

Types of Trunk Signaling

This topic describes the trunk and line-seizure signaling types.



There must be signaling standards between the lines and trunks of a telephone network, just as there are signaling standards between a telephone and the telephone company. Trunk signaling serves to initiate the connection between the switch and the network. There are five different types of trunk signaling and each applies to different kinds of interfaces, such as FXS, FXO, and E&M.



Loop-start signaling allows a user or the telephone company to seize a line or trunk when a subscriber is initiating a call. It is primarily used on local loops rather than on trunks.

A telephone connection exists in one of the following states:

- Idle (on hook)
- Telephone seizure (off hook)
- CO seizure (ringing)

A summary of the loop-start signaling process is as follows:

1. When the line is in the idle state, or on hook, the telephone or PBX opens the two-wire loop. The CO or FXS has *battery* on ring and *ground* on tip.

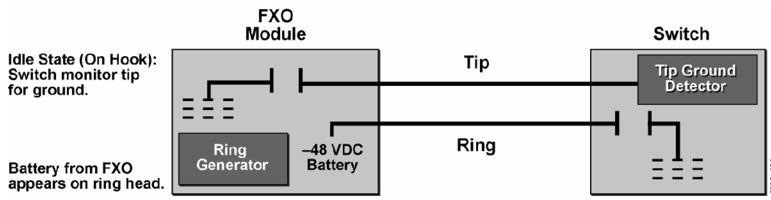
2. If a user lifts the handset off the cradle to place a call, the switch hook goes off hook and closes the loop (line seizure). The current can now flow through the telephone circuit. The CO or FXS module detects the current and returns a dial tone.

3. When the CO or FXS module detects an incoming call, it applies AC ring voltage superimposed over the –48 VDC battery, causing the ring generator to notify the recipient of a telephone call. When the telephone or PBX answers the call, thus closing the loop, the CO or FXS module removes the ring voltage.

Loop-start signaling is a poor solution for high-volume trunks because it leads to glare incidents, or the simultaneous seizure of the trunk from both ends. Glare occurs, for example, when you pick up your home telephone and find that someone is already at the other end.

Glare is not a significant problem at home. It is, however, a major problem when it occurs between switches at high-volume switching centers, such as long-distance carriers.

Ground-Start Signaling

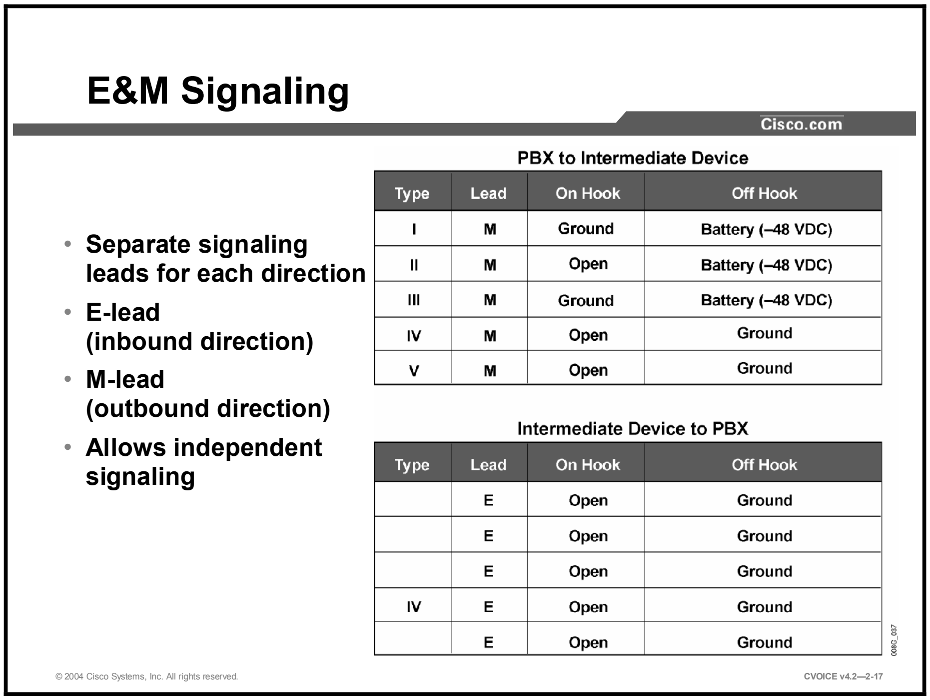


Ground-start signaling is a modification of loop-start signaling that corrects for the probability of glare. It solves the problem by providing current detection at both ends.

Although loop-start signaling works when you use your telephone at home, ground-start signaling is preferable when there are high-volume trunks involved at telephone switching centers. Because ground-start signaling uses a request or confirm switch at both ends of the interface, it is preferable over other signaling methods on high-usage trunks, such as FXOs.

FXOs require implementation of answer supervision (reversal or absence of current) on the interface for the confirmation of on hook or off hook.

Ground-start signaling is not common in Voice over IP (VoIP) networks.



E&M signaling supports tie-line type facilities or signals between voice switches. Instead of superimposing both voice and signaling on the same wire, E&M uses separate paths, or leads, for each.

Example: E&M Signaling

To call a remote office, your PBX must route a request for use of the trunk over its signal leads between the two sites. Your PBX makes the request by activating its M-lead. The other PBX detects the request when it detects current flowing on its E-lead. It then attaches a dial register to the trunk and your PBX, which sends the dialed digits. The remote PBX activates its M-lead to notify the local PBX that the call has been answered.

Types of E&M Signaling

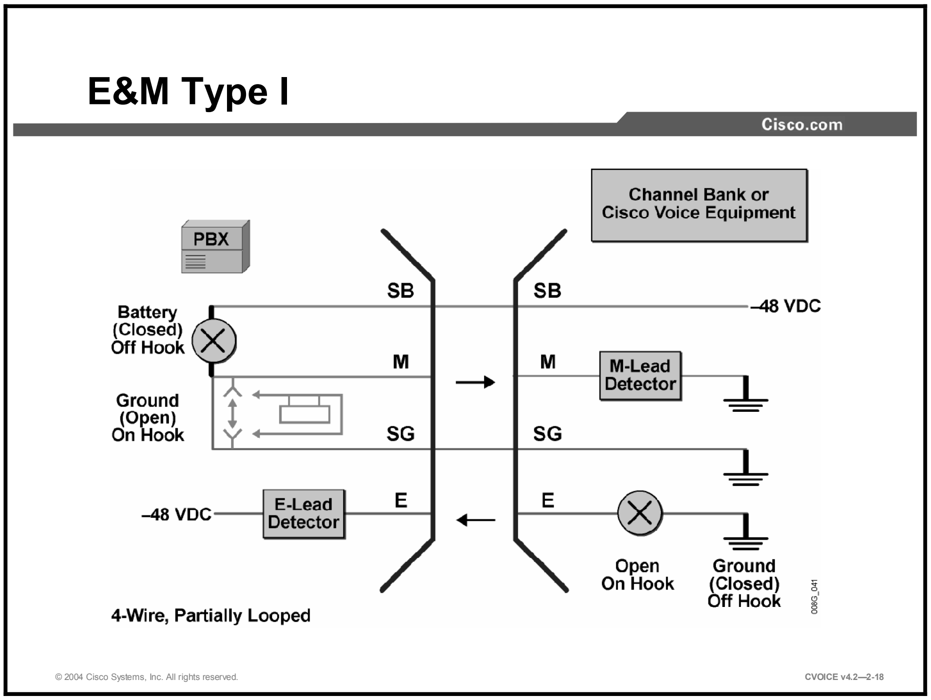
There are five types of E&M signaling: Type I, Type II, Type III, Type IV, and Type V. The E&M leads operate differently with each wiring scheme, as shown in the table.

Types of E&M Signaling

E&M Signaling Type	PBX to Intermediate Device			Intermediate Device to PBX		
	Lead	On Hook	Off Hook	Lead	On Hook	Off Hook
Type I	M	Ground	Battery (–48 VDC)	E	Open	Ground
Type II	M	Open	Battery (–48 VDC)	E	Open	Ground
Type III	M	Ground	Battery (–48 VDC)	E	Open	Ground
Type IV	M	Open	Ground	E	Open	Ground
Type V	M	Open	Ground	E	Open	Ground

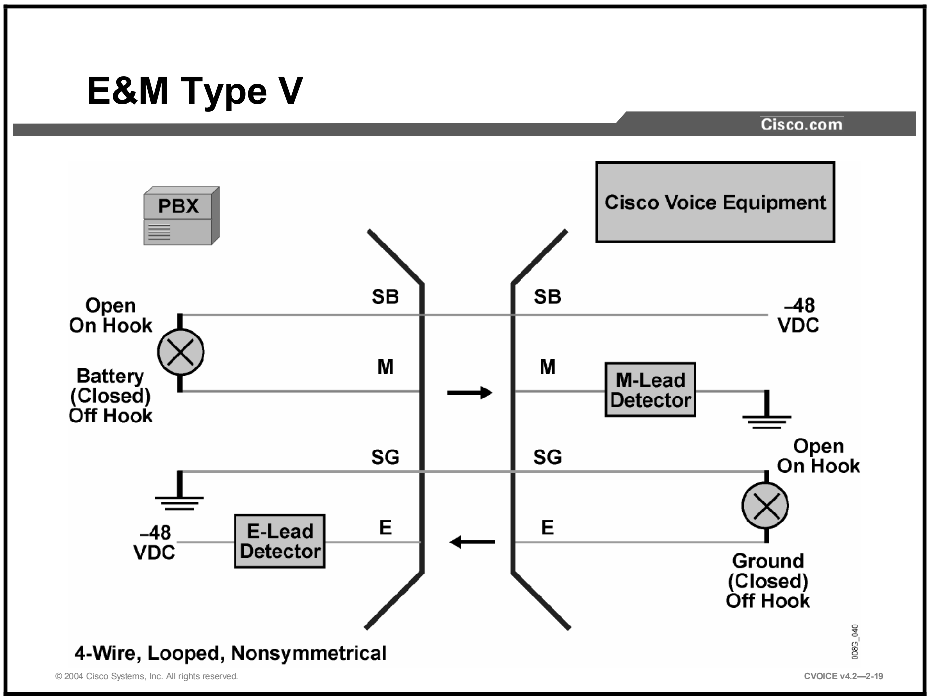
E&M Signaling Types

This topic identifies the five E&M signaling types and provides a description of each.



Four-wire E&M Type I signaling is actually a six-wire E&M signaling interface common in North America. One wire is the E-lead; the second wire is the M-lead, and the remaining two pairs of wires serve as the audio path. In this arrangement, the PBX supplies power, or *battery,* for both the M-leads and E-leads. This arrangement also requires that a common ground be connected between the PBX and the Cisco voice equipment.

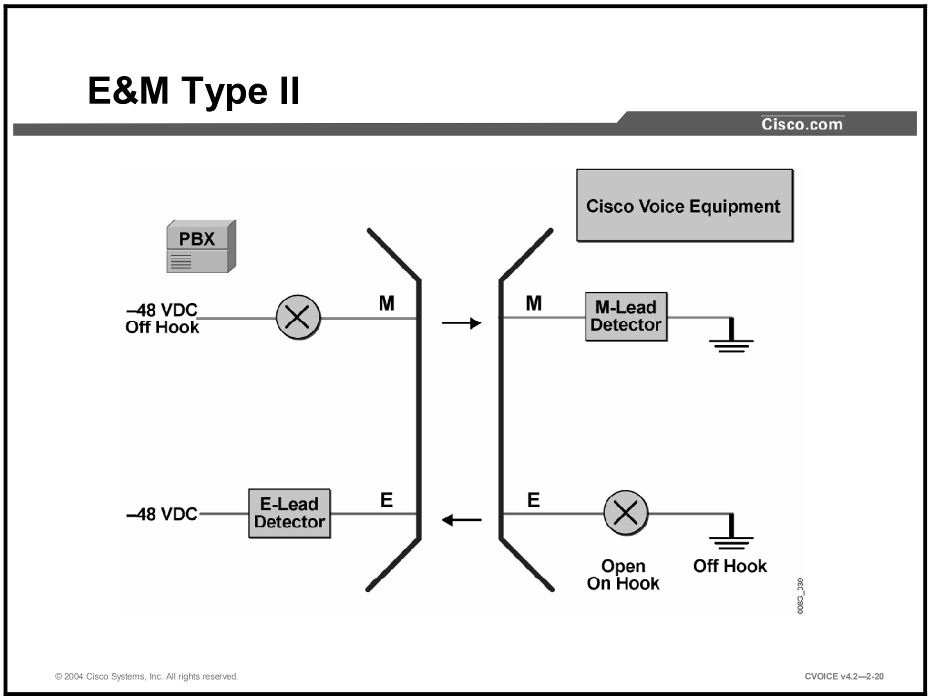
With the Type I interface, the Cisco voice equipment (tie-line equipment) generates the E signal to the PBX by grounding the E-lead. The PBX detects the E signal by sensing the increase in current through a resistive load. Similarly, the PBX generates the M signal by sourcing a current to the Cisco voice equipment (tie-line equipment), which detects it via a resistive load.



Type V is another six-wire E&M signaling type and the most common E&M signaling form outside of North America. In Type V, one wire is the E-lead and the other wire is the M-lead.

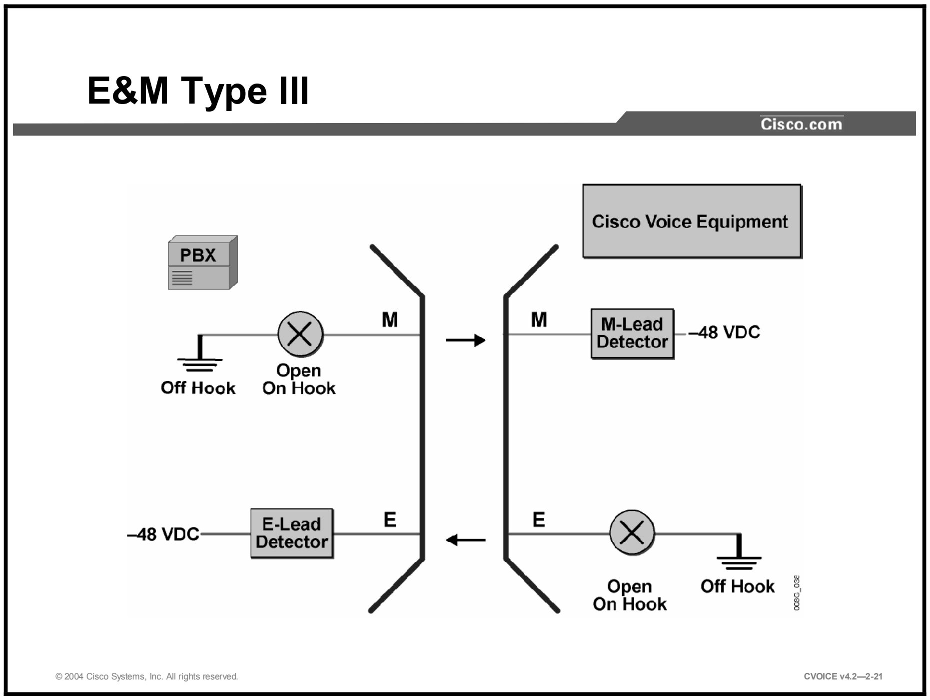
Type V is a modified version of the Type I interface. In the Type V interface, the Cisco voice equipment (tie-line equipment) supplies battery for the M-lead while the PBX supplies battery for the E-lead. As in Type I, Type V requires that a common ground be connected between the PBX and the Cisco voice equipment.

E&M Type II

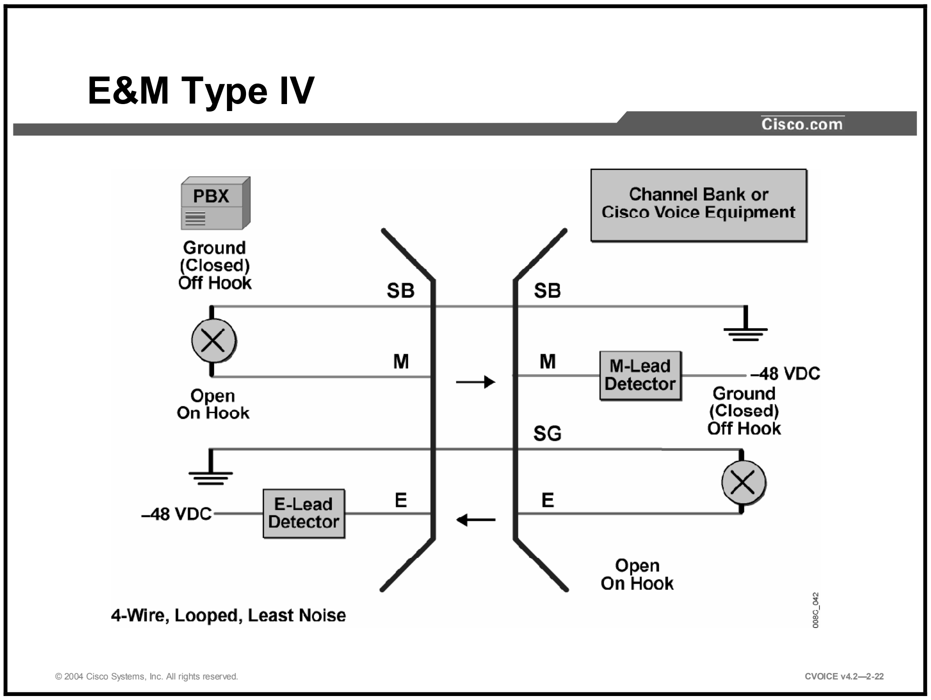


Types II, III, and IV are eight-wire interfaces. One wire is the E-lead, the other wire is the M-lead. Two other wires are signal ground (SG) and signal battery (SB). In Type II, SG and SB are the return paths for the E-lead and M-lead, respectively.

The Type II interface exists for applications where a common ground between the PBX and the Cisco voice equipment (tie-line equipment) is not possible or practical; for example, the PBX is in one building on a campus and the Cisco equipment is in another. Because there is no common ground, each of the signals has its own return. For the E signal, the tie-line equipment permits the current to flow from the PBX; the current returns to the PBX SG lead or reference. Similarly, the PBX closes a path for the current to generate the M signal to the Cisco voice equipment (tie-line equipment) on the SB lead.



Type III is useful for environments where the M-lead is likely to experience electrical interference and falsely signal its attached equipment. When idle, Type III latches the M-lead via an electrical relay to the SG lead. When the PBX activates the M-lead, it first delatches the SG lead via the relay and signals normally, as in Type II. Type III is not a common implementation.

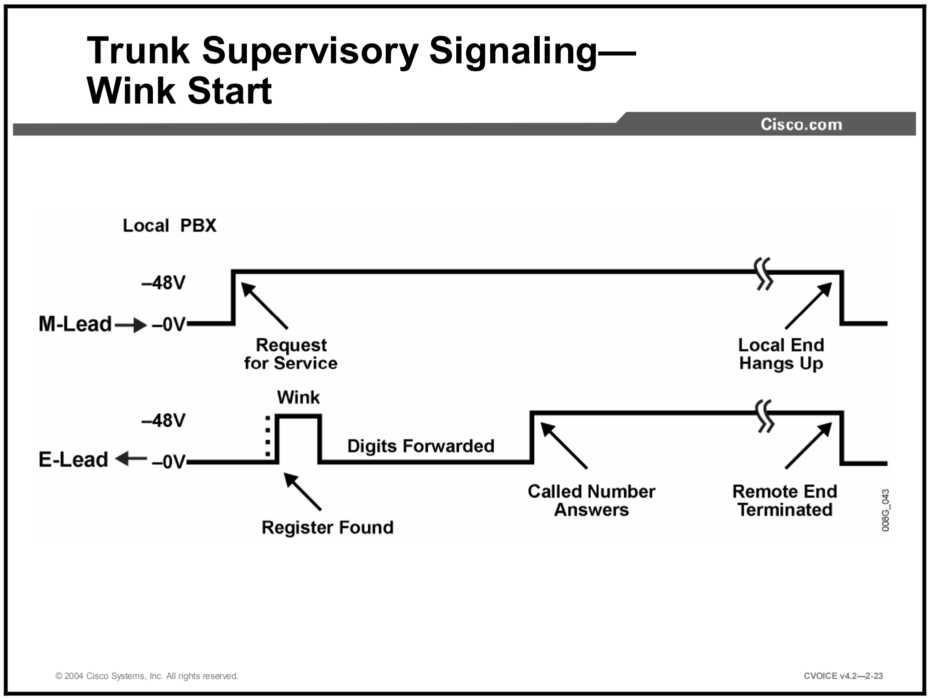


Type IV is a variation of Type II. In this arrangement, the battery source and ground are reversed on the SB and M wires (as compared to Type II). This means that both the SB and SG wires are grounded. Type IV signaling is symmetric and requires no common ground. Each side closes a current loop to signal, which detects the flow of current through a resistive load to indicate the presence of the signal. Cisco voice equipment does not support Type IV.

The most common E&M signaling interface in the United States is Type I, and the most common in European countries is Type V. Other variations exist for special applications and purposes. Cisco does not support Type IV.

Trunk Signal Types Used by E&M

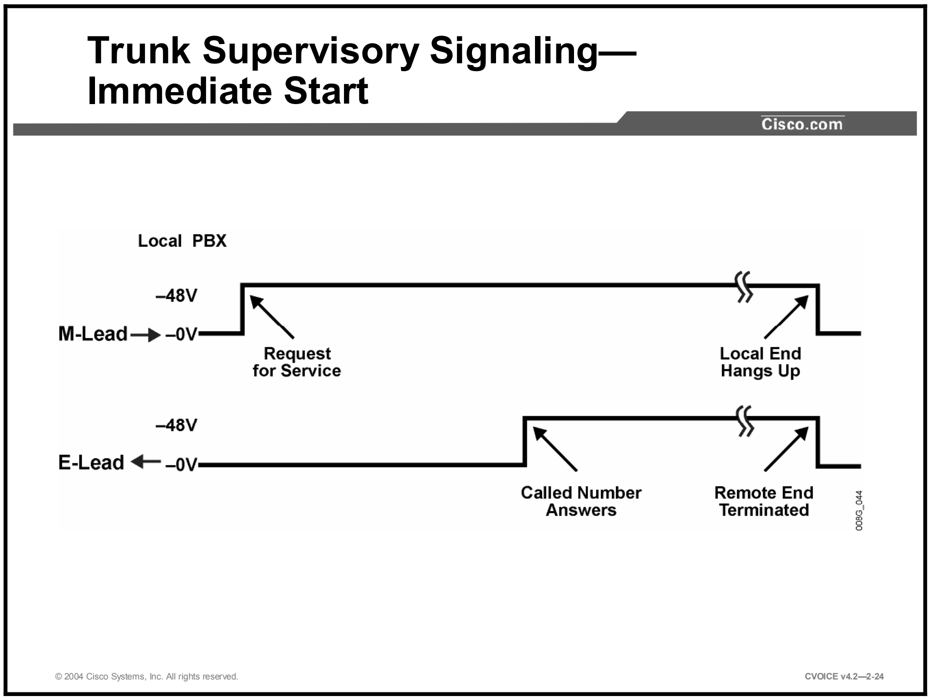
This topic describes Wink-Start, immediate-start, and delay-start signaling as used by E&M signaling.



Tie trunks have bidirectional supervisory signaling that allows either end to initiate a trunk seizure. In this way, one PBX seizes the trunk, which then waits for an acknowledgment reply from the remote end. The local end must differentiate between a return acknowledgment and a remote-end request for service. Wink-Start signaling is the most common E&M trunk seizure signal type.

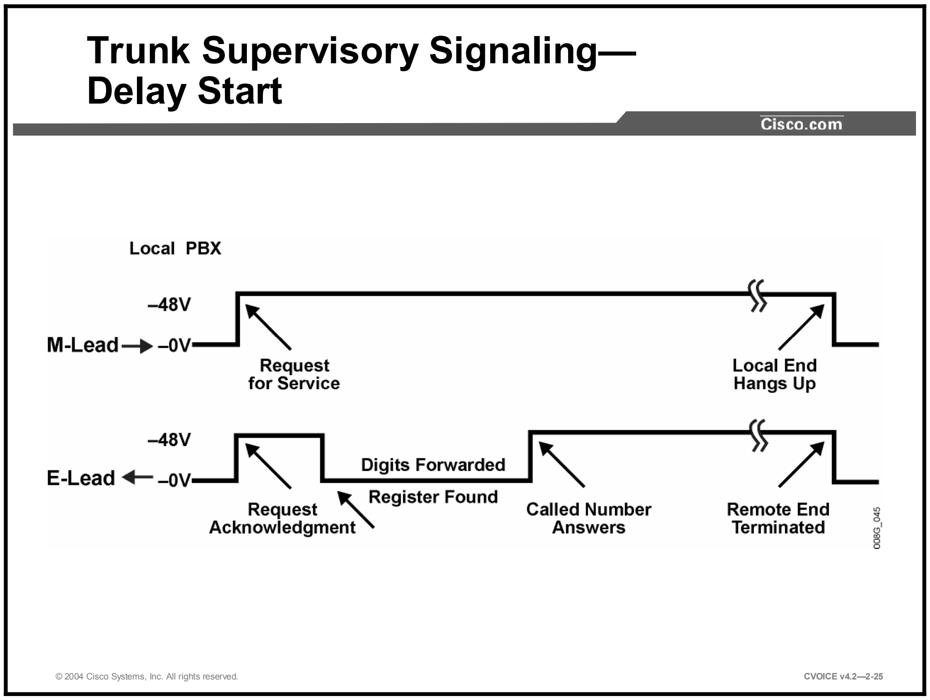
The following scenario summarizes the Wink-Start protocol event sequence:

1. The calling office seizes the line by activating its M-lead.
2. Instead of returning an off-hook acknowledgment immediately, the called switch allocates memory for use as a dial register, in the area of memory it uses to store incoming digits.
3. The called switch toggles its M-lead on and off for a specific time (usually 170 to 340 ms). (This on-hook/off-hook/on-hook sequence constitutes the *wink*.)
4. The calling switch receives the wink on its E-lead and forwards the digits to the remote end. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
5. The called party answers the telephone, and the called PBX raises its M-lead for the duration of the call.



If the timing of the returned wink is too short or impossible to detect, the trunk uses immediate start. This occurs occasionally if a PBX vendor implements Wink Start but does not conform to the standards. The following scenario summarizes the sequence of events for the immediate-start protocol:

1. The calling PBX seizes the line by activating its M-lead.
2. Instead of receiving an acknowledgment, the calling PBX waits a predetermined period (a minimum of 150 ms) and forwards the digits blindly. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
3. The called PBX acknowledges the calling PBX only after the called party answers the call by raising its M-lead.

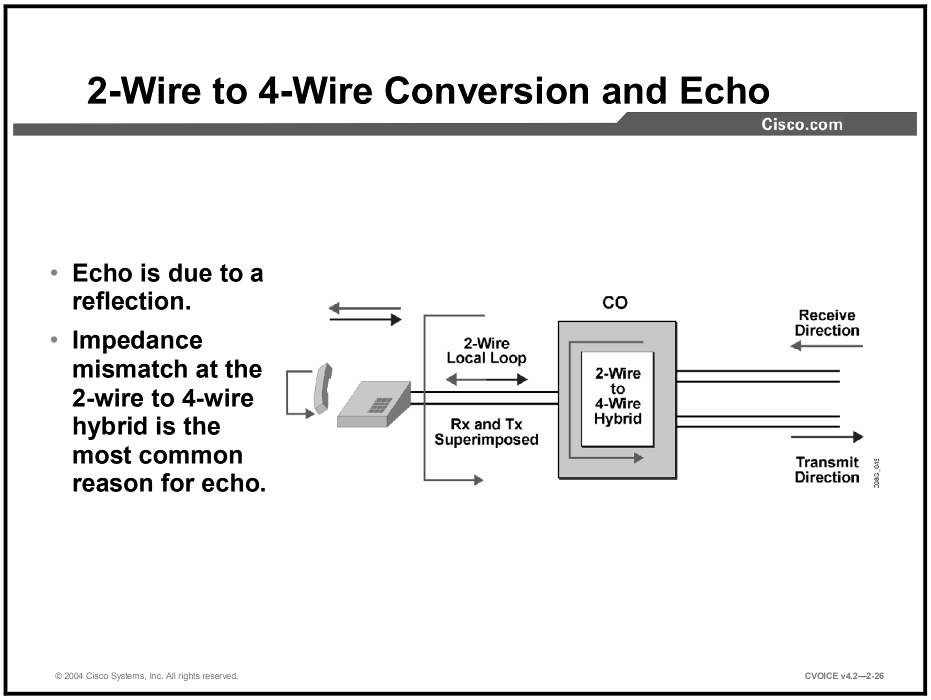


Delay start is the original start protocol for E&M. It is used when all of the equipment is mechanical and requires time to process requests. The following scenario summarizes delay-start signaling:

1. When you place a call, your calling switch goes off hook by activating its M-lead.
2. The called switch acknowledges the request by activating its M-lead, and then rotates armatures and gears to reset its dial register to zero.
3. When the dial register at the called switch is in the ready state, the called switch deactivates its M-lead.
4. The calling switch then sends dialed digits. DTMF tones are forwarded across the E&M link in the audio path, not on the M-lead.
5. When the called party answers, the called switch again activates its M-lead.

Line Quality

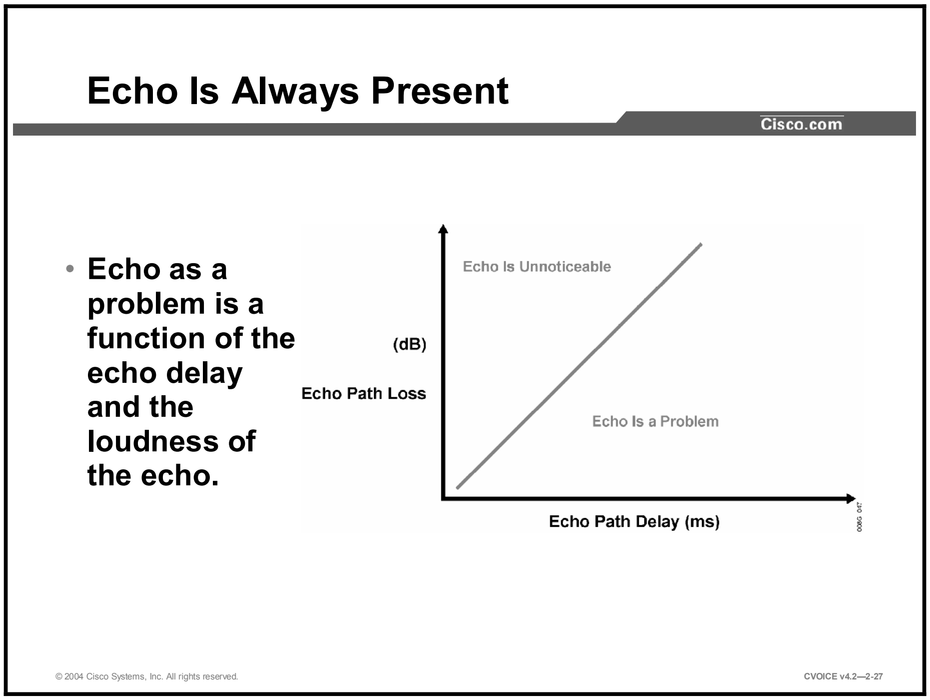
This topic describes impairments commonly found in analog telephone circuits and offers solutions to the problem of echo.



Although a local loop consists of two wires, when it reaches the switch, the connection changes to four wires with a two- to four-wire hybrid converter. Trunks then transport the signal across the network.

Telephone networks can experience two types of echo: acoustic echo and electrical echo. Acoustic echo frequently occurs with speakerphones, when the received voice on the speaker excites the microphone and travels back to the speaker. Electrical echo occurs when there is an electrical inconsistency in the telephony circuits. This electrical inconsistency is called impedance mismatch.

If the lines have a good impedance match, the hybrid is considered balanced, with little or no reflected energy. However, if the hybrid is inadequately balanced, and a portion of the transmit voice is reflected back toward the receive side, echo results.



Some form of echo is always present. However, echo can become a problem under the following conditions:

- The magnitude or loudness of the echo is high.
- The delay time between when you speak and when you hear your voice reflected is significant.
- The listener hears the speaker twice.

The two components of echo are loudness and delay. Reducing either component reduces overall echo. When a user experiences delay, the conversation can get choppy, and the words of the participants sometimes overlap.

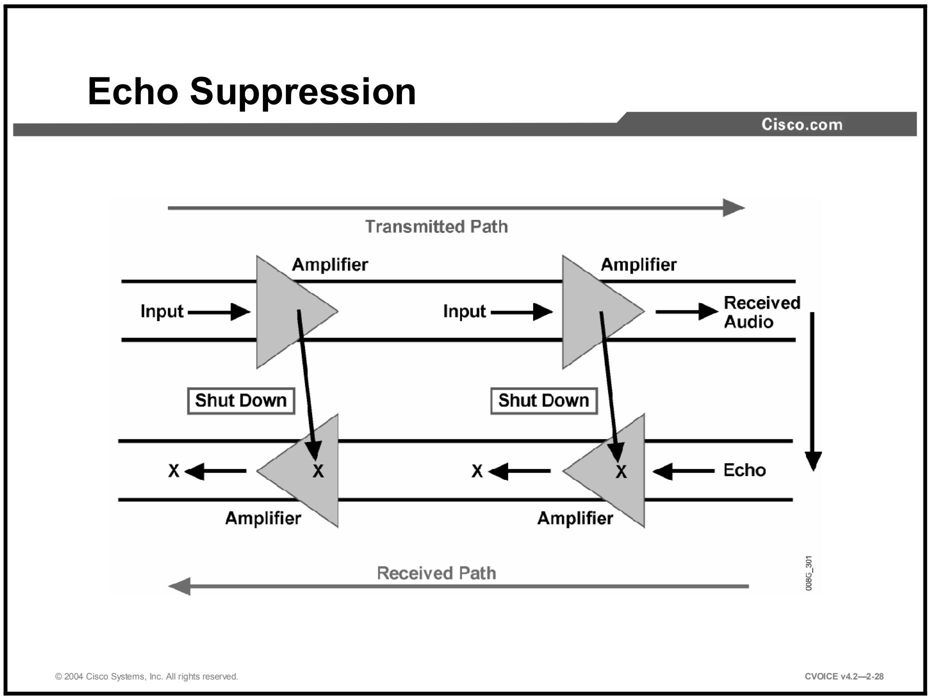
Note	Echo tolerance varies. For most users, however, echo delay over 50 ms is generally problematic.

Management of Echo

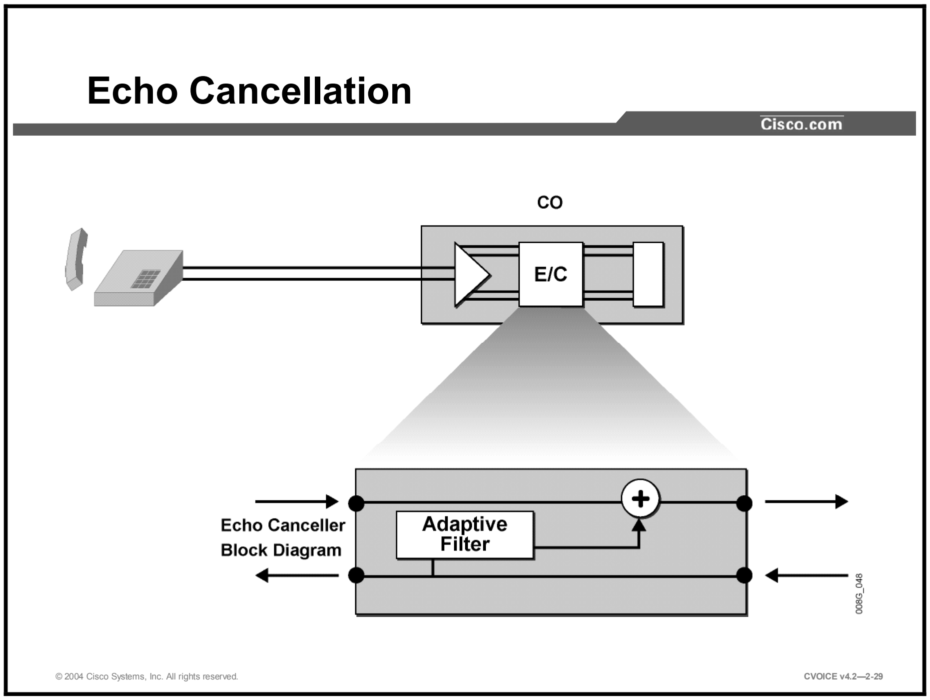
There are two ways to solve an echo problem in your telephone network:

- Echo suppression
- Echo cancellation

This topic explains and compares the two approaches to echo management.



The echo suppressor works by transmitting speech in the forward direction and prohibiting audio in the return direction. The echo suppressor essentially breaks the return transmission path. This solution works sufficiently for voice transmission. However, for full-duplex modem connections, the action of the echo suppressor prevents communication. Therefore, when modems handshake, the answering modem returns a tone of 2025 Hz to the calling modem, which serves to disable the echo suppressors along the transmission path.



Echo suppression has shortcomings in addressing certain echo conflict situations. Echo cancellation is a more sophisticated method of eliminating echo.

Rather than breaking or attenuating the return path (as in echo suppression), echo cancellation uses a special circuit to build a mathematical model of the transmitted speech pattern and subtract it from the return path. This echo elimination method is depicted in the figure.

Note Echo cancellation applies the same technology that is used in audio headphones to cancel ambient noise.

Echo cancellation is the most common method of removing echo in the telephone network today, and is used when it is necessary to adjust for echo on a Cisco device.

Note The echo canceller removes the echo from one end of the circuit only. If echo is an issue at both ends of the circuit, you must apply another echo canceller at the other end.

Example: Echo Cancellation

The headsets used by airline pilots feature a suppression circuit, which cancels ambient noise so that the pilot hears only the audio from the headset. Any ambient noise from the cockpit is cancelled. This is the same technology used in echo cancellers.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- A local loop consists of a pair of twisted wires.
- The three types of local-loop signaling are supervisory, address, and informational signaling.
- The three types of supervisory signaling are on hook, off hook, and ringing.
- Pulse dialing and dual tone multifrequency are two types of address signaling.
- Call-progress indicators are used to notify subscribers of call status.
- The primary function of a trunk is to provide a path between switches.

 CVOICE v4.2—2-30

Summary (Cont.)

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- The types of trunk signaling include loop start, ground start, E&M Wink Start, E&M immediate start, and E&M delay start.
- There are five E&M signaling types: Type I, Type II, Type III, Type IV, and Type V.
- Three signal types used by E&M are Wink Start, immediate start, and delay start.
- The two components of echo that impair line quality are delay and amplitude.
- Two methods to eliminate echo are suppression and cancellation.

 CVOICE v4.2—2-31

References

For additional information, refer to this resource:

- McQuerry, S., McGrew, K., Foy, S. *Cisco Voice over Frame Relay, ATM, and IP*. Indianapolis, Indiana: Cisco Press; 2001.

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 2-1: Lab Familiarity

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) In most local-loop connections, to what does the ring wire tie?

A) battery

B) ground

C) telephone

D) switch

Q2) At what point does current flow around the loop?

A) when the ring wire connects to the negative side of a power source

B) when the tip wire connects to the ground

C) when the telephone goes off hook

D) when the telephone goes on hook

Q3) What are the three different types of local-loop signaling? (Choose three.)

A) address signaling

B) coding signaling

C) control signaling

D) informational signaling

E) remote signaling

F) supervisory signaling

Q4) What two tools do subscriber and telephone companies use to notify each other of call status? (Choose two.)

A) audible tones

B) digital signatures

C) electrical current

D) pulse packets

E) tagged packets

Q5) What three scenarios occur when the switch hook is in a closed state? (Choose three.)

A) only the ringer is active

B) the telephone is on hook

C) the current flows through the electrical loop

D) the current cannot flow through the electrical loop

Q6) What is the ring tone in the United States?

A) one 2-second tone followed by 2 seconds of silence

B) one 2-second tone followed by 4 seconds of silence

C) two 0.4-second rings separated by 0.2 seconds of silence, followed by 2 seconds of silence

D) two 2-second rings separated by 4 seconds of silence

Q7) In the United States, what is the required ratio of the break-and-make cycle for rotary-dial telephones?

A) 60 percent break to 40 percent make

B) 40 percent break to 60 percent make

C) 80 percent break to 20 percent make

D) 20 percent break to 80 percent make

Q8) On a DTMF phone, which two scenarios are associated with each button on the keypad? (Choose two.)

A) a series of pulses

B) a set of high frequencies for each button column

C) a break-and-make cycle

D) a set of low frequencies for each button row

Q9) Which call-progress indicator is used to notify customers that the telephone company is processing the call?

A) busy

B) confirmation tone

C) dial tone

D) ringback

Q10) Which tone is used to indicate that all the local telephone circuits are busy?

A) busy

B) ringback

C) all circuits busy

D) reorder

Q11) Match the trunk type with its description.

A) private trunk lines

B) CO trunks

C) interoffice trunks

D) FX trunks

_____ 1. trunk interfaces that connect to station equipment or office equipment

_____ 2. a trunk circuit that connects two local telephone company COs

_____ 3. a direct connection between a PBX and the local CO

_____ 4. dedicated circuits that connect PBXs

Q12) Which statement describes the problem of glare in loop-start signaling?

A) The trunk is simultaneously seized from both ends.

B) The signal is so loud that it cannot be understood.

C) The trunk is overloaded with too many messages.

D) The ring voltage is so high that it interferes with communication.

Q13) Which trunk signaling method uses separate wires for voice and signaling?

A) loop start

B) ground start

C) E&M

D) CAS

Q14) Which E&M signaling type is NOT supported by Cisco voice equipment?

A) Type I

B) Type II

C) Type III

D) Type IV

E) Type V

Q15) Which type of E&M signaling is useful when the PBX and the Cisco equipment are in different buildings on the campus?

A) Type I

B) Type II

C) Type III

D) Type IV

E) Type V

Q16) Which type of E&M signaling is used when all the equipment is mechanical?

A) Wink Start

B) immediate start

C) delay start

Q17) Which type of E&M signaling is the most commonly used?

A) Wink Start

B) immediate start

C) delay start

Q18) How much echo delay is generally acceptable?

A) below 50 ms

B) below 100 ms

C) below 120 ms

D) below 150 ms

Q19) Which two factors contribute to echo problems? (Choose two.)

A) attenuation

B) crosstalk

C) delay

D) electric current on the line

E) incorrect voltage

F) loudness

Q20) Which method is used to reduce echo on Cisco devices?

A) echo suppression

B) echo cancellation

C) echo correction

D) echo return loss

Q21) In what situations can echo suppression cause problems?

A) for voice transmission

B) for half-duplex modem connections

C) for full-duplex modem connections

D) for Group 6 fax

Q22) On what type of circuit can “glare” occur?

A) a loop-start circuit

B) a ground-start circuit

C) a Wink-Start circuit

D) a delay-start circuit

Quiz Answer Key

Q1) A

Relates to: Local-Loop Connections

Q2) C

Relates to: Local-Loop Connections

Q3) A, D, F

Relates to: Types of Local-Loop Signaling

Q4) A, C

Relates to: Types of Local-Loop Signaling

Q5) A, B, D

Relates to: Supervisory Signaling

Q6) B

Relates to: Supervisory Signaling

Q7) A

Relates to: Address Signaling

Q8) B and D

Relates to: Address Signaling

Q9) B

Relates to: Informational Signaling

Q10) D

Relates to: Informational Signaling

Q11) 1-D, 2-C, 3-B, 4-A

Relates to: Trunk Connections

Q12) A

Relates to: Types of Trunk Signaling

Q13) C

Relates to: Types of Trunk Signaling

Q14) D

Relates to: E&M Signaling Types

Q15) B

Relates to: E&M Signaling Types

Q16) C

Relates to: Trunk Signal Types Used by E&M

Q17) A

Relates to: Trunk Signal Types Used by E&M

Q18) A

Relates to: Line Quality

Q19) C, F

Relates to: Line Quality

Q20) B

Relates to: Management of Echo

Q21) C

Relates to: Management of Echo

Q22) A

Relates to: Types of Trunk Signaling

Understanding Analog-to-Digital Voice Encoding

Overview

This lesson covers the fundamentals of digital voice encoding. You will learn the basics of voice digitization and the various compression schemes that are used to transport voice while using less bandwidth. You will also learn about voice quality measurement techniques to help you choose the compression scheme that best suits your needs.

Relevance

To deploy VoIP networks, the voice must be digitized. Understanding how voice digitization works and what compression schemes are offered helps you to understand the bandwidth requirements for each type of compression.

Objectives

Upon completing this lesson, you will be able to choose a voice compression scheme that best suits your needs. This includes being able to meet these objectives:

- Identify the steps for converting analog signals to digital signals
- Identify the steps for converting digital signals to analog signals
- State the purpose of the Nyquist Theorem
- Explain quantization
- Name two types of voice compression techniques
- Describe the similarities and differences between G.729 and G.729 (Annex A) compression
- List three common voice compression standards and their bandwidth requirements
- State the purpose of voice quality measurement and the method of calculation for each type of measurement

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Understanding of analog voice telephony
- Familiarity with PBXs and voice switches
- Understanding of analog voice signaling

Outline

The outline lists the topics included in this lesson.

Outline

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- **Overview**
- **Basic Voice Encoding: Converting Analog to Digital**
- **Basic Voice Encoding: Converting Digital to Analog**
- **The Nyquist Theorem**
- **Quantization**
- **Voice Compression and Codec Standards**
- **G.729 and G.729A Compared**
- **Compression Bandwidth Requirements**
- **Voice Quality Measurement**
- **Summary**
- **Quiz**
- **Lab**

 CVOICE v4.2—2-2

Basic Voice Encoding: Converting Analog to Digital

This topic describes the process of converting analog signals to digital signals.

Digitizing Analog Signals

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1. **Sample the analog signal regularly.**
2. **Quantize the sample.**
3. **Encode the value into a binary expression.**
4. **Compress the samples to reduce bandwidth, optional step.**

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Digitizing speech was a project first undertaken by the Bell System in the 1950s. The original purpose of digitizing speech was to deploy more voice circuits with a smaller number of wires. This evolved into the T1 and E1 transmission methods of today.

To convert an analog signal to a digital signal, you must perform these steps:

Analog-to-Digital Signal Conversion

Step	Procedure	Description
1.	Sample the analog signal regularly.	The sampling rate must be twice the highest frequency to produce playback that appears neither choppy nor too smooth.
2.	Quantize the sample.	Quantization consists of a scale made up of eight major divisions or chords. Each chord is subdivided into 16 equally spaced steps. The chords are not equally spaced but are actually finest near the origin. Steps are equal within the chords but different when they are compared between the chords. Finer graduations at the origin result in less distortion for low-level tones.
3.	Encode the value into 8-bit digital form.	PBX output is a continuous analog voice waveform. T1 digital voice is a snapshot of the wave encoded in ones and zeros.
4.	(Optional) Compress the samples to reduce bandwidth.	Although not essential to convert analog signals to digital, signal compression is widely used to reduce bandwidth.

The three components in the analog-to-digital conversion process are further described as follows:

- **Sampling:** Sample the analog signal at periodic intervals. The output of sampling is a pulse amplitude modulation (PAM) signal.
- **Quantization:** Match the PAM signal to a segmented scale. This scale measures the amplitude (height) of the PAM signal and assigns an integer number to define that amplitude.
- **Encoding:** Convert the integer base-10 number to a binary number. The output of encoding is a binary expression in which each bit is either a 1 (pulse) or a 0 (no pulse).

This three-step process is repeated 8000 times per second for telephone voice-channel service. Use the fourth optional step—compression—to save bandwidth. This optional step allows a single channel to carry more voice calls.

Note The most commonly used method of converting analog to digital is pulse code modulation (PCM).

Basic Voice Encoding: Converting Digital to Analog

This topic describes the process of converting digital signals back to analog signals.

**Basic Voice Encoding:
Converting Digital to Analog**

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1. **Decompress the samples, if compressed.**
2. **Decode the samples into voltage amplitudes, rebuilding the PAM signal.**
3. **Filter the signal to remove any noise.**

CVOICE v4.2—2-4

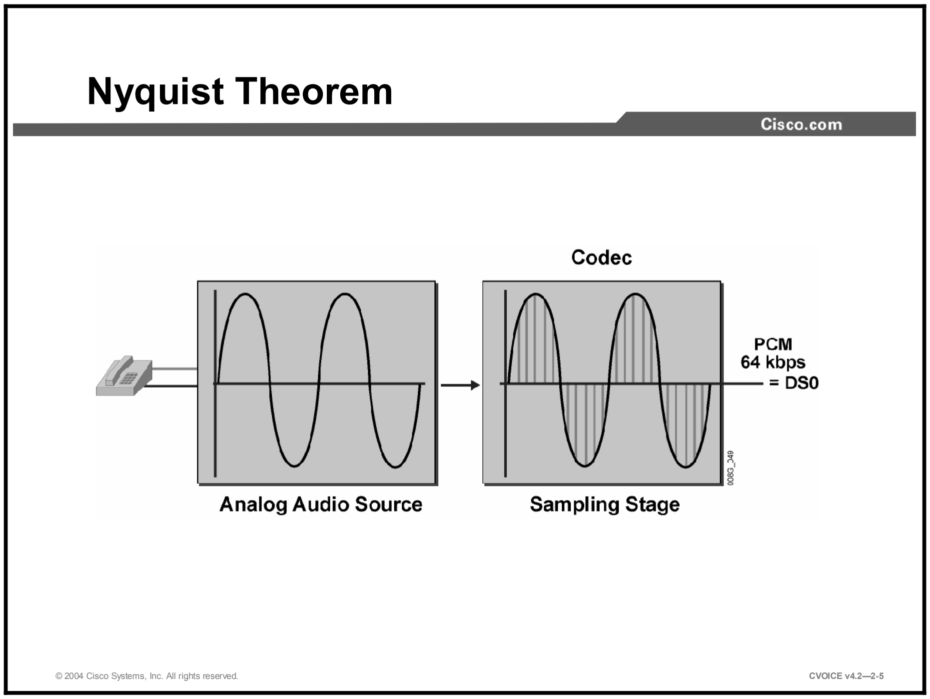
After the receiving terminal at the far end receives the digital PCM signal, it must convert the PCM signal back into an analog signal.

The process of converting digital signals back into analog signals includes the following two steps:

- **Decoding:** The received 8-bit word is decoded to recover the number that defines the amplitude of that sample. This information is used to rebuild a PAM signal of the original amplitude. This process is simply the reverse of the analog-to-digital conversion.
- **Filtering:** The PAM signal is passed through a properly designed filter that reconstructs the original analog wave form from its digitally coded counterpart.

The Nyquist Theorem

This topic describes the Nyquist Theorem, which is the basis for digital signal technology.



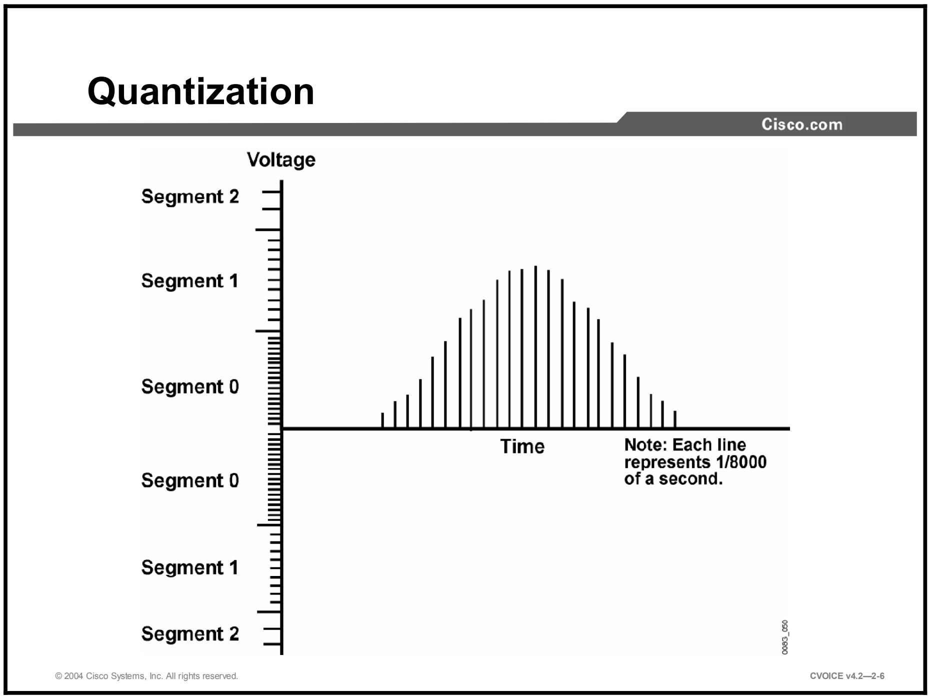
Digital signal technology is based on the premise stated in the Nyquist Theorem: when a signal is instantaneously sampled at the transmitter in regular intervals and has a rate of at least twice the highest channel frequency, then the samples will contain sufficient information to allow an accurate reconstruction of the signal at the receiver.

Example: Nyquist Theorem

While the human ear can sense sounds from 20 to 20,000 Hz, and speech encompasses sounds from about 200 to 9000 Hz, the telephone channel was designed to operate at about 300 to 3400 Hz. This economical range carries enough fidelity to allow callers to identify the party at the far end and sense their mood. Nyquist decided to extend the digitization to 4000 Hz, to capture higher-frequency sounds that the telephone channel may deliver. Therefore, the highest frequency for voice is 4000 Hz, or 8000 samples per second; that is, one sample every 125 microseconds.

Quantization

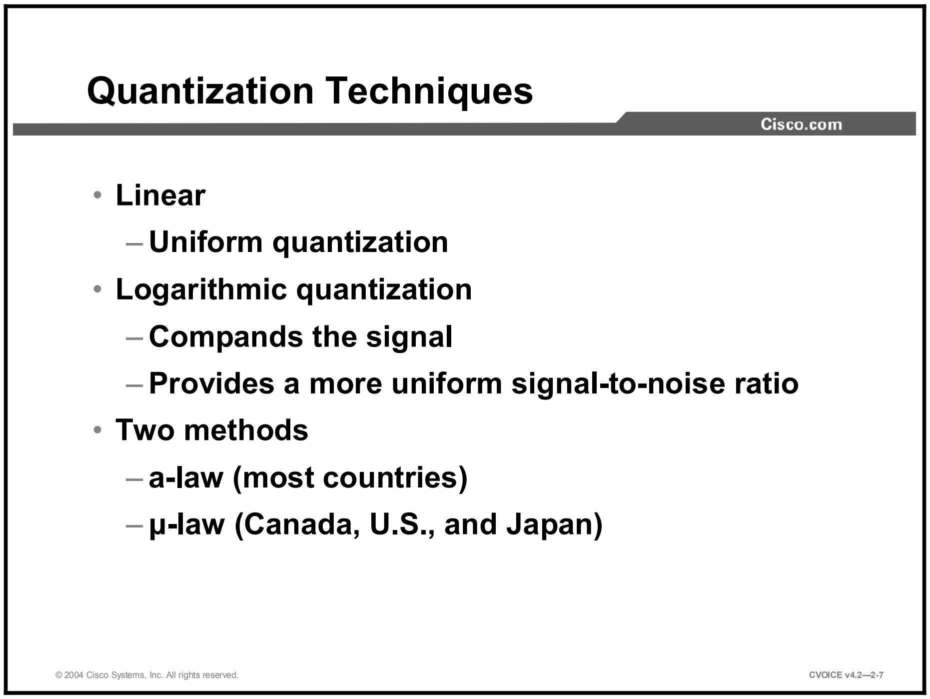
This topic explains quantization and its techniques.



Quantization involves dividing the range of amplitude values that are present in an analog signal sample into a set of discrete steps that are closest in value to the original analog signal. Each step is assigned a unique digital code word.

The figure here depicts quantization. In this example, the x-axis is time and the y-axis is the voltage value (PAM).

The voltage range is divided into 16 segments (0 to 7 positive, and 0 to 7 negative). Starting with segment 0, each segment has fewer steps than the previous segment, which reduces the signal-to-noise ratio (SNR) and makes the segment uniform. This segmentation also corresponds closely to the logarithmic behavior of the human ear. If there is an SNR problem, it is resolved by using a logarithmic scale to convert PAM to PCM.



Linear sampling of analog signals causes small-amplitude signals to have a lower SNR, and therefore poorer quality, than larger amplitude signals. The Bell System developed the *µ-law* method of quantization, which is widely used in North America. The International Telecommunication Union (ITU) modified the original µ-law method and created *a-law*, which is used in countries outside of North America.

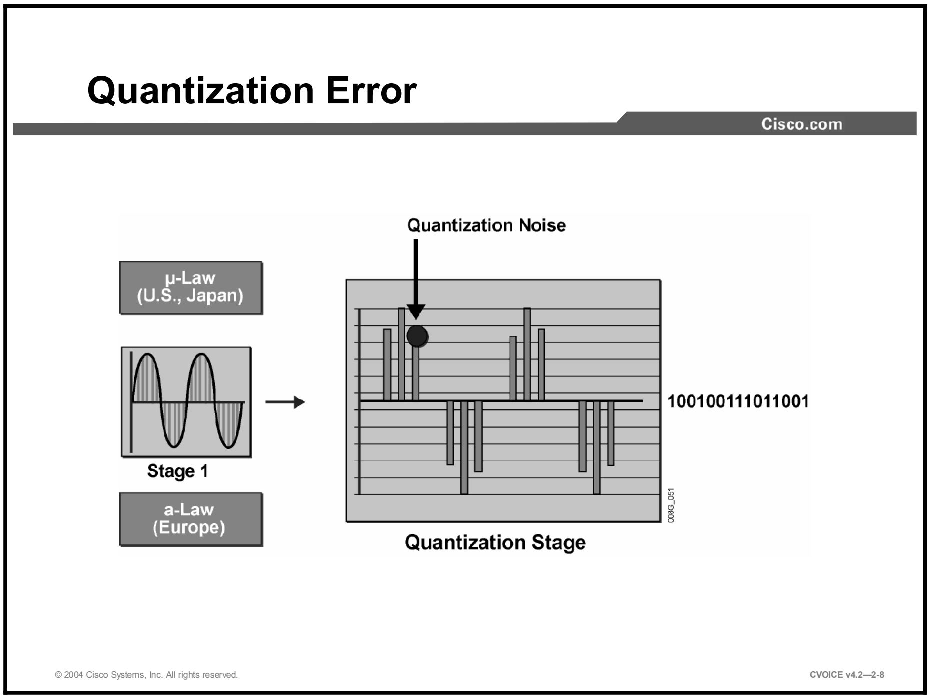
By allowing smaller step functions at lower amplitudes—rather than higher amplitudes—µ-law and a-law provide a method of reducing this problem. Both µ-law and a-law "compand" the signal; that is, they both compress the signal for transmission and then expand the signal back to its original form at the other end.

To calculate the bit rate of digital voice, you must use this formula:

- 2 * 4 kHz * 8 bits per sample = 64,000 bits per second (64 kbps). 64 kbps is a digital signal level 0 (DS-0) rate.

The result of using µ-law and a-law is a more accurate value for smaller amplitude and uniform signal-to-noise quantization ratio (SQR) across the input range.

Note	For communication between a µ-law country and an a-law country, the µ-law country must change its signaling to accommodate the a-law country.



Both µ-law and a-law are linear approximations of a logarithmic input/output relationship. They both generate 64-kbps bit streams using 8-bit code words to segment and quantize levels within segments.

The difference between the original analog signal and the quantization level assigned is called quantization error, which is the source of distortion in digital transmission systems. Quantization error is any random disturbance or signal that interferes with the quality of the transmission or the signal itself.

Example: Quantization Noise

Due to the quantization error, the recreated signal at the receiving end will experience quantization noise. The quantization noise is mostly insignificant, since a single sample represents only 1/8000th of a second. However, frequent quantization errors will cause perceptible quantization noise. For this reason, the recreated signal at the receiving end is sent through a low-pass filter, which filters out the noise.

Voice Compression and Codec Standards

This topic describes two types of voice-compression schemes: waveform coding and source coding.

Voice Compression Techniques

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- Waveform algorithms
  - PCM
  - ADPCM
- Source algorithms
  - LDCELP
  - CS-ACELP

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The following describes the two voice compression techniques:

- **Waveform algorithms (coders):** Waveform algorithms have the following functions and characteristics:
  - Sample analog signals at 8000 times per second
  - Use predictive differential methods to reduce bandwidth
  - Highly impact voice quality because of reduced bandwith
  - Do not take advantage of speech characteristics
- **Source algorithms (coders):** Source algorithms have the following functions and characteristics:
  - Source algorithm coders are called vocoders, or voice coders. A vocoder is a device that converts analog speech into digital speech, using a specific compression scheme that is optimized for coding human speech.
  - Vocoders take advantage of speech characteristics.
  - Bandwidth reduction occurs by sending linear-filter settings.
  - Codebooks store specific predictive waveshapes of human speech. They match the speech, encode the phrases, decode the waveshapes at the receiver by looking up the coded phrase, and match it to the stored waveshape in the receiver codebook.

Example: Waveform Compression

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- **PCM**
  - **Waveform coding scheme**
- **ADPCM**
  - **Waveform coding scheme**
  - **Adaptive: automatic companding**
  - **Differential: encode changes between samples only**
- **ITU standards:**
  - **G.711 rate: 64 kbps = (2 * 4 kHz) * 8 bits/sample**
  - **G.726 rate: 32 kbps = (2 * 4 kHz) * 4 bits/sample**
  - **G.726 rate: 24 kbps = (2 * 4 kHz) * 3 bits/sample**
  - **G.726 rate: 16 kbps = (2 * 4 kHz) * 2 bits/sample**

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Standard PCM is known as ITU standard G.711.

Adaptive differential pulse code modulation (ADPCM) coders, like other waveform coders, encode analog voice signals into digital signals to adaptively predict future encodings by looking at the immediate past. The adaptive feature of ADPCM reduces the number of bits per second that the PCM method requires to encode voice signals.

ADPCM does this by taking 8000 samples per second of the analog voice signal and turning them into a linear PCM sample. ADPCM then calculates the predicted value of the next sample, based on the immediate past sample, and encodes the difference. The ADPCM process generates 4-bit words, thereby generating 16 specific bit patterns.

The ADPCM algorithm from the Consultative Committee for International Telegraph and Telephone (CCITT) transmits all 16 possible bit patterns. The ADPCM algorithm from the American National Standards Institute (ANSI) uses 15 of the 16 possible bit patterns. The ANSI ADPCM algorithm does not generate a 0000 pattern.

The ITU standards for compression are as follows:

- **G.711 rate:** 64 kbps = (2 * 4 kHz) * 8 bits/sample
- **G.726 rate:** 32 kbps = (2 * 4 kHz) * 4 bits/sample
- **G.726 rate:** 24 kbps = (2 * 4 kHz) * 3 bits/sample
- **G.726 rate:** 16 kbps = (2 * 4 kHz) * 2 bits/sample

Note CCITT is now called International Telecommunication Union Telecommunication Standardization Sector (ITU-T).

Example: Source Compression

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- **CELP**
  - **Hybrid coding scheme**
  - **High-quality voice at low bit rates, processor intensive**
- **G.728: LDCELP—16 kbps**
- **G.729: CS-ACELP—8 kbps**
  - **G.729 Annex A variant—8 kbps, less processor intensive, allows more voice channels encoded per DSP**
  - **Annex B variant—VAD and CNG**

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Code excited linear prediction (CELP) compression transforms analog voice signals as follows:

- The input to the coder is converted from an 8-bit to a 16-bit linear PCM sample.
- A codebook uses feedback to continuously learn and predict the voice waveform.
- The coder is excited by a white noise generator.
- The mathematical result (recipe) is sent to the far-end decoder for synthesis and generation of the voice waveform.

Low-delay CELP (LDCELP) is similar to Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP), except for the following:

- LDCELP uses a smaller codebook and operates at 16 kbps to minimize delay—or look-ahead—to 2 to 5 ms.
- The 10-bit codeword is produced from every five speech samples from the 8-kHz input with no look-ahead.
- Four of these 10-bit codewords are called a subframe; they take approximately 2.5 ms to encode. CS-ACELP uses eight 10-bit codewords.

Two of these subframes are combined into a 5-ms block for transmission. CS-ACELP is a variation of CELP that performs these functions:

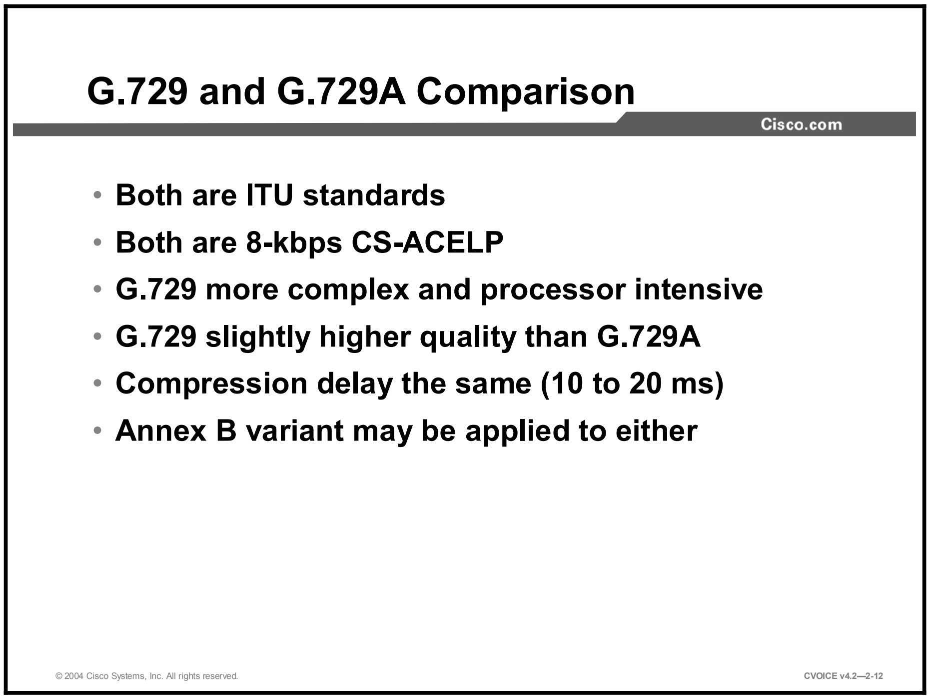
- Codes 80-byte frames, which take approximately 10 ms to buffer and process.
- Adds a look-ahead of 5 ms. A look-ahead is a coding mechanism that continuously analyzes, learns, and predicts the next waveshape.
- Adds noise reduction and pitch-synthesis filtering to processing requirements.

Example: G.729 Annex B

The G.729 Annex B (G.729B) variant adds voice activity detection (VAD) in strict compliance with G.729B standards. When this coder-decoder (codec) variant is used, VAD is not tunable for music threshold. However, when Cisco VAD is configured, music threshold is tunable.

G.729 and G.729A Compared

This topic compares G.729 and G.729 Annex A (G.729A) compression.



G.729, G.729 Annex A (G.729A), G.729 Annex B (G.729B), and G.729A Annex B (G.729AB) are variations of CS-ACELP.

There is little difference between the ITU recommendations for G.729 and G.729A. All of the platforms that support G.729 also support G.729A.

G.729 is the compression algorithm that Cisco uses for high-quality 8-kbps voice. When properly implemented, G.729 sounds as good as the 32-kbps ADPCM. G.729 is a high-complexity, processor-intensive compression algorithm that monopolizes processing resources.

Although G.729A is also an 8-kbps compression, it is not as processor-intensive as G.729. It is a medium-complexity variant of G.729 with slightly lower voice quality. G.729A is not as high-quality as G.729 and is more susceptible to network irregularities, such as delay, variation, and tandeming. Tandeming causes distortion that occurs when speech is coded, decoded, and then coded and decoded again, much like the distortion that occurs when a videotape is repeatedly copied.

Example: Codec Complexity

On Cisco IOS gateways, you must use the variant (G.729 or G.729A) that is related to the codec complexity configuration on the voice card. This variant does not show up explicitly in the Cisco IOS command-line interface (CLI) codec choice. For example, the CLI does not display **g729r8** (alpha code) as a codec option. However, if the voice card is defined as medium-complexity, then the **g729r8** option is the G.729A codec.

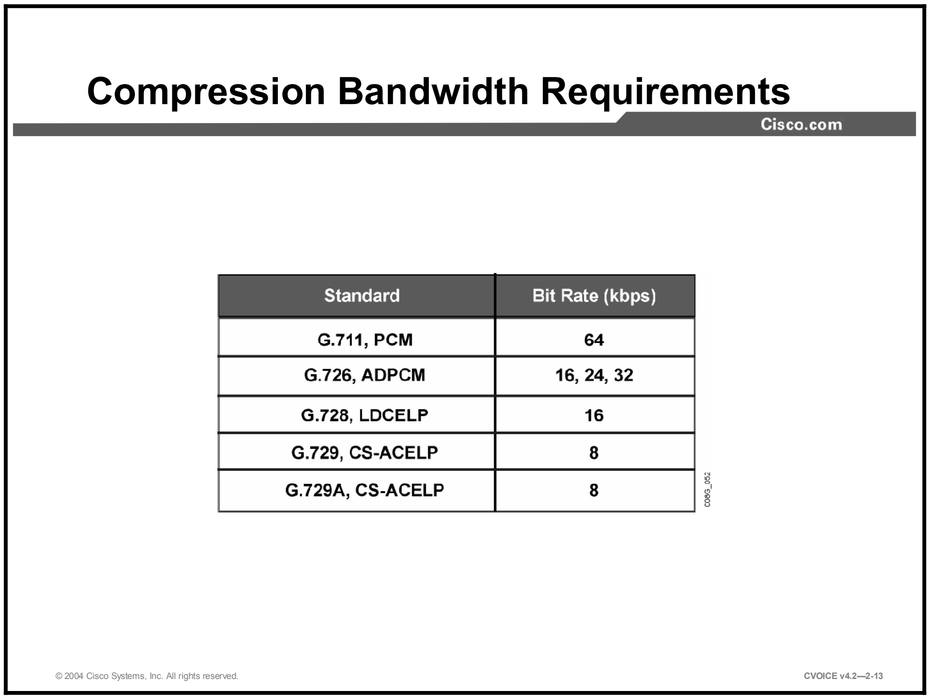
G.729B is a high-complexity algorithm, and G.729AB is a medium-complexity variant of G.729B with slightly lower voice quality. The difference between the G.729 and G.729B codec is that the G.729B codec provides built-in Internet Engineering Task Force (IETF) VAD and comfort noise generation (CNG).

The following G.729 codec combinations interoperate:

- G.729 and G.729A
- G.729 and G.729
- G.729A and G.729A
- G.729B and G.729AB
- G.729B and G.729B
- G.729AB and G.729AB

Compression Bandwidth Requirements

This topic lists the bandwidth requirements for various ITU compression standards.



The following three common voice compression techniques are standardized by the ITU-T:

- **PCM:** Amplitude of voice signal is sampled and quantized at 8000 times per second. Each sample is then represented by one octet (8 bits) and transmitted. For sampling, you must use either a-law or µ-law to reduce the signal-to-noise ratio.
- **ADPCM:** The difference between the current sample and its predicted value (based on past samples). ADPCM is represented by 2, 3, 4, or 5 bits. This method reduces the bandwidth requirement at the expense of signal quality.
- **CELP:** Excitation value and a set of linear-predictive filters (settings) are transmitted. The filter setting transmissions are less frequent than excitation values and are sent on an as-needed basis.

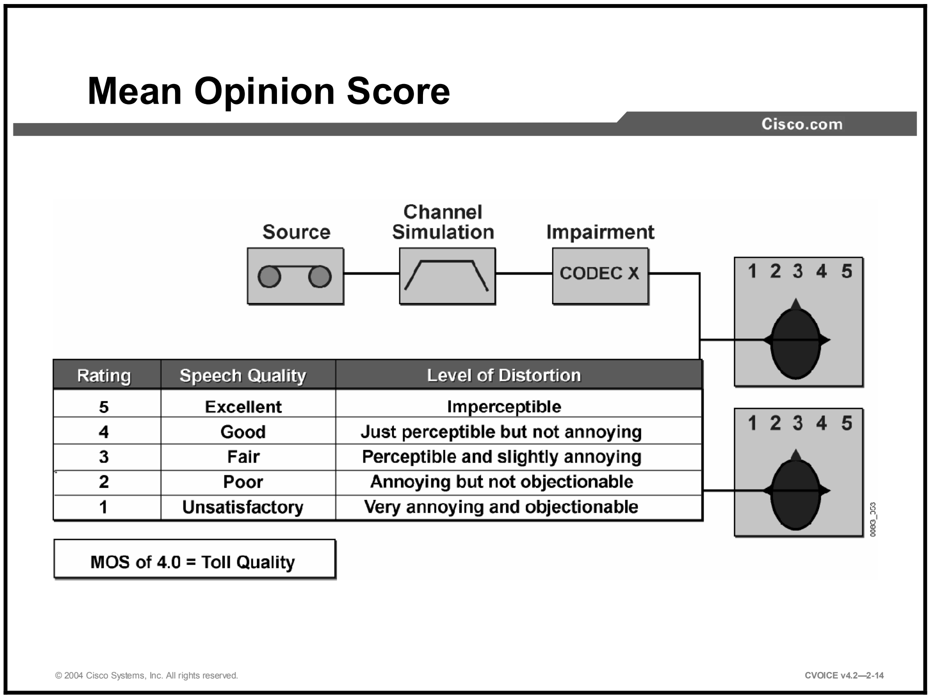
The table describes the codecs and compression standards:

Codecs and Compression Standards

Codec	Compression Technique	Bit Rate (kbps)
G.711	PCM	64
G.726	ADPCM	16, 24, 32
G.728	LDCELP	16
G.729	CS-ACELP	8
G.729A	CS-ACELP	8

Voice Quality Measurement

This topic describes two methods that are used to subjectively measure the quality of voice transported on a telephone line. Because different compression schemes produce different quality results, a method of comparing them is necessary.

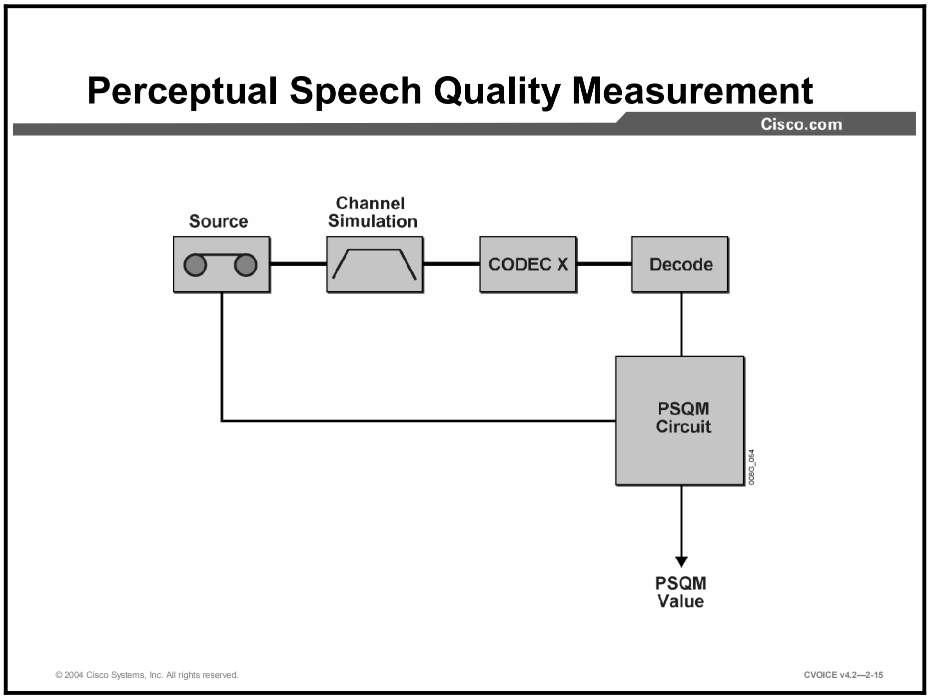


The figure depicts mean opinion score (MOS). MOS is a system of grading the voice quality of telephone connections. The MOS is a statistical measurement of voice quality derived from the judgments of several subscribers.

Graded by humans and very subjective, the range of MOS is 1 to 5, where 5 is direct conversation.

Voice Quality of Telephone Connections

Rating	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying and objectionable



A newer, more objective measurement is available that is quickly overtaking MOS scores as the industry quality measurement of choice for coding algorithms. Perceptual Speech Quality Measurement (PSQM), as per ITU standard P.861, provides a rating on a scale of 0 to 6.5, where 0 is best and 6.5 is worst.

PSQM is implemented in test equipment and monitoring systems that are available from vendors other than Cisco. Some PSQM test equipment converts the 0-to-6.5 scale to a 0-to-5 scale to correlate to MOS. PSQM works by comparing the transmitted speech to the original input and yields a score. Various vendor test equipment is now capable of providing a PSQM score for a test voice call over a particular packet network.

In 1998, British Telecom developed a predictive voice quality measurement algorithm called Perceptual Analysis Measurement System (PAMS). PAMS can predict subjective speech quality measurement methods, such as MOS, when fidelity is affected by such things as waveform codecs, vocoders, and various speaker dependencies, such as language. PAMS, unlike PSQM, includes automatic normalization for levels.

ITU standard P.862 supercedes P.861 and describes a voice quality measurement technique that combines PSQM and PAMS. Originally developed by KPN Research, the Netherlands, and British Telecommunications (BT), Perceptual Evaluation of Speech Quality (PESQ) is an objective measuring tool that can “predict” results of subjective measuring tests, such as MOS. PESQ can be found in test equipment from a variety of vendors.

Example: Measuring Voice Quality

Cisco voice equipment does not perform voice quality measurements. There are a number of vendors who offer voice quality measurement products, some of which are designed to work with Cisco CallManager and CiscoWorks.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- **The three parts of the analog-to-digital conversion process are sampling, quantization, and encoding.**
- **The two parts of the digital-to-analog conversion process are decoding and filtering.**
- **Digital signal technology is based on the Nyquist Theorem.**
- **Quantization involves dividing the range of amplitude values of an analog signal sample.**
- **The two techniques used for voice compression are waveform compression and source compression.**

CVOICE v4.2—2-16

Summary (Cont.)

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- **G.729 and G.729A compression algorithms are similar variations of CS-ACELP.**
- **The three common voice compression standards are PCM, ADPCM, and CELP.**
- **The two types of voice quality measurement are MOS and PSQM.**

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References

For additional information, refer to this resource:

- McQuerry, S., McGrew, K., Foy, S. *Cisco Voice over Frame Relay, ATM, and IP.* Indianapolis, Indiana: Cisco Press; 2001.

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 2-1: Lab Familiarity

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which of these steps is optional in analog-to-digital conversion?

A) compression

B) encoding

C) quantization

D) sampling

Q2) In telephone voice-channel service, how often is the analog-to-digital signal conversion sampling process repeated?

A) 8000 times per call

B) 8000 times per voice sample

C) 8000 times per second

D) 8000 times per minute

Q3) What device is used to reconstruct the digital signal to its original analog signal?

A) modem

B) compander

C) filter

D) codec

Q4) What is recovered from each 8-bit word when digital signals are converted back to analog signals?

A) the PAM signal

B) the number that defines the amplitude of the sample

C) the analog waveform

D) the base-10 number that corresponds to the 8-bit number

Q5) What is the frequency range of a telephone channel?

A) 20 to 20,000 Hz

B) 200 to 9000 Hz

C) 300 to 3400 Hz

D) 4000 to 8000 Hz

Q6) According to the Nyquist Theorem, what is the highest sound frequency possible when sampling once every 125 microseconds?

A) 3400 Hz

B) 4000 Hz

C) 8000 Hz

D) 20,000 Hz

Q7) What is the problem with linear sampling of analog signals?

A) Small-amplitude signals have a higher signal-to-noise ratio.

B) The signal cannot be sampled instantaneously at the transmitter.

C) The sampling interval is too long for high-amplitude signals.

D) The samples do not contain enough information for accurate conversion.

Q8) Which quantization method is used in the United States?

A) a-law

B) μ-law

C) PCM

D) Nyquist

Q9) Which two coding schemes are examples of waveform algorithms? (Choose two.)

A) PCM

B) ADPCM

C) CELP

D) LDCELP

E) CS-ACELP

Q10) Which two are functions of a source algorithm? (Choose two.)

A) samples analog signals at 8000 times per second

B) uses predictive differential method to reduce bandwidth

C) uses codebooks to match stored waveshapes in the receiver codebook

D) takes advantage of speech characteristics

Q11) What are two differences between G.729 and G.729A compression? (Choose two.)

A) the platforms that support them

B) complexity

C) compression delay

D) originating standards body

E) demands on the processor

Q12) Which two G.729 codecs provide built-in IETF VAD and CNG? (Choose two.)

A) G.729

B) G.729A

C) G.729B

D) G.729AB

Q13) Which compression technique is used with G.711 codecs?

A) ADPCM

B) CS-ACELP

C) LDCELP

D) PCM

Q14) Which compression standard has the lowest bandwidth requirement?

A) ADPCM

B) CS-ACELP

C) LDCELP

D) PCM

Q15) What is the level of distortion for a MOS rating of 4?

A) imperceptible

B) just perceptible but not annoying

C) perceptible and slightly annoying

D) annoying but not objectionable

Q16) What is the rating scale for measuring voice quality with PSQM?

A) 1 to 5 where 1 is worst

B) 1 to 5 where 5 is worst

C) 0 to 6.5 where 0 is worst

D) 0 to 6.5 where 6.5 is worst

Quiz Answer Key

Q1) A

Relates to: Basic Voice Encoding: Converting Analog to Digital

Q2) C

Relates to: Basic Voice Encoding: Converting Analog to Digital

Q3) D

Relates to: Basic Voice Encoding: Converting Digital to Analog

Q4) B

Relates to: Basic Voice Encoding: Converting Digital to Analog

Q5) C

Relates to: The Nyquist Theorem

Q6) B

Relates to: The Nyquist Theorem

Q7) A

Relates to: Quantization

Q8) B

Relates to: Quantization

Q9) A, B

Relates to: Voice Compression and Codec Standards

Q10) C and D

Relates to: Voice Compression and Codec Standards

Q11) B, E

Relates to: G.729 and G.729A Compared

Q12) C, D

Relates to: G.729 and G.729A Compared

Q13) D

Relates to: Compression Bandwidth Requirements

Q14) B

Relates to: Compression Bandwidth Requirements

Q15) B

Relates to: Voice Quality Measurement

Q16) D

Relates to: Voice Quality Measurement

Understanding Signaling Systems

Overview

This lesson describes the various signaling systems used between telephony systems. It also explores signaling between PBXs, signaling between PBXs and COs, and specialized signaling, such as ISDN.

Relevance

Configuring Cisco Systems voice equipment to interface with other equipment requires an understanding of the signaling that conveys supervision between the systems. Proper troubleshooting also requires an understanding of these signaling systems.

Objectives

Upon completing this lesson, you will be able to describe the appropriate signaling method to deploy in a telephony system. This includes being able to meet these objectives:

- State the uses and types of CAS systems used for T1
- State the uses and types of CAS systems used for E1
- State the uses and types of CCS systems
- Name the benefits of using ISDN to convey voice
- Name the benefits of using QSIG to convey voice
- Name the functions of DPNSS signaling and its benefits
- Name the functions of SIGTRAN signaling and its benefits
- Name the functions of SS7 and its benefits
- Describe how separate signaling systems can be interfaced

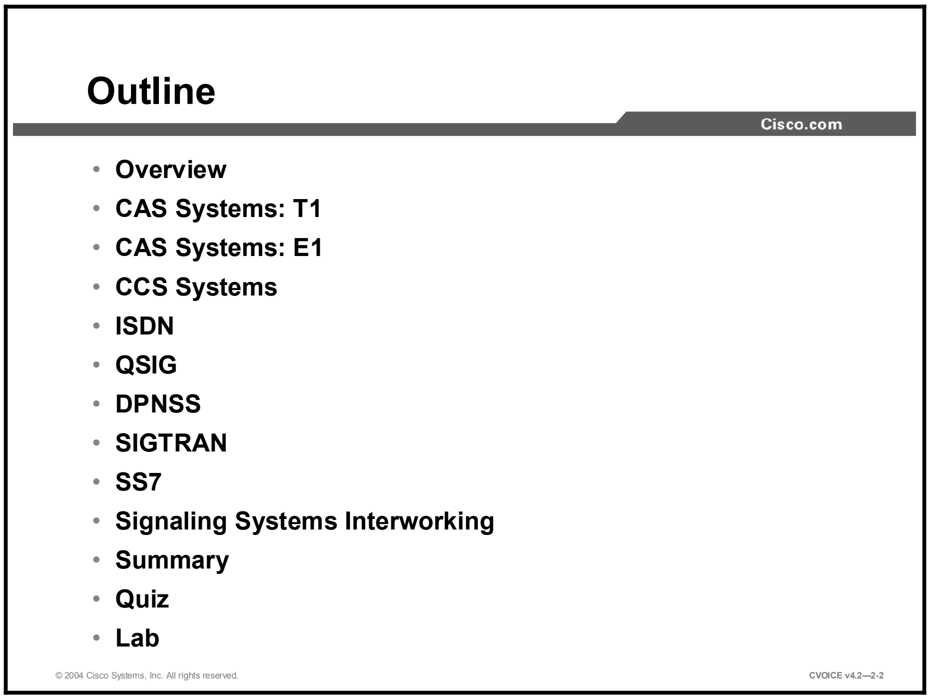
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with PBX systems
- Understanding of the need for signaling between systems
- Understanding of address, supervisory, and call-progress signals

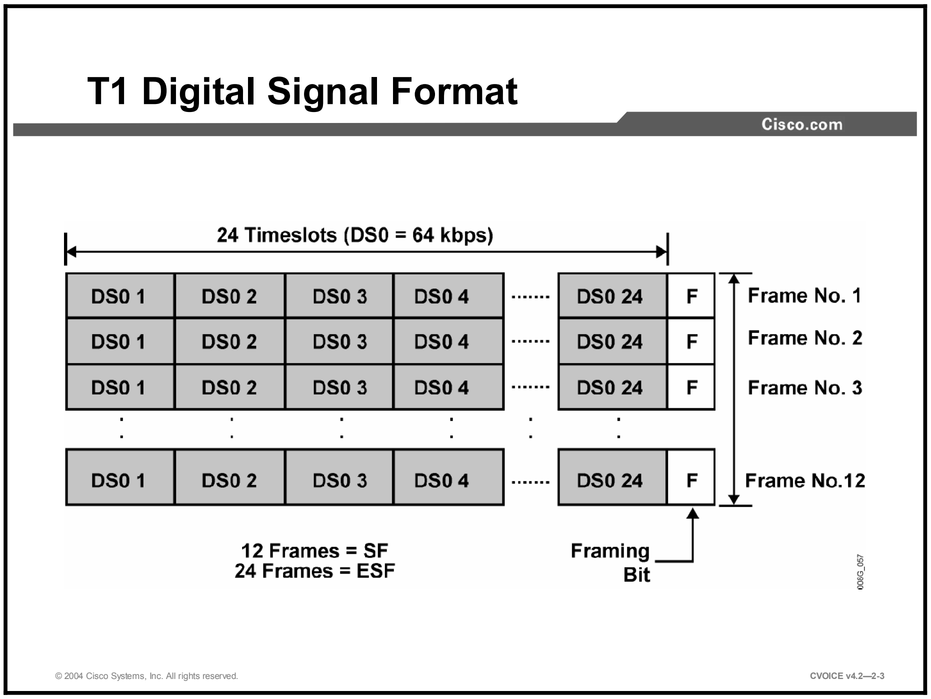
Outline

The outline lists the topics included in this lesson.



CAS Systems: T1

This topic describes channel associated signaling (CAS) and its uses with T1 transmission.

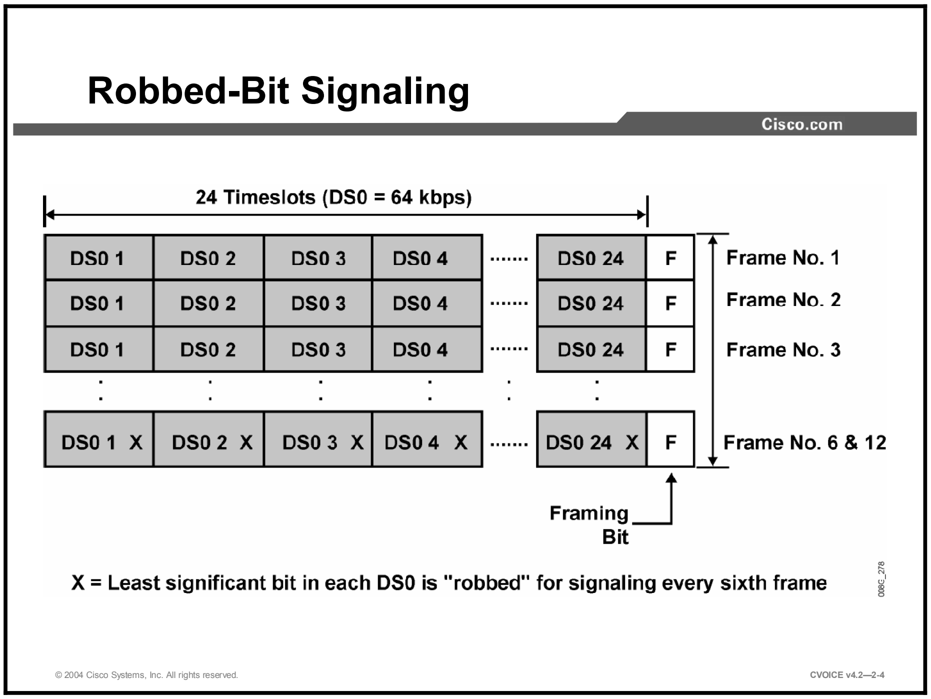


CAS is a signaling method commonly used between PBXs. Although this can manifest itself in many forms, some methods are more common than others. Signaling systems can also be implemented between a PBX and a Cisco voice device.

PBXs and Cisco devices use T1 and E1 to convey voice. Originally, this was the main purpose of T1, which carries signaling information using two methodologies, CAS and common channel signaling (CCS). The figure illustrates the format of the T1 digital signal.

The characteristics of the T1 digital signal format are as follows:

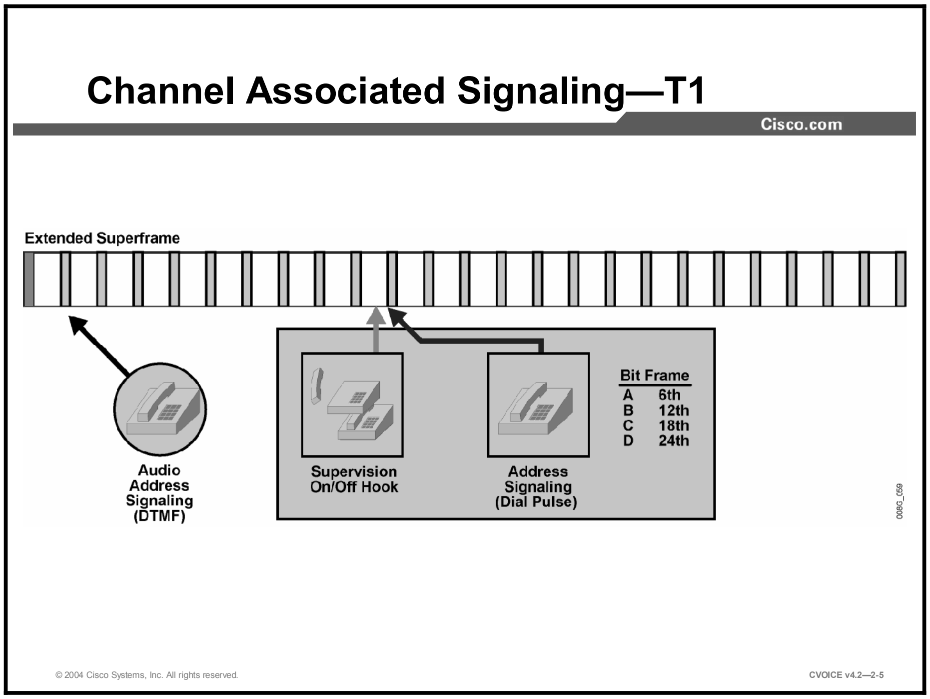
- A T1 frame is 193 bits long—8 bits from each of the 24 timeslots (digital service zeros [DS0s]) plus 1 bit for framing. A T1 repeats every 125 microseconds, resulting in 8000 samples per second (8 bits * 24 timeslots + 1 framing bit * 8000 samples/second = 1.544 Mbps).
- T1 has two major framing and/or format standards:
  - Super Frame (SF), or D4, specifies 12 frames in sequence. The D4 framing pattern used in the F position in the figure is 100011011100 (a 1 goes with the first frame, a 0 goes with the second frame, a 0 goes with the third frame, and so on all the way through 12 frames). This unique framing pattern allows the receiving T1 equipment to synchronize within four frames, since any four consecutive frame bits are unique within the 12-bit pattern. Because there are 8000 T1 frames transmitted per second, 8000 F bits are produced and used for framing.
  - Extended Superframe (ESF) format was developed as an upgrade to SF and is now dominant in public and private networks. Both types of format retain the basic frame structure of one framing bit followed by 192 data bits. However, ESF repurposes the use of the F bit. In ESF, of the total 8000 F bits used in T1, 2000 are used for framing, 2000 are used for cyclic redundancy check (CRC) (for error checking only), and 4000 are used as an intelligent supervisory channel to control functions end to end (such as loopback and error reporting).



Because each DS0 channel carries 64 kbps, and G.711 is 64 kbps, there is no room to carry signaling. Implemented for voice, the T1 uses every sixth frame to convey signaling information. In every sixth frame, the least significant bit (LSB) for each of the voice channels is used to convey the signaling. Although this implementation detracts from the overall voice quality (because only seven bits represent a sample for that frame), the impact is not significant. This method is called robbed-bit signaling (RBS). When SF employs this method, the signaling bits are conveyed in both the 6th (called the “A” bit) and 12th (called the “B” bit) frames. For control signaling, A and B bits provide both near- and far-end off-hook indication.

The A and B bits can represent different signaling states or control features (on hook or off hook, idle, busy, ringing, and addressing). The robbed bit is the least significant bit from an 8-bit word.

ESF also uses RBS in frames 6, 12, 18, and 24, which yields ABCD signaling options, providing additional control and signaling information.



Because the signaling occurs within each DS0, it is referred to as in band. Also, because the use of these bits is exclusively reserved for signaling each respective voice channel, it is referred to as CAS.

The robbed bits are used to convey E&M status or FXS/FXO status and provide call supervision for both on hook and off hook.

Example: Channel Associated Signaling

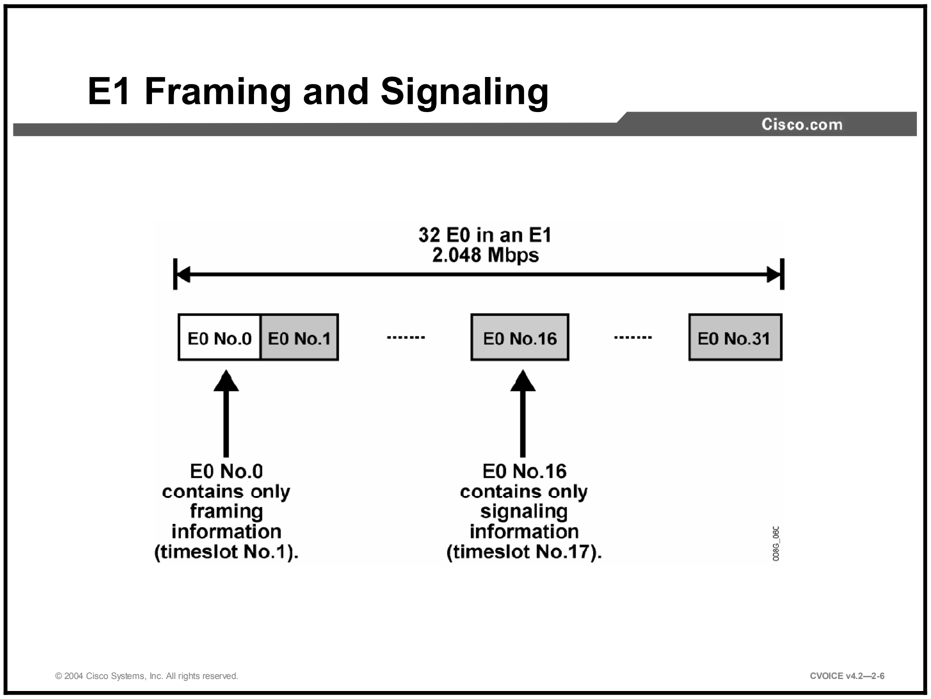
D4 has a 12-frame structure and provides AB bits for signaling.

ESF has a 24-frame structure and provides ABCD bits for signaling.

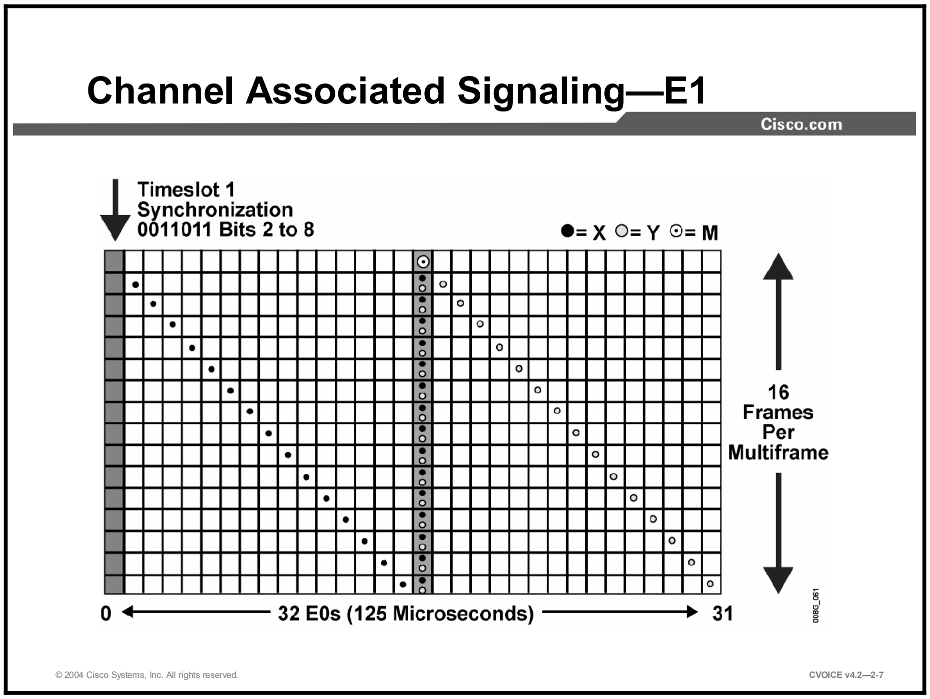
DTMF, or tone, can be carried in band in the audio path; however, other supervisory signals must still be carried via CAS.

CAS Systems: E1

This topic describes CAS and its uses with E1 transmission.



In E1 framing and signaling, 30 of the 32 available channels, or timeslots, are used for voice and data. Framing information uses timeslot 1, while timeslot 17 (E0 16) is used for signaling by all the other timeslots. This signaling format is also known as CAS because the use of the bits in the 17th timeslot is exclusively reserved for the purpose of signaling each respective channel. However, this implementation of CAS is considered out of band because the signaling bits are not carried within the context of each respective voice channel, as is the case with T1.



In the E1 frame format, 32 timeslots make up a frame. A multiframe consists of 16 E1 frames, as depicted in the figure.

The timeslots are numbered 1 though 32. Multiframe timeslots are configured as follows:

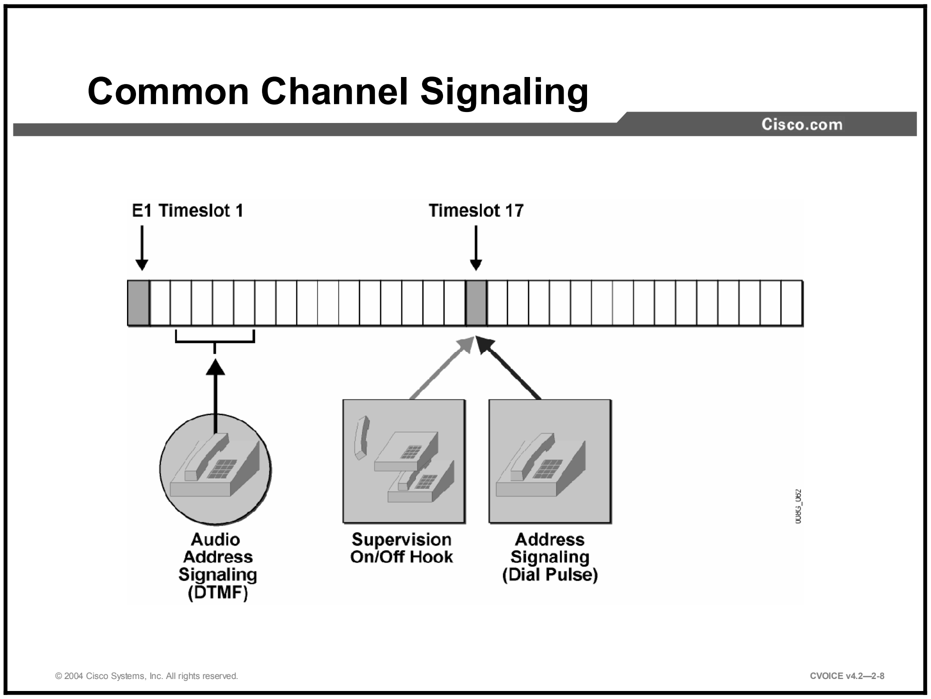
- Timeslot 1 carries only framing information.
- Timeslot 17, in the first frame of the 16-frame multiframe, declares the beginning of the multiframe, which is indicated by the M symbol in the figure.
- The remaining slot 17s carry signaling information for all the other timeslots:
  - Slot 17 of the first frame declares the beginning of a 16-frame multiframe (M).
  - Slot 17 of the second frame carries ABCD for voice slot 2 (X) and ABCD for voice slot 18 (Y).
  - Slot 17 of the third frame carries ABCD for voice slot 3 (X) and ABCD for voice slot 19 (Y).
  - This process continues for all of the remaining frames.

Example: E1 Channel Associated Signaling

E1 CAS is directly compatible with T1 CAS, because both methods use AB or ABCD bit signaling. Although the signaling for E1 CAS is carried in a single common timeslot, it is still referred to as CAS because each individual signaling timeslot represents a specific pair of voice channels.

CCS Systems

This topic describes common channel signaling (CCS) systems.



CCS differs from CAS in that all channels use a common channel and protocol for call setup. Using E1 as an example, a signaling protocol, such as the ISDN Q.931, would be deployed in timeslot 17 to exchange call-setup messages with its attached telephony equipment.

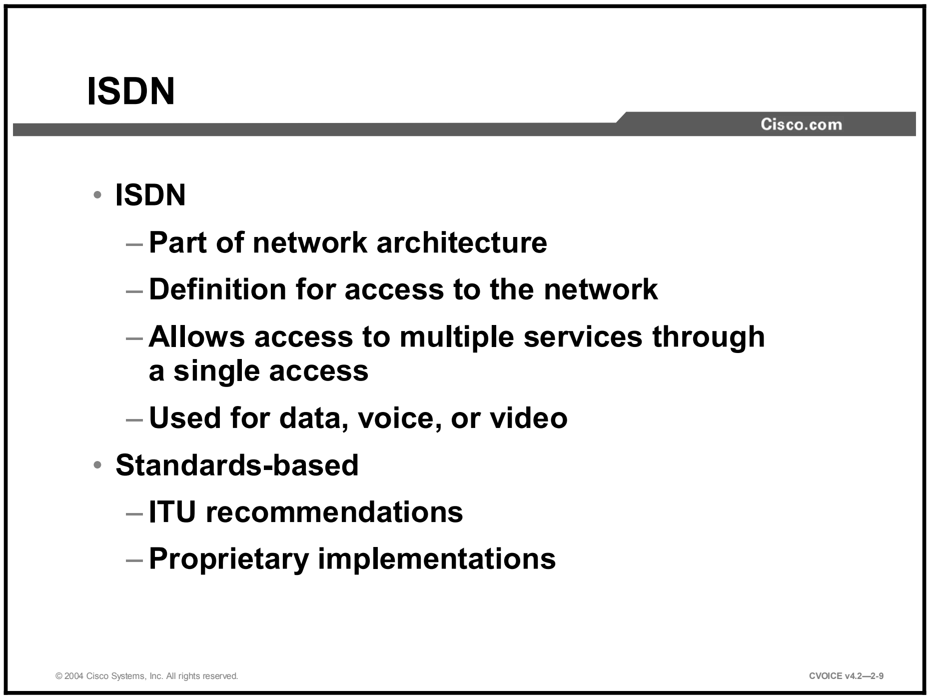
Example: CCS Signaling

Examples of CCS signaling are as follows:

- **Proprietary implementations:** Some PBX vendors choose to use CCS for T1 and E1 and implement a proprietary CCS protocol between their PBXs. In this implementation, Cisco devices are configured for Transparent Common Channel Signaling (T-CCS) because they do not understand proprietary signaling information.
- **ISDN:** Uses Q.931 in a common channel to signal all other channels.
- **Digital Private Network Signaling System (DPNSS):** An open standard developed by British Telecom for implementation by any vendor who chooses to use it. DPNSS also uses a common channel to signal all other channels.
- **Q Signaling (QSIG):** Like ISDN, uses a common channel to signal all other channels.
- **Signaling System 7 (SS7):** An out-of-band network implemented and maintained by various telephone companies and used for signaling and other supplemental services.

ISDN

This topic describes how to implement ISDN as a signaling system to support voice.



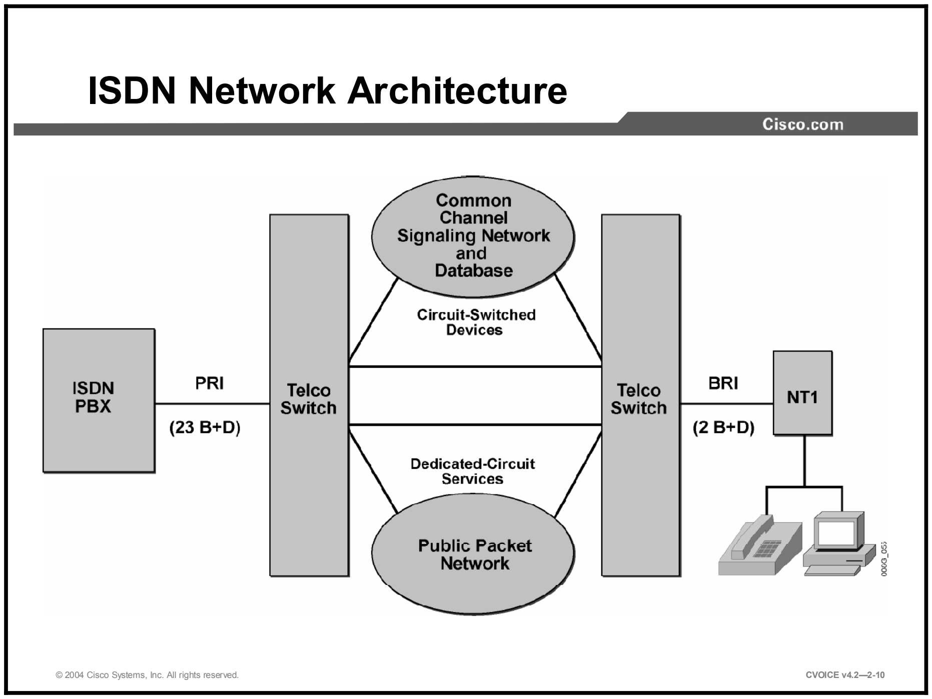
ISDN is an access specification to a network. You may have studied ISDN as an access method for dialup data systems. Because it is a digital system, ISDN makes connections rapidly.

ISDN can be implemented in two different ways: BRI and PRI. BRI features two bearer (B) channels, while PRI supports 23 (for T1) or 30 (for E1) B channels. Each implementation also supports a data (D) channel, used to carry signaling information (CCS).

The following are benefits of using ISDN to convey voice:

- Each B channel is 64 kbps, making it perfect for G.711 PCM.
- ISDN has a built-in call control protocol known as ITU-T Q.931.
- ISDN can convey standards-based voice features, such as call forwarding.
- ISDN supports standards-based enhanced dialup capabilities, such as Group 4 fax and audio channels.

Note	ISDN BRI voice is commonly used in Europe; ISDN PRI voice is used worldwide.

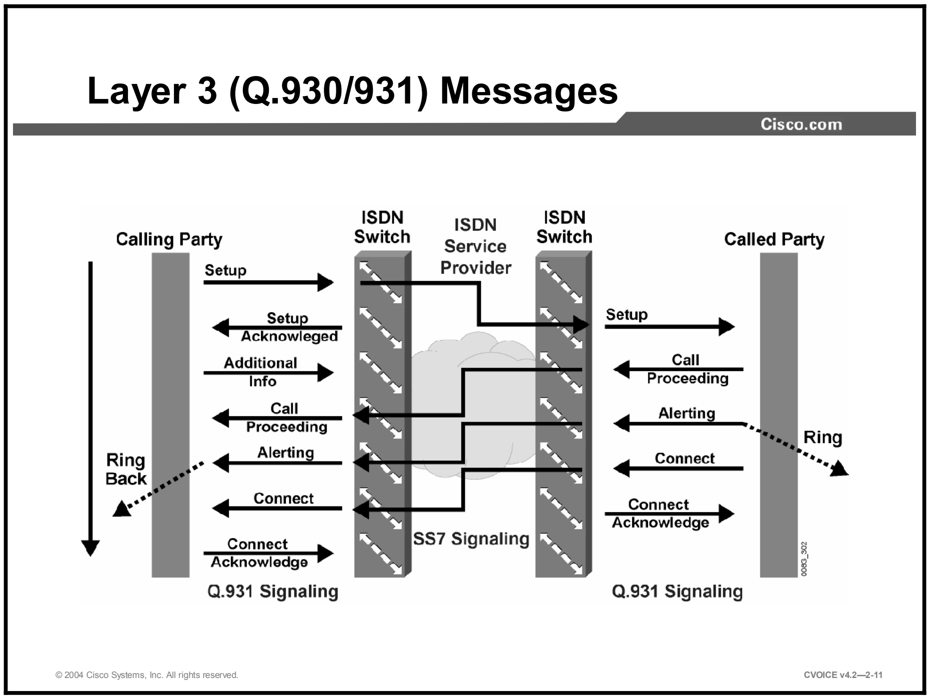


The figure here depicts the architecture of an ISDN network. The B channel carries information, such as voice, data, and video, at 64-kbps DS0.

The D channel carries call signaling between customer premises equipment (CPE) and the network, usually as the Q.931 protocol but sometimes as the QSIG protocol.

BRI operates using the average local copper pair. It uses two B channels and one signaling channel. It is represented as 2 B+D.

PRI implemented on T1 uses 23 B channels and one signaling channel. It is represented as 23 B+D. PRI implemented on E1 uses 30 B channels and one signaling channel. It is represented as 30 B+D.



Layer 3, Q.931, uses a standard set of messages to communicate. These standard commands cover the following areas:

- **Call establishment:** Initially sets up a call. Messages travel between the user and the network. Call establishment events include alerting, call proceeding, connect, connect acknowledgment, progress, setup, and setup acknowledgment.
- **Call information phase:** Data sent between the user and the network after the call is established. This allows the user to, for example, suspend and then resume a call. Events in the call information phase include: hold, hold acknowledgment, hold reject, resume, resume acknowledgment, resume reject, retrieve, retrieve acknowledgment, retrieve reject, suspend, suspend acknowledgment, suspend reject, and user information.
- **Call clearing:** Terminates a call. The following events occur in the call-clearing phase: disconnect, release, release complete, restart, and restart acknowledgment.
- **Miscellaneous messages:** Negotiates network features (supplementary services). Miscellaneous services include congestion control, facility, information, notify, register, status, and status inquiry.

Example: ISDN Messages

ISDN Layer 3 messages, or Q.931, are carried within ISDN Layer 2 frames, called Q.921. Cisco ISDN equipment allows the administrator to monitor these messages as they occur using various **debug** commands.

QSIG

This topic lists the benefits of using QSIG to support voice.

QSIG Protocol

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Layer				
Layers 4-7	Remote Operations Service Element (ROSE) association control service element (ACSE)			End-to-end protocol network transparent
Network	In progress			QSIG procedures for supplementary services
	ISO 11582, ETS300, 239 ECMA 165			QSIG generic functional procedures
	ISO 11574, ETS200 171/172, EDMA 142/143			QSIG basic call
Link Layer	ECMA 141, ETS300 402			Interface-dependent protocols
Physical	Basic rate 1,430	Primary rate 1,431		
Media	Copper	Copper	Optical	

 CVOICE v4.2—2-12

The QSIG protocol, which is based on the ISDN Q.931 standard, provides signaling for private integrated services network exchange (PINX) devices. PINX includes everything from PBXs and multiplexers to Centrex. By using QSIG PRI signaling, a Cisco device can route incoming voice calls from a PINX across a WAN to a peer Cisco device, which can then transport the signaling and voice packets to a second PINX. QSIG is implemented on PRI interfaces only. ISDN PRI QSIG voice-signaling provides the following benefits:

- Connects the Cisco device with digital PBXs that use the QSIG form of CCS
- Provides transparent support for supplementary PBX services so that proprietary PBX features are not lost when connecting PBXs to Cisco networks
- Provides QSIG support based on widely used ISDN Q.931 standards and the European Telecommunication Standards Institute (ETSI) implementation standards, which include the following specifications:
  - **European Computer Manufacturers Association (ECMA)-143:** Private Integrated Services Network (PISN) – Circuit Mode Bearer Services – Inter-Exchange Signalling Procedures and Protocol (QSIG-BC) (This specification covers QSIG basic call services.)
  - **ECMA-142:** Private Integrated Services Network (PISN) – Circuit Mode 64kbit/s Bearer Services – Service Description, Functional Capabilities and Information Flows (BCSD)
  - **ECMA-165:** Private Integrated Services Network (PISN) – Generic Functional Protocol for the Support of Supplementary Services – Inter-Exchange Signalling Procedures and Protocol (QSIG-GF)

DPNSS

This topic describes the DPNSS signaling protocol.

DPNSS

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- **Developed by British Telecom and selected PBX vendors**
- **Operates over standard ISDN physical interfaces**
- **Defined by the following documents:**
  - **BTNR 188: DPNSS**
  - **BTNR 188-T: Testing**
  - **BTNR 189: Interworking With Other Signaling Systems**
  - **BTNR 189-I: Interworking with ISDN**

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British Telecom and selected PBX manufacturers originally developed Digital Private Network Signaling System (DPNSS) in the early 1980s. It was developed and put into use before the ISDN standards were completed because customers wanted to make use of digital facilities as soon as possible.

DPNSS operates over standard ISDN physical interfaces and is described in four documents:

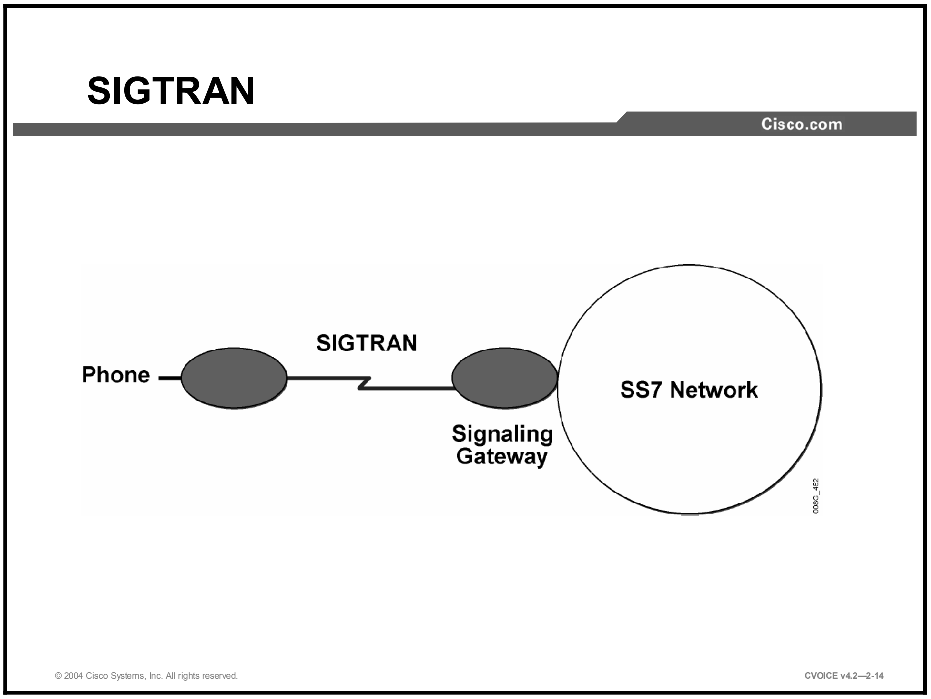
- BTNR 188: *Digital Private Networking Signalling System No 1*, Issue 6, January 1995.
- BTNR 188-T: *Digital Private Networking Signalling System No 1: Testing Schedule*.
- BTNR 189: *Interworking between DPNSS1 and other Signalling Systems*, Issue 3, March 1988.
- BTNR 189-I: *Interworking between DPNSS1 and ISDN Signalling Systems*, Issue 1, December 1992.

Example: Cisco DPNSS Support

Cisco Systems supports DPNSS on various gateway platforms, such as the 2600, 3600, and 5300 series. DPNSS is not a common signaling system but is still in use in various parts of the world.

SIGTRAN

This topic describes the SIGTRAN signaling system.



SIGTRAN is a signaling protocol defined in RFC 2719 and RFC 2960. It describes the way the IP protocol carries SS7 messages in a VoIP network. SIGTRAN relies on the Stream Control Transport Protocol at Layer 4 of the TCP/IP protocol stack.

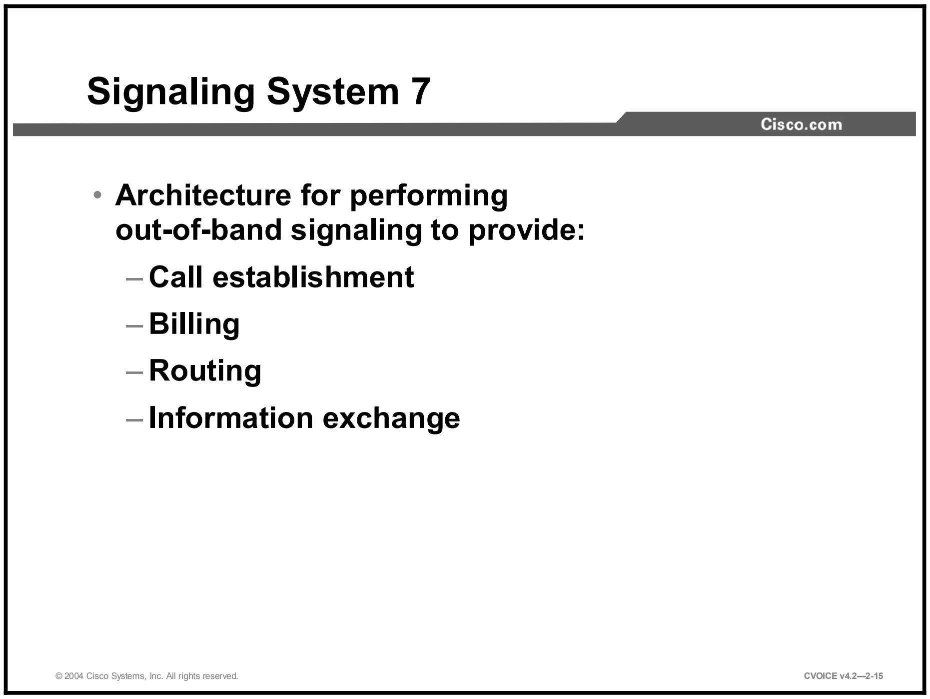
Using SIGTRAN, a service provider may interconnect a private VoIP network to the PSTN and ensure that SS7 signals are conveyed end to end.

Example: SIGTRAN Application

SIGTRAN is implemented on Cisco IP Transfer Point (ITP) equipment as well as the Cisco SC 2200 Signaling Controller.

SS7

This topic lists the primary functions and benefits of Signaling System 7 (SS7).

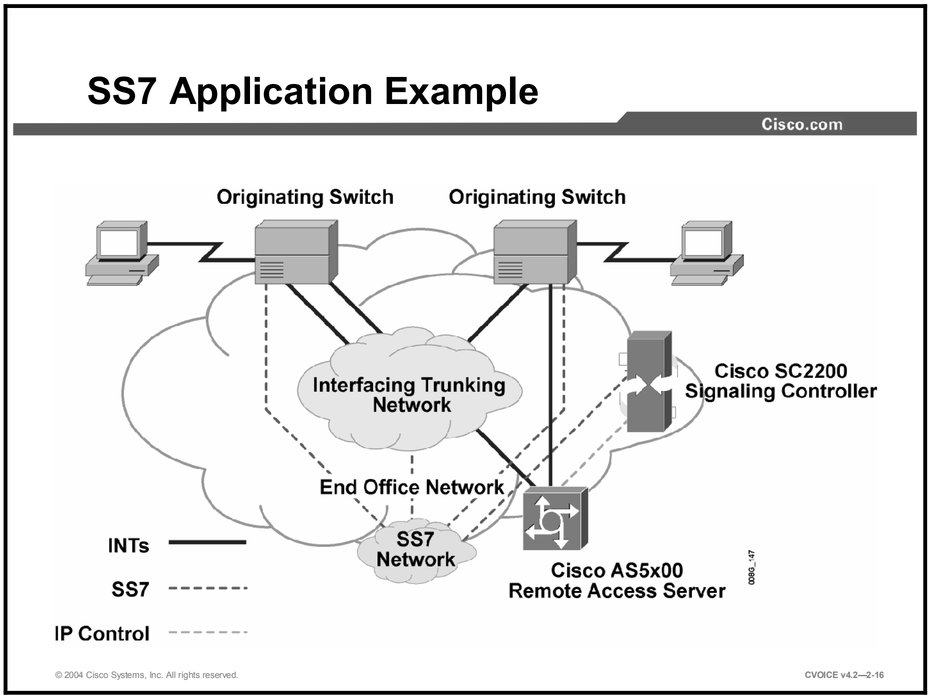


The ITU-T (formerly the CCITT) developed SS7 in 1981. The primary functions and benefits of SS7 are as follows:

- Fast call setup is handled by high-speed circuit-switched connections.
- PBX transaction capabilities (that is, call forwarding, call waiting, call screening, and call transfer) are extended to the entire network.
- A dedicated control channel exists for all signaling functions.
- Each associated trunk group needs only one set of signaling facilities.
- Information, such as address digits, can be transferred directly between control elements.
- There is no chance of mutual interference between voice and control channel, because SS7 is out-of-band signaling.
- Because the control channel is not accessible by the user, possible fraudulent use of the network is avoided.
- Connections involving multiple switching offices can be set up more quickly.

Note	The channel used for CCS does not need to be associated with any particular trunk group.

Example: SS7 Application

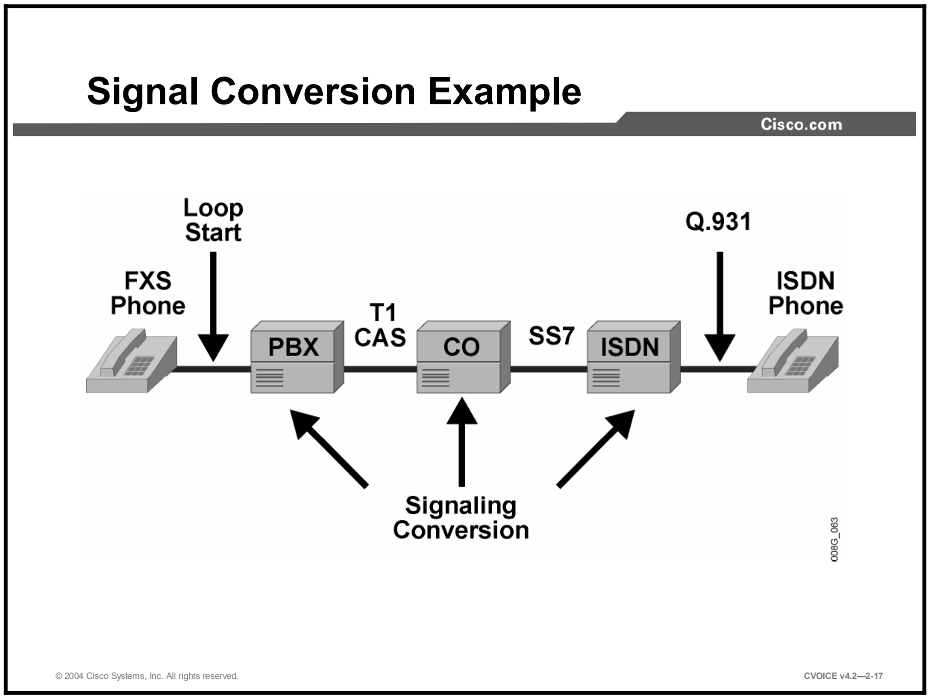


The figure here depicts an implementation of SS7. As a function of customer networks, SS7 can be implemented as CCS across a telephony network enterprise. Using Cisco equipment, service providers can implement SS7 on their networks. Cisco has developed several solutions that support off-loading IP traffic from public networks and that support the direct connection of network access servers to the PSTN using SS7 links. These solutions utilize the Cisco SC2200, BTS 10200, and AS5x00, giving service providers a proven and cost-efficient SS7 solution for connecting dial-access servers and voice gateways to the PSTN.

The Cisco AS5x00 family provides carrier-class, high-density connectivity for VoIP and dial subscribers. The product set supports a wide range of IP services (including voice) and enables carriers and Internet service providers (ISPs) to cost effectively support increased subscriber services and an increasing subscriber base.

Signaling Systems Interworking

This topic describes how different signaling systems interoperate on the same voice system.



In some implementations, it is necessary to convert from one signaling format to another. Conversion is necessary to allow different systems to signal each other. The figure illustrates an example of signal conversion.

Example: Signal Conversion

The FXS phone is using FXS loop-start signaling to connect to the PBX. The user dials "9" for an outside line, which carries the call on the T1 by using CAS. After the call reaches the CO, it travels via an SS7 signaled circuit to an ISDN switch. The call is then conveyed via Q.931 to the ISDN telephone at the called party location.

Other conversion applications exist in voice telephony. The telephony equipment must have the capability to perform these conversions transparently to the end users.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- PBXs and Cisco devices use T1 to convey voice.
- PBXs and Cisco devices use E1 to convey voice.
- Some examples of CCS are proprietary implementations, ISDN, DPNSS, and QSIG.
- ISDN can be implemented in two different ways: BRI and PRI.
- The QSIG protocol provides signaling of PBX devices.
- DPNSS is an early digital signaling system.
- SIGTRAN conveys SS7 over IP networks.
- SS7 can be implemented as a common signaling system across a telephony network enterprise.
- Conversion is necessary to convert from one signaling format to another.

 CVOICE v4.2—2-18

References

For additional information, refer to this resource:

- *Voice Network Signaling and Control*
  http://www.cisco.com/warp/public/788/signalling/net_signal_control.html

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 2-1: Lab Familiarity

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) In a T1 using CAS, what is the F bit in the SF used for?

A) framing

B) CRC

C) supervisory channel

D) signaling

Q2) In CAS, what four types of control information are provided by the A and B bits in SF? (Choose four.)

A) near- and far-end off-hook

B) idle

C) ringing

D) addressing

E) congestion

F) reorder

Q3) How many timeslots are there in an E1 frame using CAS?

A) 16

B) 24

C) 30

D) 32

Q4) In an E1 frame using CAS, which bits are reserved for signaling?

A) all the bits in timeslot 16

B) all the bits in timeslot 17

C) the first bit in timeslot 17

D) the least significant bit in timeslot 1

Q5) Which signaling method is NOT an example of CCS?

A) DPNSS

B) ISDN

C) QSIG

D) RBS

E) SS7

Q6) If vendors use a proprietary CCS protocol between their PBXs, Cisco devices must be configured using which protocol?

A) CCS

B) CCSS7

C) SS7

D) T-CCS

Q7) What type of information is carried in the D channel of ISDN?

A) voice

B) video

C) data

D) signaling

Q8) How many B channels and D channels are implemented on a PRI T1?

A) 1 B channel and 1 D channel

B) 2 B channels and 1 D channel

C) 23 B channels and 1 D channel

D) 30 B channels and 1 D channel

Q9) What type of interfaces support QSIG?

A) BRI

B) PRI

C) T1

D) E1

Q10) Which ISDN standard is QSIG based on?

A) Q.2931

B) Q.931

C) Q.920

D) Q.93B

Q11) DPNSS was developed by which organization?

A) IETF

B) British Telecom

C) ISO

D) ITU

Q12) DPNSS operates over what type of physical interface?

A) Ethernet

B) SMDS

C) ISDN

D) DSL

Q13) At Layer 4 of the TCP/IP protocol stack, SIGTRAN relies on what protocol?

A) User Datagram Protocol

B) Transmission Control Protocol

C) Internet Voice Communications Messaging Protocol

D) Stream Control Transport Protocol

Q14) SIGTRAN describes the way that the IP protocol carries which kind of messages?

A) SS7

B) Q.931

C) CallManager

D) ICMP

Q15) Which of the three following signaling protocols provide billing information and call establishment and routing? (Choose three.)

A) SS7

B) SigTran

C) Q.931

D) ESF

E) SF

Q16) SS7 was developed by which standards body?

A) IETF

B) ISO

C) ITU

D) Cisco

Q17) Why is signal conversion necessary?

A) To allow different systems to signal each other

B) To convert G.711 to G.729

C) To convert voice to data

D) To encode and compress voice

Q18) Which signaling method would be required between a CO and an ISDN switch?

A) loop start

B) T1 CAS

C) SS7

D) E1 CAS

E) RBS

Quiz Answer Key

Q1) D

Relates to: CAS Systems: T1

Q2) A, B, C, D

Relates to: CAS Systems: T1

Q3) D

Relates to: CAS Systems: E1

Q4) B

Relates to: CAS Systems: E1

Q5) D

Relates to: CCS Systems

Q6) D

Relates to: CCS Systems

Q7) D

Relates to: ISDN

Q8) C

Relates to: ISDN

Q9) B

Relates to: QSIG

Q10) B

Relates to: QSIG

Q11) B

Relates to: DPNSS

Q12) C

Relates to: DPNSS

Q13) D

Relates to: SIGTRAN

Q14) A

Relates to: SIGTRAN

Q15) A, B, C

Relates to: SS7

Q16) C

Relates to: SS7

Q17) A

Relates to: Signaling Systems Interworking

Q18) C

Relates to: Signaling Systems Interworking

Understanding Fax and Modem over VoIP

Overview

This lesson describes the implementation of fax and modem traffic over a VoIP network. It explores both Cisco Systems and standard implementations of faxing, and various methods used to transport modem traffic over VoIP.

Relevance

This lesson provides information about fax and modem traffic over VoIP for purposes of implementation. This lesson is necessary to educate the student in the methods used to implement a VoIP network.

Objectives

Upon completing this lesson, you will be able to implement an effective method of transporting fax and modem traffic over a VoIP network. This includes being able to meet these objectives:

- State the method used for conveying fax using Cisco fax relay
- State the method used for conveying fax using T.38 fax relay
- State the applications and the method used to convey fax using T.37 store-and-forward fax
- Describe how fax pass-through operates in a VoIP network
- Describe how modem pass-through operates in a VoIP network
- Describe how modem relay operates in a VoIP network

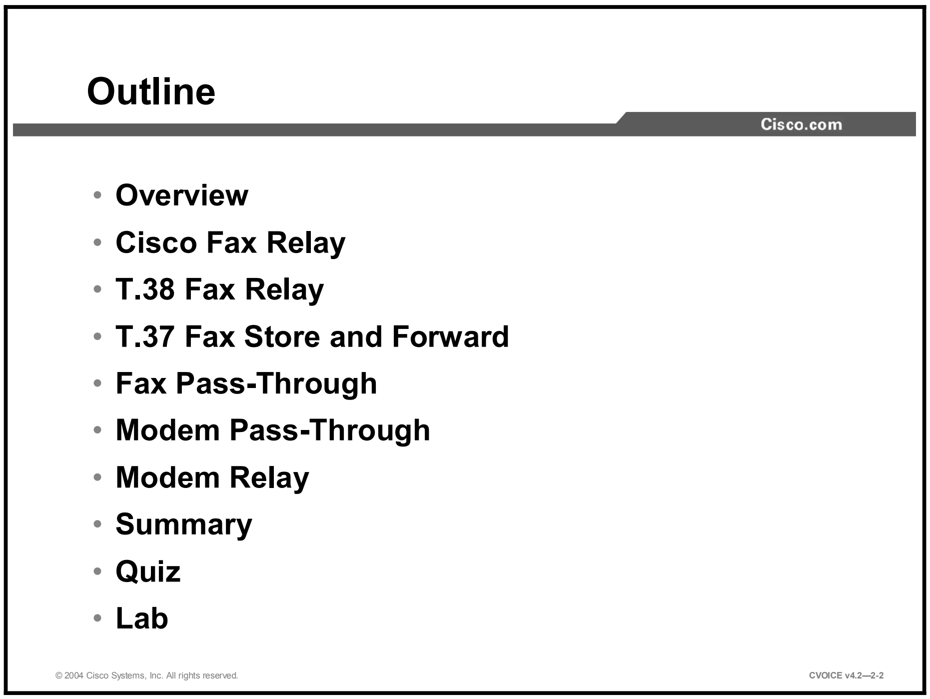
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of basic VoIP
- Familiarity with fax technology
- Familiarity with modem technology
- Familiarity with basic digital signal processor (DSP) use

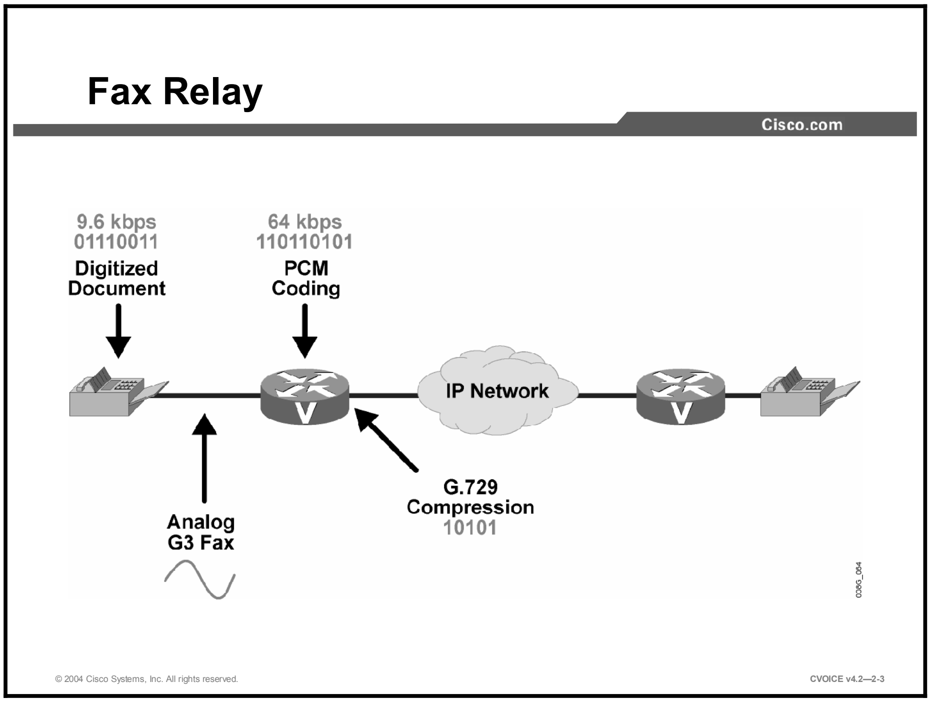
Outline

The outline lists the topics included in this lesson.

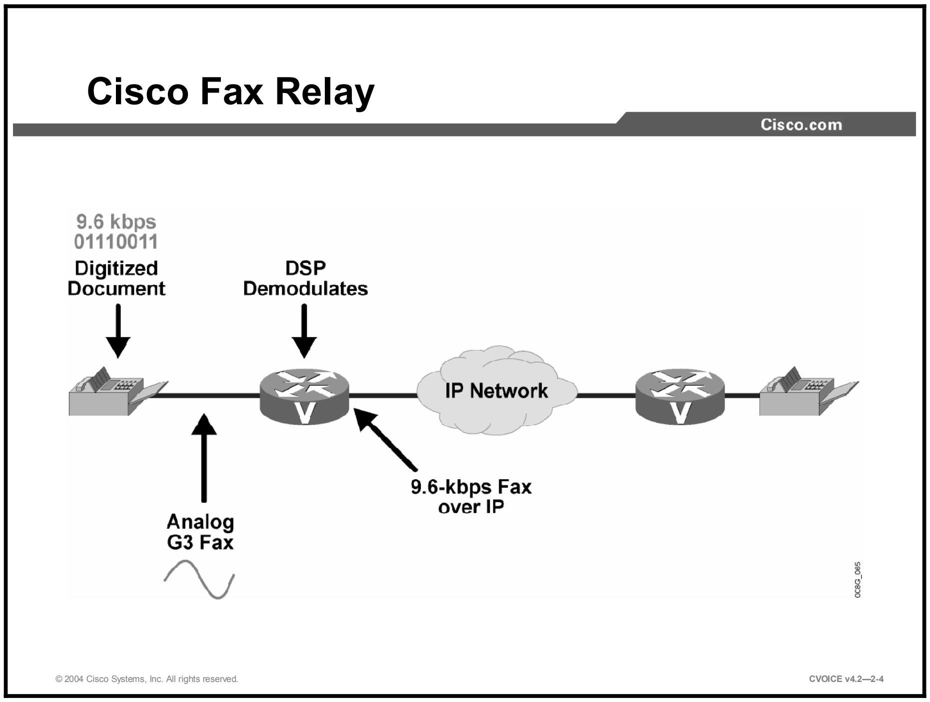


Cisco Fax Relay

This topic describes how the Cisco fax relay operates in a VoIP network.



The figure depicts a VoIP network set up for fax relay. Initially, fax calls are digitized representations of the contents on paper. The digitized bit stream is then converted to analog for transmission over voice circuits. If Cisco equipment treated fax calls like voice calls, the analog waveform would then be converted to G.711 PCM at 64 kbps and subsequently compressed before transmission across the VoIP network. Treating fax calls like voice calls is impractical because there are too many conversions and because the coding and compression schemes are designed to convey human speech, not fax modem tones.



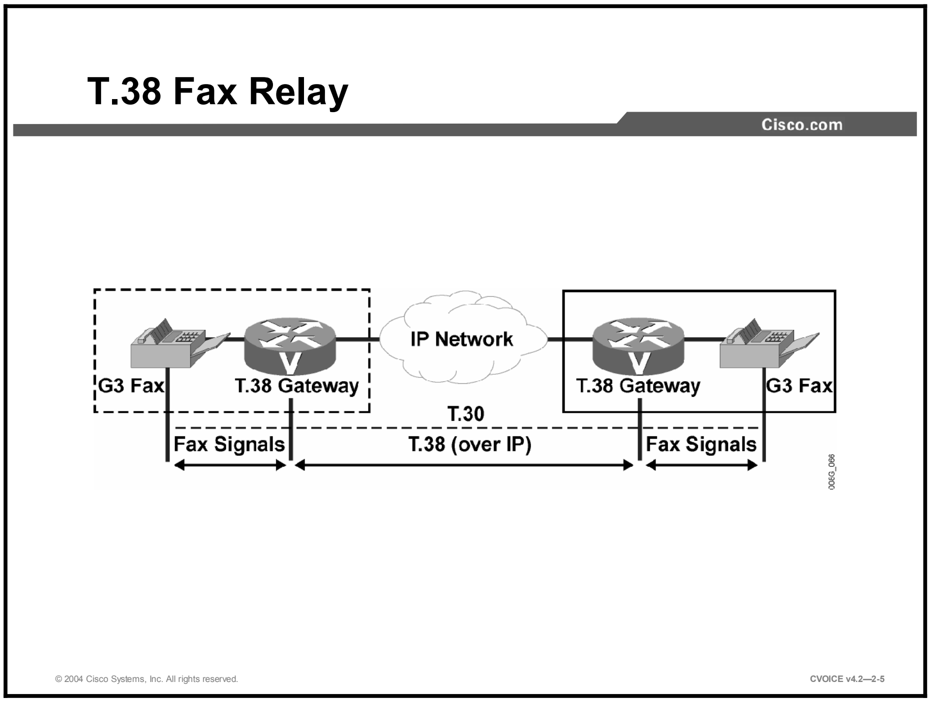
The figure depicts Cisco fax relay operating in a VoIP network. Using Cisco fax relay, the DSP chip first sets up the call as an end-to-end voice call. The DSP then recognizes the tones as those coming from a fax machine. The local DSP assumes the role of a fax modem, converting the analog data back to the original digitized bit stream. Acting as the fax modem, the DSP is downshifted in speed to 9.6 kbps to save bandwidth over the IP path. The bit stream is then packaged in VoIP packets and identified as a fax. The remote DSP assumes a similar role and converts the bit stream to analog for reception by the remote fax machine.

Example: Cisco Fax Relay

Cisco fax relay is a proprietary protocol supported only on Cisco voice equipment. Cisco Systems pioneered this protocol in the early 1990s before standards were developed and ratified.

T.38 Fax Relay

This topic describes how ITU standard T.38 fax relay operates in a VoIP network.



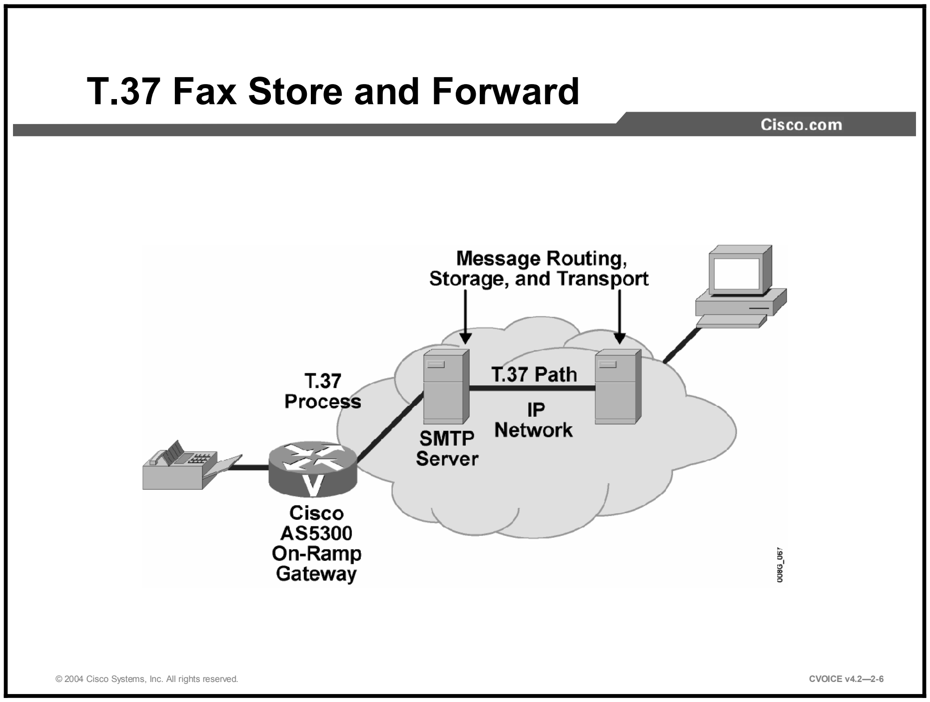
The T.38 approach is similar to the Cisco approach, but represents the industry standard. The figure depicts T.38 fax relay. In this approach, a T.38 gateway is required at both ends. A Cisco voice-enabled router can be configured as a T.38 gateway, and the transmission methodology is similar to the Cisco fax relay method. However, with T.38 it is possible to deliver documents to virtual fax machines, such as PCs and servers that are configured with software that is compatible with T.38 fax.

Example: T.38 Fax Relay

The T.38 method of fax relay is a derivative of Cisco fax relay but represents the industry standard. The T.38 method uses DSP chips to intercept the fax signal at each end and convert the signal to its original digital format, for example, 9600 bps Group 3 fax. The resulting bit stream is then packetized into the T.38 packet format and sent across the IP network. Since T.38 is an open standard from the ITU, it is compatible between different vendors.

T.37 Fax Store and Forward

This topic describes how ITU standard T.37 store-and-forward fax operates in a VoIP network.



The figure depicts the T.37 fax store-and-forward method. The T.37 standard is a way of delivering faxed documents as e-mail attachments. T.37 works by scanning a document, converting that document to tagged image file format (TIFF), and sending it to an e-mail address as an attachment using Simple Mail Transfer Protocol (SMTP). Cisco implements T.37 fax store and forward by using special gateways that are configured as on-ramps (transmitters of faxes) or off-ramps (receivers of faxes).

When configured on an on-ramp gateway, store-and-forward fax provides a way to transform standard fax messages into e-mail messages and TIFF attachments that travel to PCs on the network. When configured on an off-ramp gateway, e-mail with TIFF attachments transform into standard fax messages. These messages travel out of the packet network to standard Group 3 fax devices on the PSTN. When configured on both the on-ramp and off-ramp gateways of a network, store-and-forward fax allows temporary storage in the packet network for standard fax messages that cannot be delivered immediately.

Store-and-forward fax uses two different interactive voice response (IVR) applications for on-ramp and off-ramp functionality. The applications are implemented in two Tool Command Language (TCL) scripts that you can download from Cisco.com.

SMTP facilitates the basic functionality of store-and-forward fax, with additional functionality that provides confirmation of delivery using existing SMTP mechanisms such as Extended Simple Mail Transfer Protocol (ESMTP). Store-and-forward fax requires that you configure gateway dial peers and the following types of parameters:

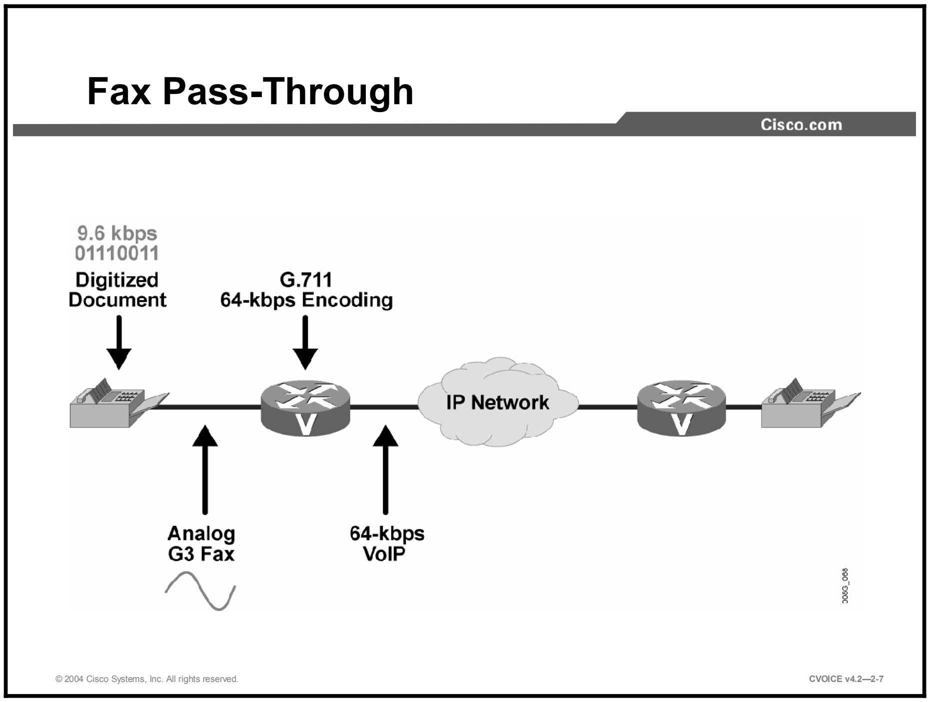
- **IVR application parameters and IVR security and accounting parameters:** Load the applications on the router and enable accounting and authorization for the application.
- **Fax parameters:** Specify the cover sheet and header information for faxes that are generated in the packet network.
- **Message Transfer Agent (MTA) parameters:** Define delivery parameters for the e-mail messages that accompany fax TIFF images.
- **Message disposition notification (MDN) parameters:** Specify the generation of messages to notify e-mail originators when their fax e-mail messages are delivered.
- **Delivery status notification (DSN) parameters:** Instruct the SMTP server to send messages to e-mail originators to inform them of the status of their e-mail messages.
- **Gateway security and accounting parameters:** Define authentication, authorization, and accounting (AAA) for faxes that enter or exit the packet network.

Example: Store-and-Forward Fax

Internet users are offered free store-and-forward faxes from various ISPs. Once subscribed, it is possible to send and receive fax messages using the service. If someone wants to send you a fax, they dial your assigned fax number from any standard fax machine. The store-and-forward on-ramp gateway at the service provider converts the fax back to its digital bit stream, then converts it into a graphics file (usually JPG). The completed graphics file is then e-mailed to you as an attachment.

Fax Pass-Through

This topic describes how fax pass-through operates in a VoIP network.



This figure depicts fax pass-through in a VoIP network. Fax pass-through occurs when incoming T.30 fax data is not demodulated or compressed for its transit through the packet network. The two fax machines communicate directly with each other over a transparent IP connection.

When a gateway detects a fax tone, it switches the call to a high-bandwidth codec. The fax traffic, still in PCM form, travels in-band over VoIP using G.711 with no VAD. This method of transporting fax traffic takes a constant 64-kbps (payload) stream end to end for the duration of the call. It is very sensitive to packet loss, jitter, and latency in the IP network, although packet redundancy can be used to mitigate the effects of packet loss.

Fax pass-through is supported under the following call control protocols:

- H.323
- Session initiation protocol (SIP)
- Media Gateway Control Protocol (MGCP)

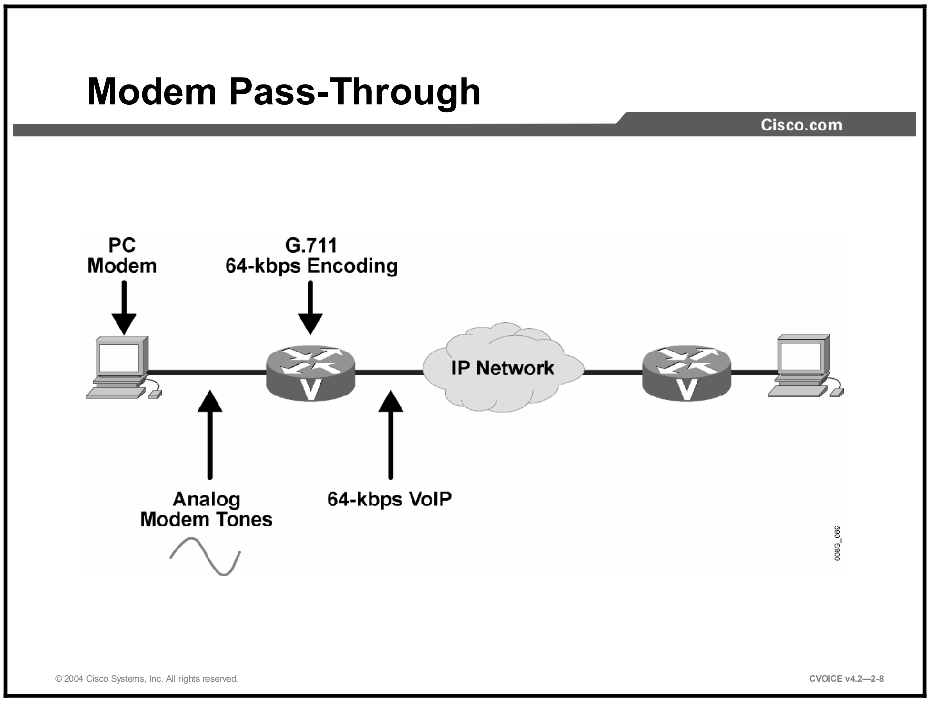
Note	Echo cancellation is enabled and preferred for pass-through using Cisco IOS Release 12.0(3) T and later. Earlier versions of Cisco IOS software required that you disable echo cancellation.

Example: Fax Pass-Through Application

Fax pass-through is applicable when connecting to a third party voice gateway that does not support T.38 fax relay. Fax pass-through treats the fax call as a simple G.711 voice call with no special handling for fax. When connecting a Cisco fax-enabled router to a third party voice-enabled router that does not support fax, you should use fax pass-through. The originating Cisco router will treat the call as a G.711 voice call and will not compress, thus preserving the analog properties of the waveshape for the receiving FAX machine. The disadvantage here is that it takes 64K plus overhead to complete the FAX call.

Modem Pass-Through

This topic describes how modem pass-through operates in a VoIP network.



The figure depicts modem pass-through in a VoIP network. Modem pass-through is similar to fax pass-through, except that there is a computer modem at each end of the connection. The two modems communicate directly with each other over a transparent IP connection.

When a gateway detects a modem tone, it switches the call to a high-bandwidth codec. The modem traffic, still in PCM form, travels in band over VoIP using G.711 with no VAD. This method of transporting modem traffic takes a constant 64-kbps (payload) stream end to end for the duration of the call. It is highly sensitive to packet loss, jitter, and latency in the IP network, although packet redundancy can be used to mitigate the effects of packet loss. Packet redundancy is defined in RFC 2198, and describes a way in which RTP carries the modem audio essentially twice. The redundant packets are sent in case there is packet loss. This scheme obviously produces significant overhead, and may not be acceptable in all applications. You can enable or disable packet redundancy when configuring modem pass-through.

The following call control protocols support modem pass-through:

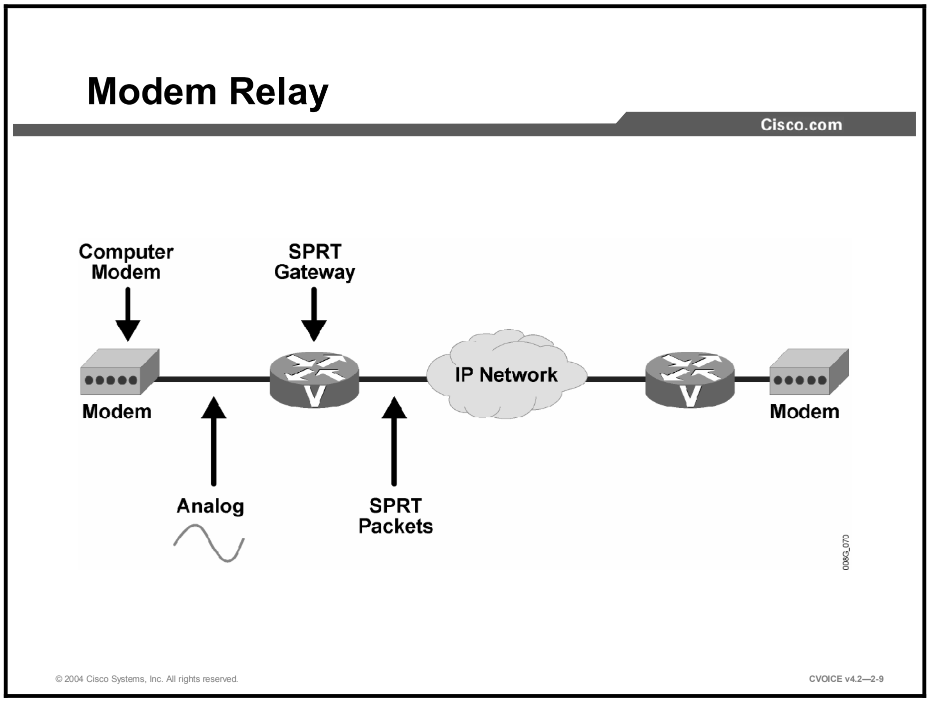
- H.323
- SIP
- MGCP

Example: Modem Pass-Through Application

Like fax pass-through, modem pass-through treats the call as a G.711 voice call with no special handling. It is utilized when the gateways serve as a dialup application for terminals or alarm systems. If your company utilizes alarm systems in multiple buildings throughout a WAN application, and the alarm system requires modems to dial into a central server, you can use modem pass-through on your VoIP network. This application will eliminate the cost of separate long distance dial telephone lines for the modems.

Modem Relay

This topic describes how modem relay operates in a VoIP network.



The figure depicts modem relay in a VoIP network. When using modem relay, computer modem signals are demodulated at one gateway, converted to digital form, and carried in Simple Packet Relay Transport (SPRT) protocol packets to the other gateway. When it reaches the other gateway, the modem signal is re-created and remodulated, and then passed to the receiving computer modem. SPRT is a protocol running over User Datagram Protocol (UDP). At the end of the modem session, the voice ports revert to the previous configuration, and the DSPs switch back to the original voice codec. This method uses less bandwidth (Real-Time Transport Protocol [RTP] is not required) and is much less sensitive to jitter and clocking mismatches than modem pass-through.

The following call control protocols support modem relay:

- H.323
- SIP
- MGCP

Example: Modem Relay Application

Modem relay can be used in systems that support SPRT gateways. SPRT is a specialized protocol that does not carry all of the overhead associated with VoIP. Cisco gateways support SPRT; use of SPRT should be limited to connecting to another vendor that does not support modem relay or fax relay.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Cisco fax relay is a method of conveying fax in a VoIP network.
- The T.38 method represents the industry standard for conveying fax in a VoIP network.
- The T.37 standard delivers faxed documents as e-mail attachments.
- Fax pass-through occurs when incoming T.30 fax data is not demodulated or compressed for its transit through a packet network.
- Modem pass-through operates similar to fax pass-through and uses a computer modem at each end of a connection.
- Modem relay uses the SPRT protocol to send signals through a series of gateways.

 CVOICE v4.2—2-10

References

For additional information, refer to this resource:

- *Cisco Fax Services over IP*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t11/faxapp/index.htm

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 2-1: Lab Familiarity

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which device acts as a fax modem in Cisco fax relay?

A) the DSP

B) the gateway

C) the modem

D) the gatekeeper

Q2) What speed does the fax modem operate at in Cisco fax relay?

A) 64 kbps

B) 52 kbps

C) 16 kbps

D) 9.6 kbps

Q3) Which Cisco device or application can be configured as a T.38 gateway?

A) Cisco analog gateways

B) Cisco call-processing agents

C) Cisco carrier-class call agents

D) Cisco voice-enabled routers

Q4) What is the advantage of using T.38 as compared to Cisco proprietary fax relay?

A) T.38 has a faster transmission rate.

B) T.38 allows delivery of documents to virtual fax machines.

C) T.38 uses a more efficient transmission methodology.

D) T.38 only uses the G.711 codec type.

Q5) How can temporary storage for delivering fax messages be provided in a packet network?

A) Configure a fax relay buffer on the on-ramp gateways.

B) Configure T.37 store-and-forward fax relay on the on-ramp gateways.

C) Configure T.37 store-and-forward fax relay on the off-ramp gateways.

D) Configure T.37 store-and-forward fax relay on both the on-ramp and off-ramp gateways.

Q6) Which parameters must be configured in T.37 store-and-forward fax relay so SMTP servers notify e-mail originators of their message status?

A) DSN parameters

B) fax parameters

C) MDN parameters

D) MTA parameters

Q7) Which of the three following call control protocols support fax pass-through? (Choose three.)

A) H.323

B) SIP

C) MGCP

D) SGCP

E) SCCP

Q8) Which feature is enabled for fax pass-through using Cisco IOS Release 12.0(3) T and later?

A) echo cancellation

B) echo suppression

C) VAD

D) PCM

Q9) What is the advantage of using modem relay as compared to modem pass-through?

A) Modem relay requires less bandwidth.

B) Modem relay is supported by more call control protocols.

C) Modem relay provides more interoperability.

D) Modem relay uses compression mechanisms.

Q10) How does modem pass-through differ from fax pass-through?

A) VAD is enabled.

B) G.711 is not used.

C) There is a computer modem at each end.

D) 64-kbps bandwidth is required.

Q11) How can you reduce the effects of packet loss in modem pass-through?

A) introduce packet redundancy

B) increase bandwidth

C) reduce latency

D) use dedicated lines

Q12) Which protocol is used to carry modem signals across the network using modem relay?

A) SMTP

B) SPRT

C) MGCP

D) RTP

Quiz Answer Key

Q1) A

Relates to: Cisco Fax Relay

Q2) D

Relates to: Cisco Fax Relay

Q3) D

Relates to: T.38 Fax Relay

Q4) B

Relates to: T.38 Fax Relay

Q5) D

Relates to: T.37 Fax Store and Forward

Q6) A

Relates to: T.37 Fax Store and Forward

Q7) A, B, C

Relates to: Fax Pass-Through

Q8) A

Relates to: Fax Pass-Through

Q9) A

Relates to: Modem Pass-Through

Q10) C

Relates to: Modem Pass-Through

Q11) A

Relates to: Modem Pass-Through

Q12) B

Relates to: Modem Relay

Module 3

Configuring Voice Interfaces

Overview

In this module you will learn basic configuration of analog and digital voice ports. You will also learn how to fine-tune voice ports with port-specific configurations.

Module Objectives

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

Module Objectives

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

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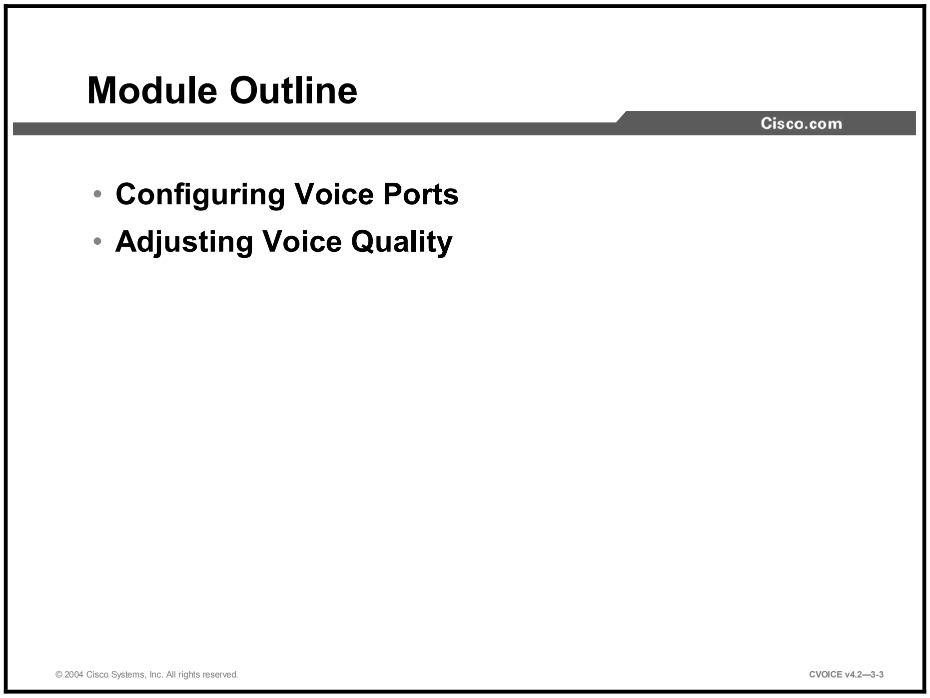
Module Outline

The outline lists the components of this module.



Configuring Voice Ports

Overview

This lesson details the configuration parameters and troubleshooting commands for analog and digital voice ports.

Relevance

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

Objectives

Upon completing this lesson, you will be able to configure analog and digital voice interfaces as new devices are introduced into the voice path. This includes being able to meet these objectives:

- Provide examples of seven types of voice port applications
- Set the configuration parameters for FXS voice ports
- Set the configuration parameters for FXO voice ports
- Set the configuration parameters for E&M voice ports
- Set timers and timing requirements on ports to adjust the time allowed for specific functions
- Set the configuration parameters for digital voice ports
- Set the configuration parameters for ISDN voice ports
- Configure T-CCS in a VoIP environment
- Use the **show**, **debug**, and **test** commands to monitor and troubleshoot voice ports

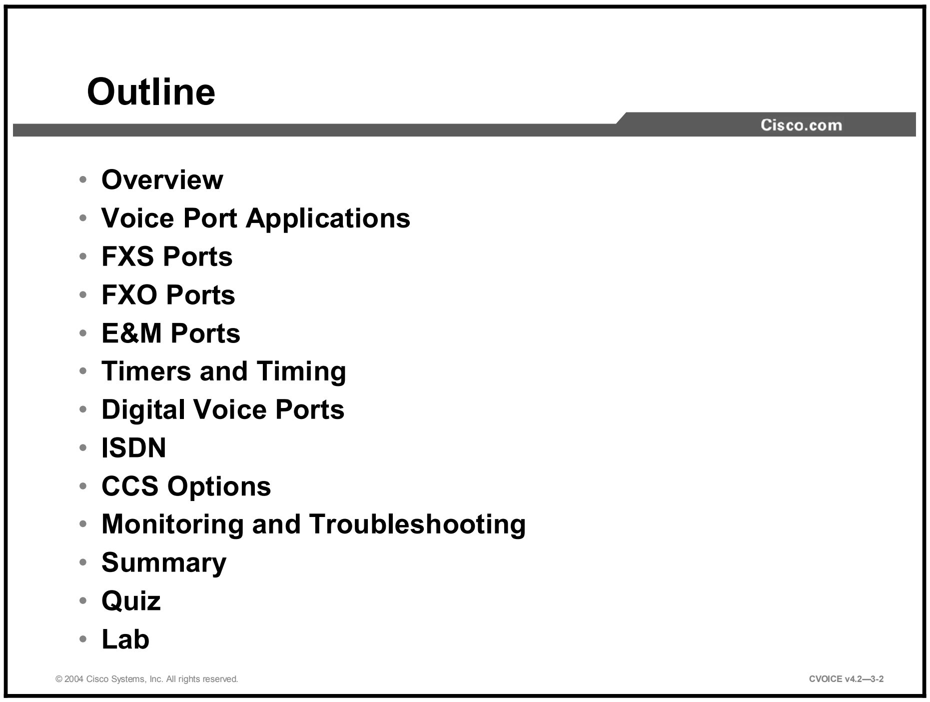
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with Cisco IOS commands
- Familiarity with analog and digital voice port usage

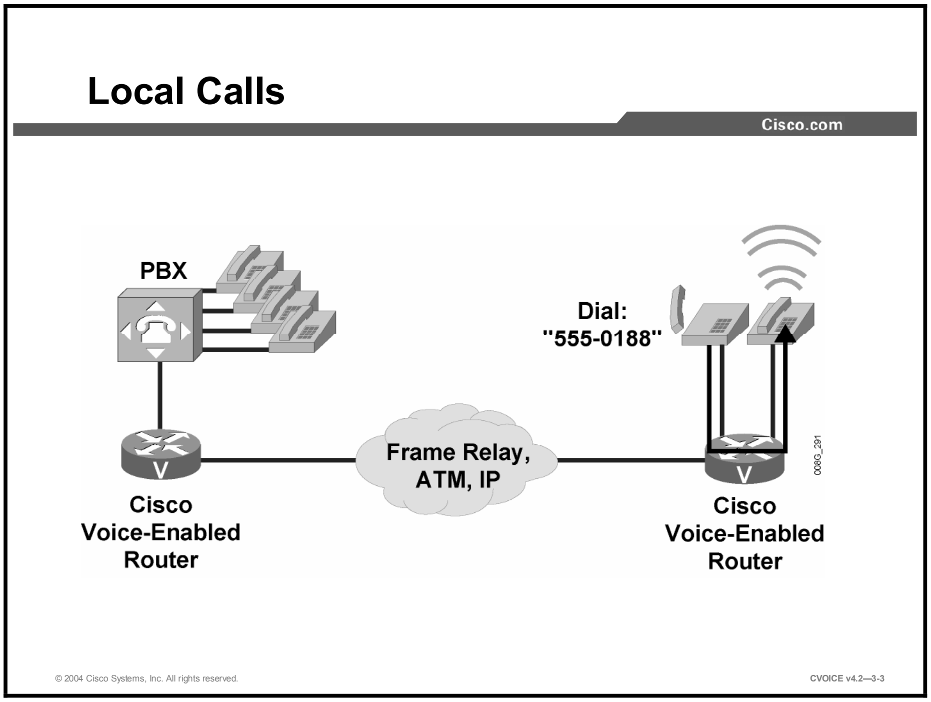
Outline

The outline lists the topics included in this lesson.

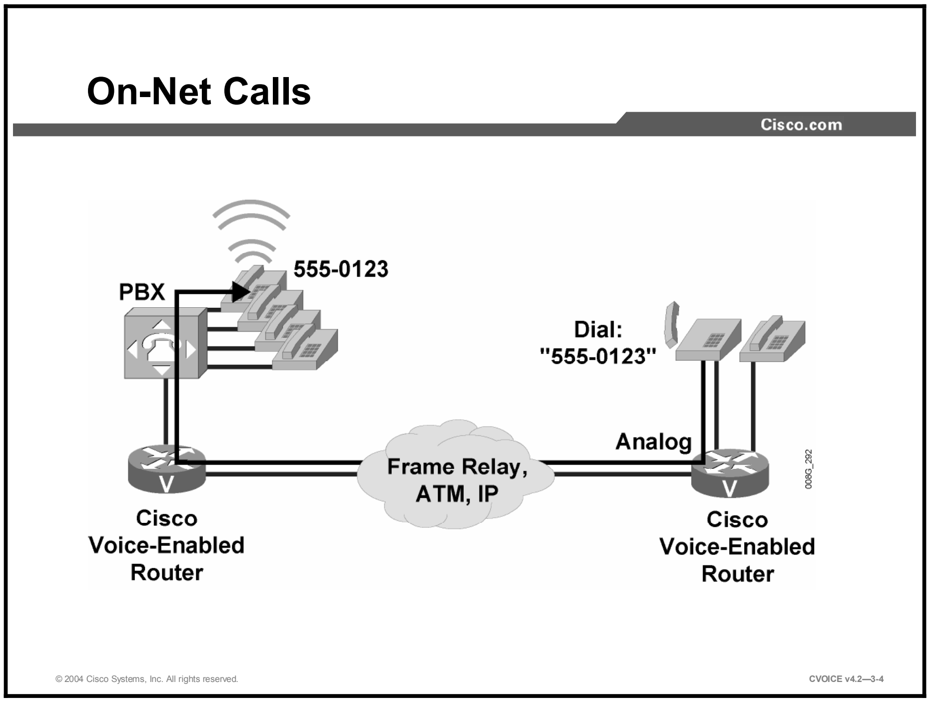


Voice Port Applications

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



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



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

Off-Net Calls

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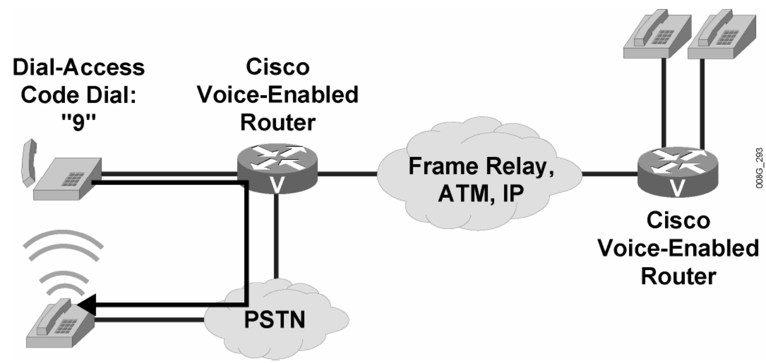

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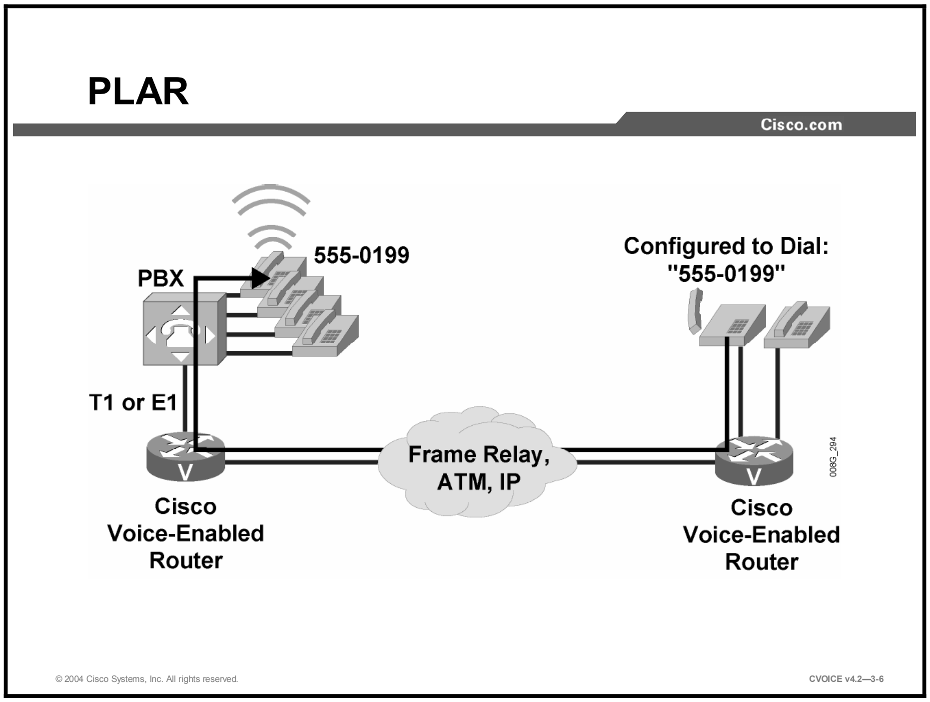
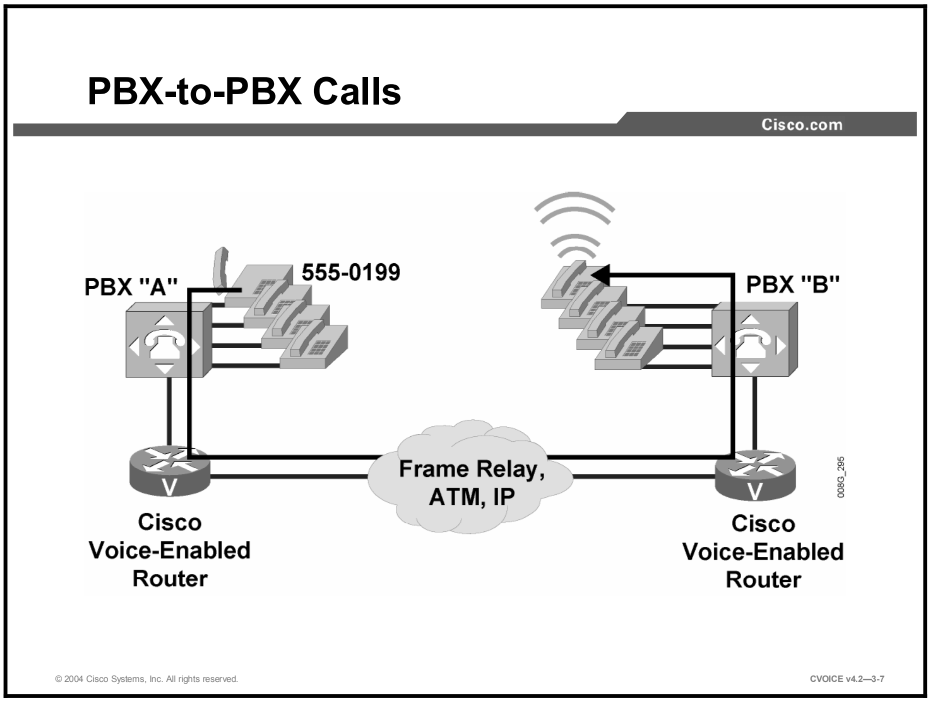
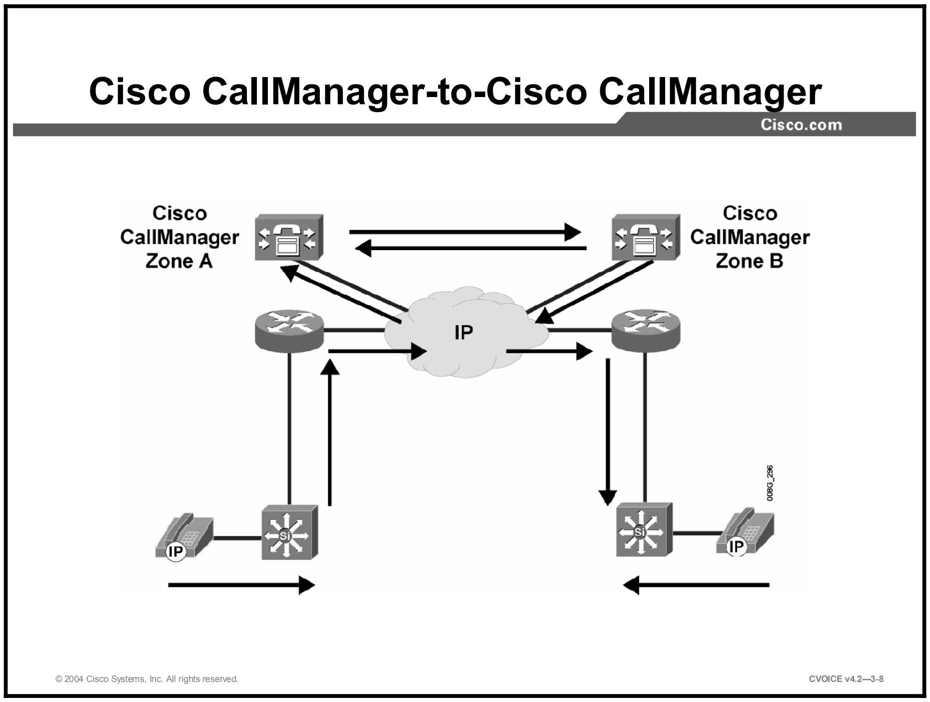
Off-net calls occur when, to gain access to the public switched telephone network (PSTN), the user dials an access code, such as "9," from a telephone that is directly connected to a Cisco voice-enabled router or PBX. The connection to the PSTN is a single analog connection via a Foreign Exchange Office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



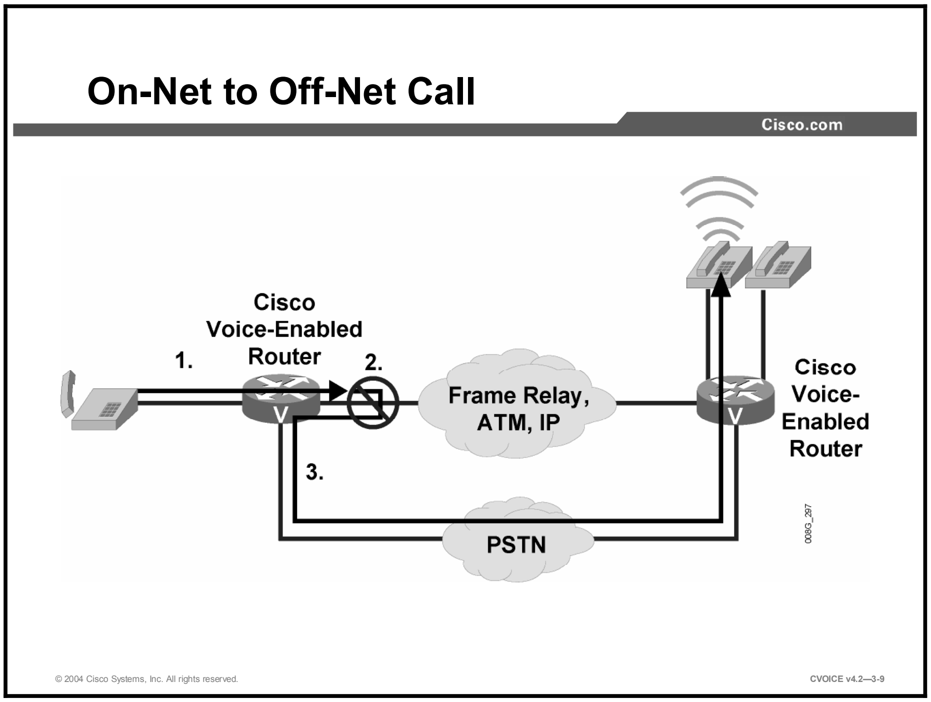
Private line, automatic ringdown (PLAR) calls automatically connect a telephone to a second telephone when the first telephone goes off hook. When this connection occurs, the user does not get a dial tone because the voice-enabled port that the telephone is connected to is preconfigured with a specific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



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



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



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

Example: Voice Port Applications

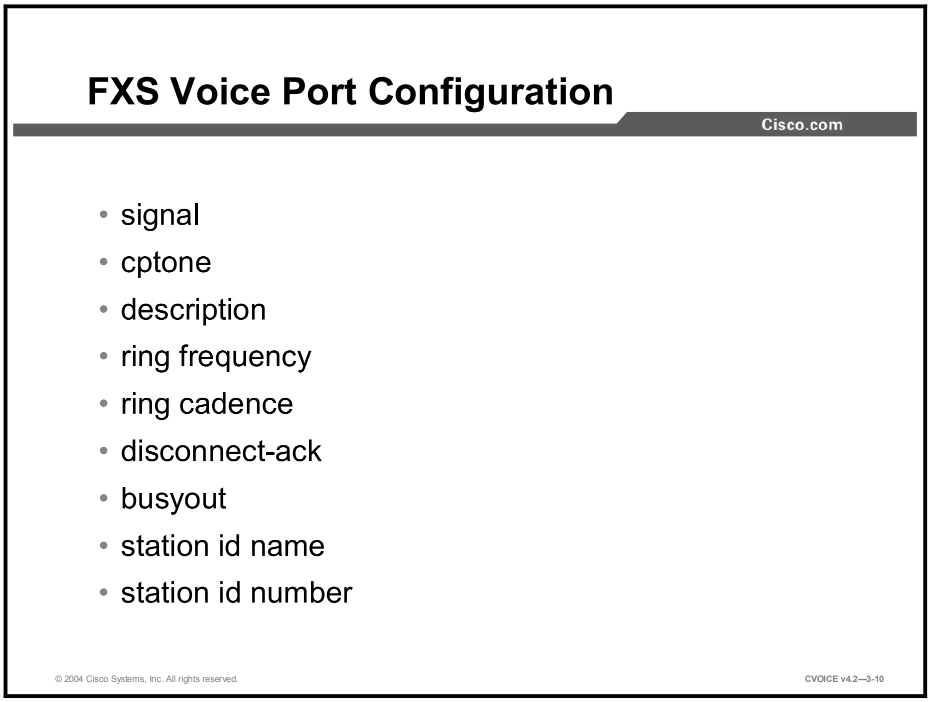
The table lists application examples for each type of call.

Voice Port Call Types

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

FXS Ports

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



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

Example: Configuring FXS Ports

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

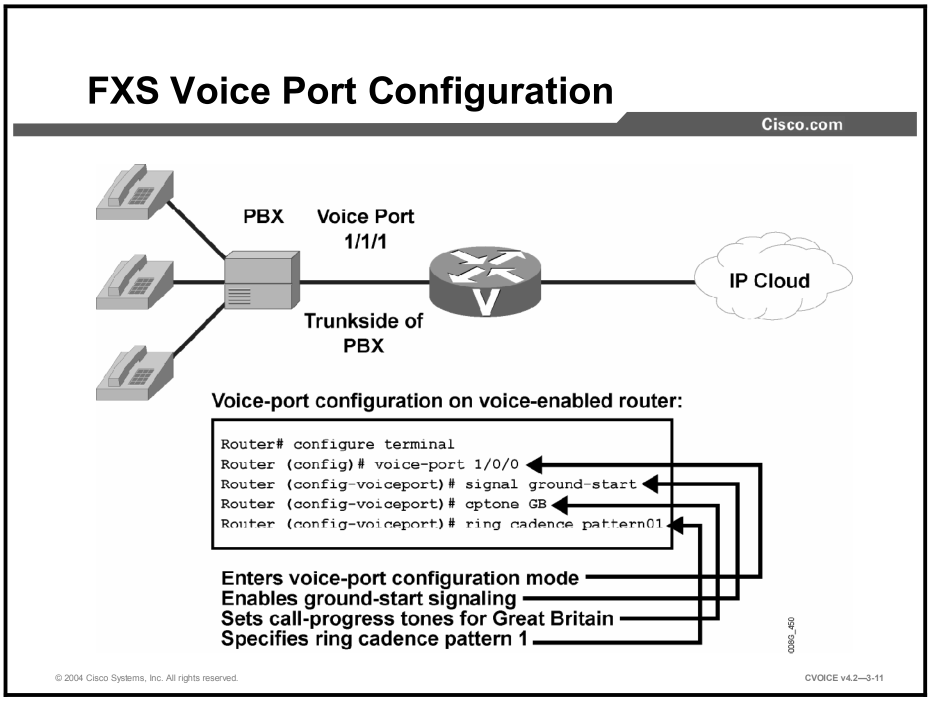
Configuration Parameters

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

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

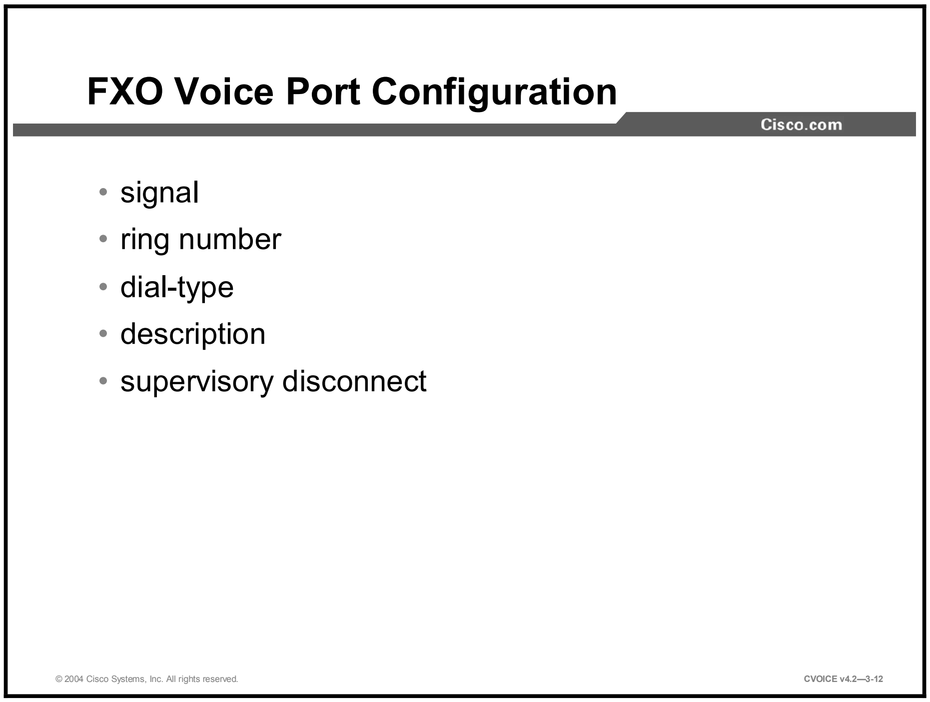
Example: FXS Port Configuration

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



FXO Ports

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



Configuration Parameters

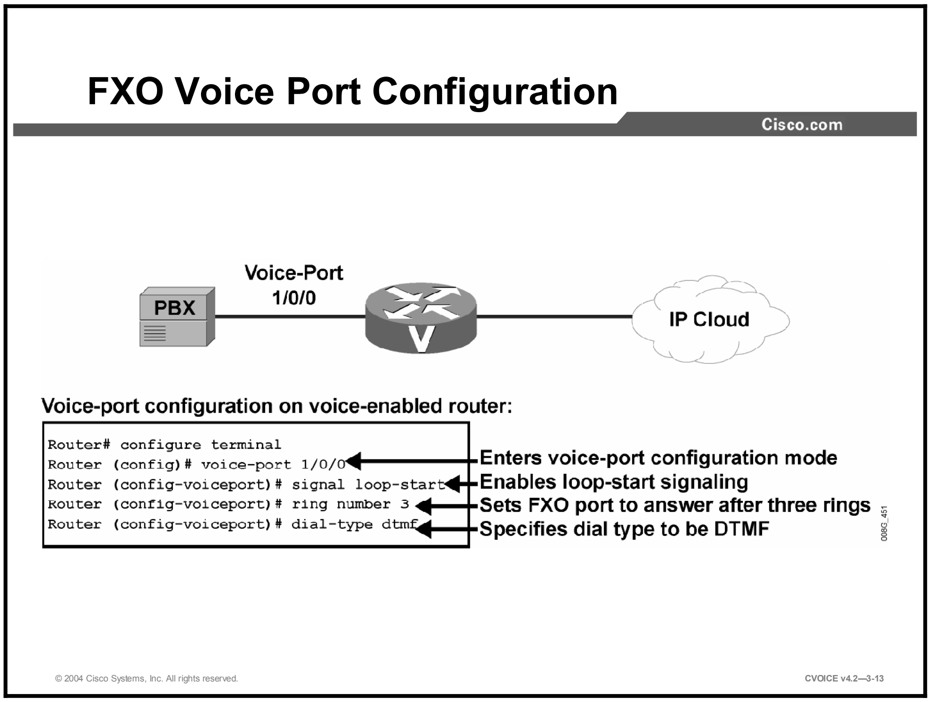
In most instances, the FXO port connection functions with default settings. FXO port configuration allows you to set parameters based on the requirements of the connection where default settings need to be altered or parameters set for fine-tuning. You can set the following configuration parameters:

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

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

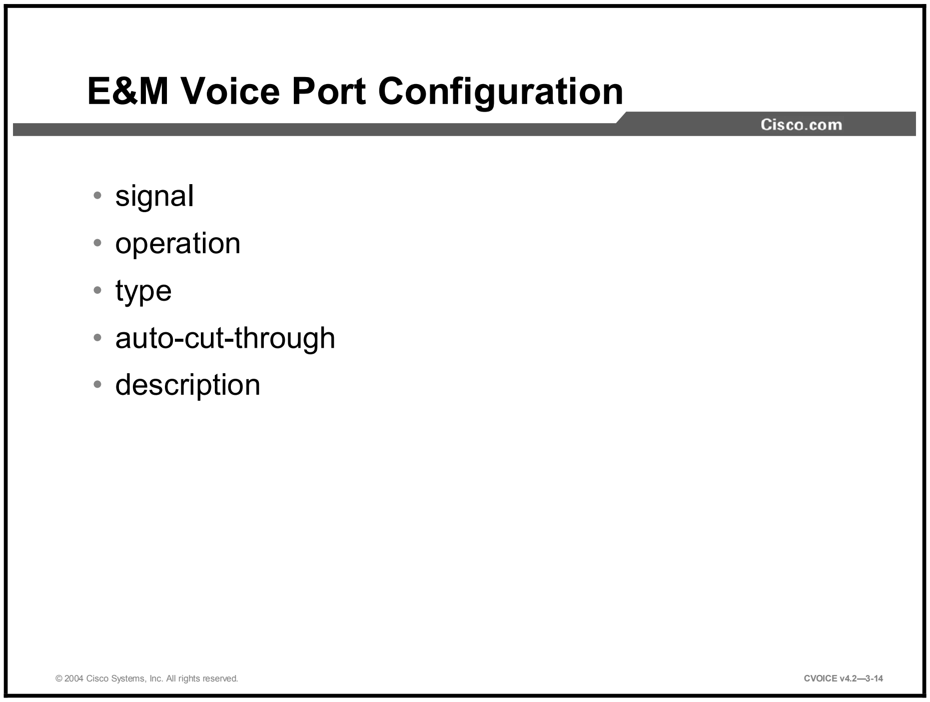
Example: FXO Port Configuration

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



E&M Ports

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



Configuration Parameters

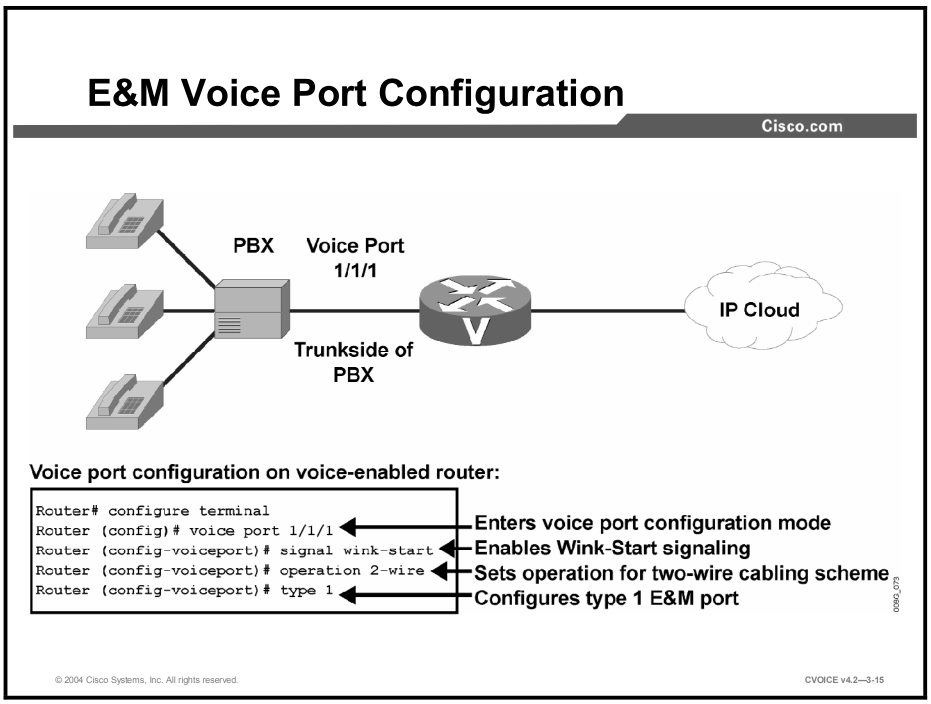
Although E&M ports have default parameters, you must usually configure these parameters to match the device that is connected to the E&M port. You can set the following configuration parameters:

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

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

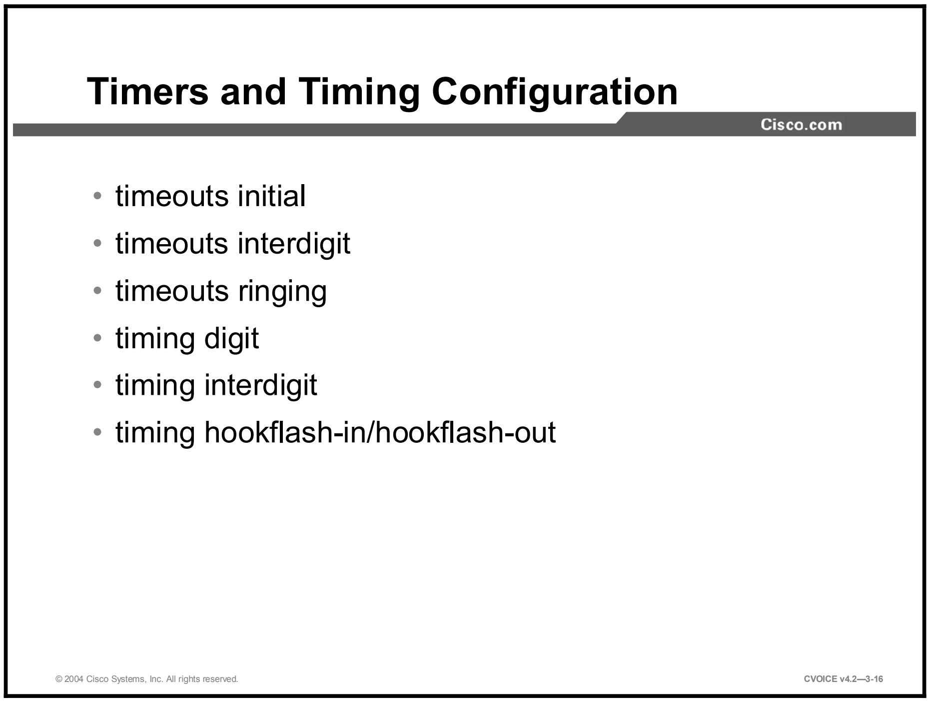
Example: E&M Port Configuration

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



Timers and Timing

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



Configuration Parameters

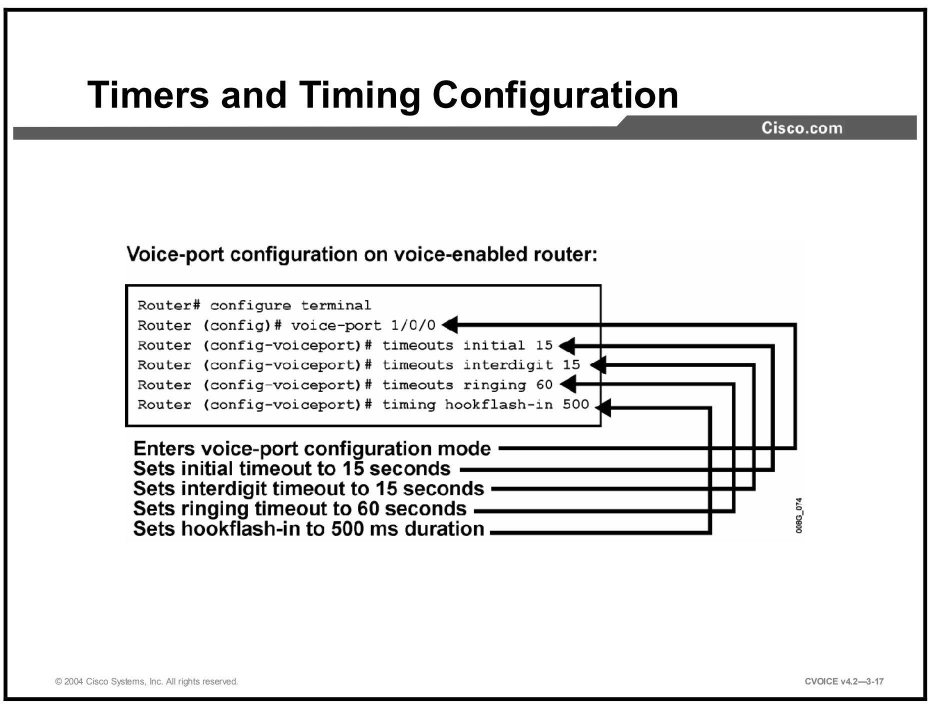
You can set a number of timers and timing parameters for fine-tuning the voice port. Following are voice port configuration parameters that you can set:

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

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

Example: Timers Configuration

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



Digital Voice Ports

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

Basic T1/E1 Controller Configuration

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

CVOICE v4.2—3-18

Configuration Parameters

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

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

T1/E1 Digital-Voice Configuration

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Create digital voice ports with the ds0-group **command**

- ***ds0-group-no***
- ***timeslot-list***
- ***signal-type***

CVOICE v4.2—3-19

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

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

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

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

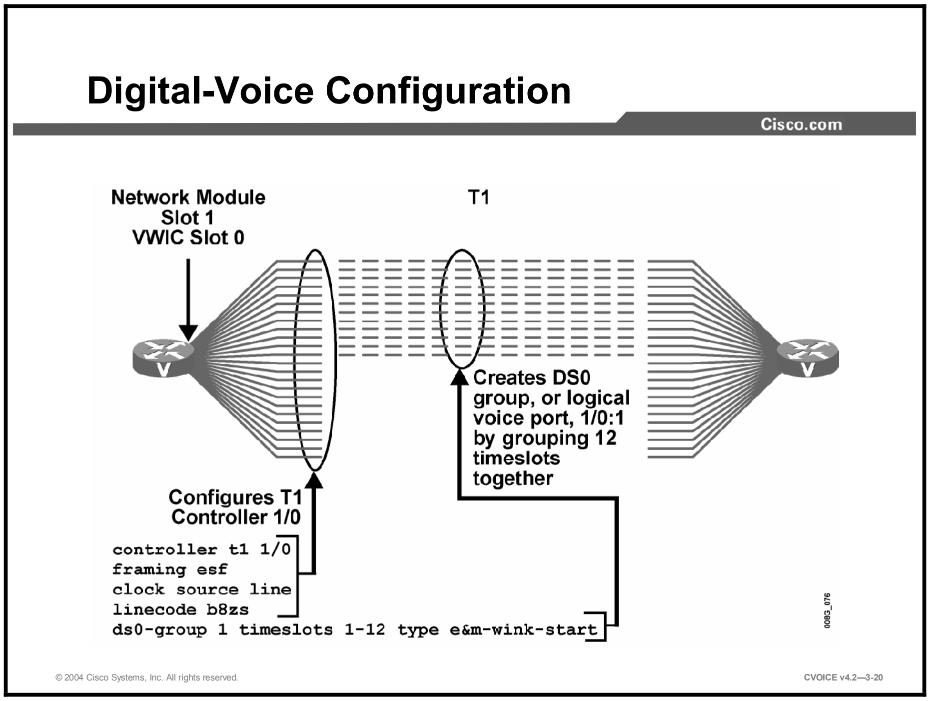
The **type** command defines the emulated analog signaling method that the router uses to connect to the PBX or PSTN. The type depends on whether the interface is T1 or E1.

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

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

Example: T1 Configuration

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



ISDN

This topic identifies ISDN configurations for voice ports.

ISDN Configuration

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

CVOICE v4.2—3-21

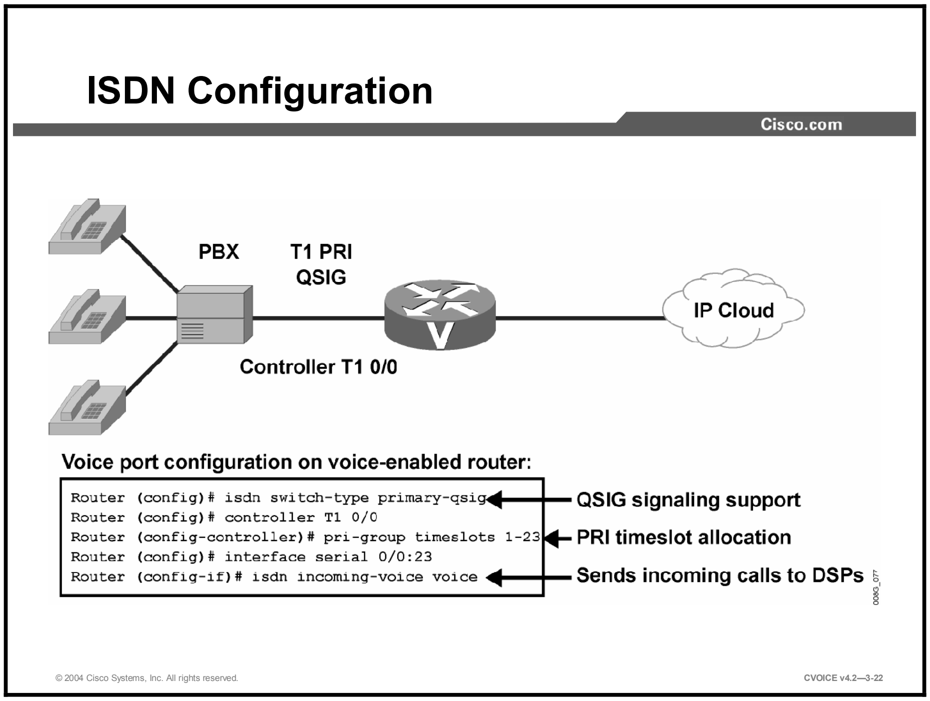
Configuration Parameters

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

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

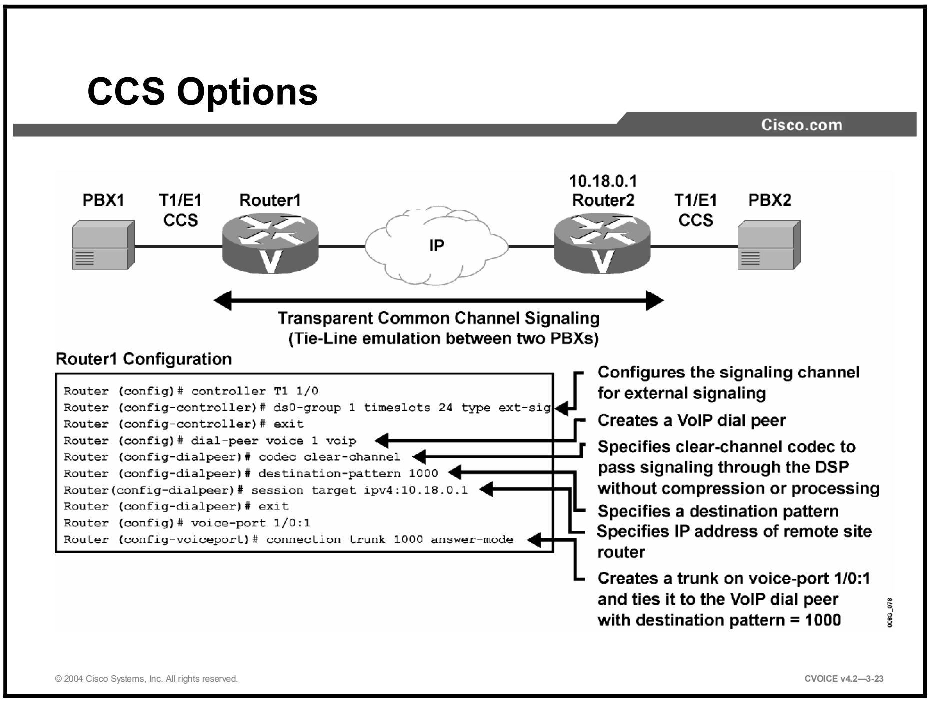
Example: ISDN QSIG Configuration

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



CCS Options

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



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

T-CCS Configuration Process

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

Step 1 Define the DS0 group.

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

Step 2 Create the dial peer.

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

Step 3 Create the voice port trunk.

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

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

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

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

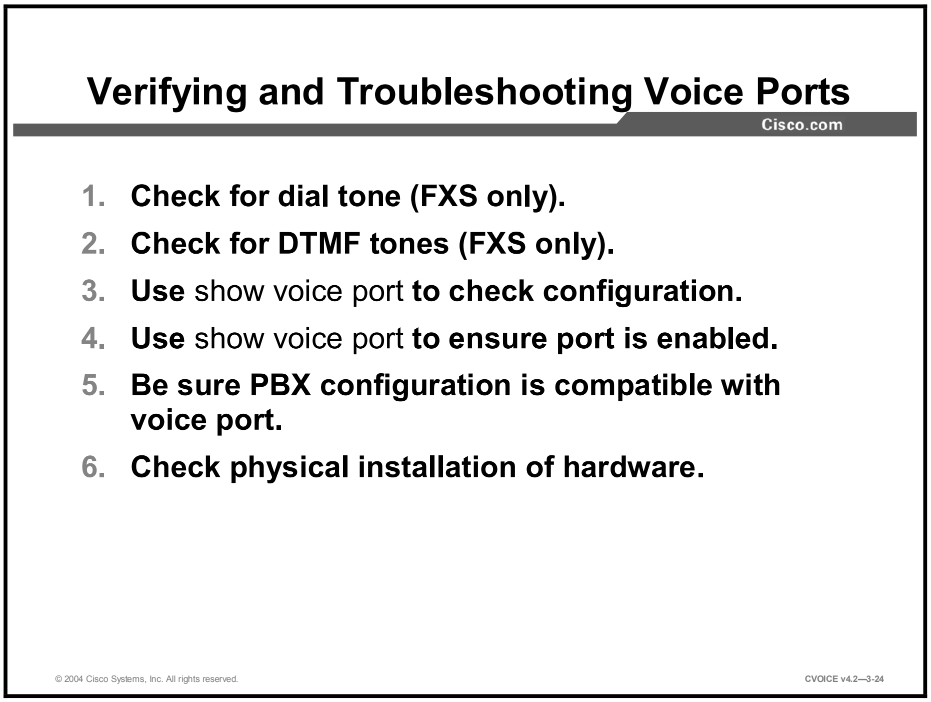
Step 3 The VoIP dial peer is configured for clear-channel codec and points to the IP address of the remote router (router 2) connecting the remote PBX (PBX2).

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

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

Monitoring and Troubleshooting

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



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

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

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

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

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

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

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

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

Commands to Verify Voice Ports

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

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 CVOICE v4.2—3-25

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

Test Commands

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

CVOICE v4.2—3-26

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

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

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

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

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

ISDN Commands

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

CVOICE v4.2—3-27

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

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- **Voice port applications include local, on-net, off-net, PLAR, PBX-to-PBX, Cisco CallManager-to-Cisco CallManager, and on-net to off-net calls.**
- **Configurable parameters on FXS ports include *signal, cptone, description, ring frequency, ring cadence, disconnect-ack, busyout, station id name*, and *station id number*.**
- **Configurable parameters on FXO ports include *signal, ring number, dial-type, description,* and *supervisory disconnect*.**
- **Configurable parameters on E&M ports include *signal, operation, type, auto-cut-through,* and *description*.**

CVOICE v4.2—3-28

Summary (Cont.)

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- **Configurable timer and timing parameters define initial digit and interdigit timing, digit and interdigit duration, as well as ringing time.**
- **Digital voice ports are created with the** ds0-group **command in the T1/E1 controller.**
- **ISDN configuration requires that the** pri-group **command specify timeslots used for voice and signaling.**
- **T-CCS allows for the transparent passing of proprietary PBX signaling across the IP network.**
- **The** show, debug, **and** test **commands are used for monitoring and troubleshooting voice functions in the network.**

CVOICE v4.2—3-29

References

For additional information, refer to these resources:

- *Configuring Voice Ports*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/12cgcr/voice_c/vcprt1/vcports.htm

- *Configuring and Troubleshooting Transparent CCS*
  http://www.cisco.com/warp/public/788/voip/trans_channel_signal.html

- *Configuring Voice Ports*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/vvfport.htm

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 3-1: Voice Port Configuration

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Match the type of voice application with its description.

A) local call

B) on-net call

C) off-net call

D) on-net to off-net call

_____ 1. a type of call for which the user dials an access code to connect to the PSTN

_____ 2. a type of call that is handled entirely by one router and does not go across an external network

_____ 3. a type of call that is rerouted through a secondary path when the primary path fails

_____ 4. a type of call that may be routed through one or more routers, but stays on the same network

Q2) If a client picked up a customer service handset and was automatically connected to customer service without dialing any digits, what kind of call would it be?

A) Cisco CallManager-to-Cisco CallManager call

B) PBX-to-PBX call

C) on-net call

D) local call

E) PLAR call

Q3) Which configuration parameter would you change to set the dial tone, busy tone, and ringback tone?

A) **cptone**

B) **ring frequency**

C) **ring cadence**

D) **description**

E) **signal**

Q4) Which situations most likely require changes to the FXS port default settings?

A) The caller and the called party are in different parts of the country.

B) The caller and the called party are in different countries.

C) The connection is a trunk to a PBX.

D) The FXS port configuration does not match the local PSTN switch configuration.

Q5) Which statement best describes supervisory disconnect signaling?

A) a power denial from the switch that lasts at least 350 ms

B) a disconnect message manually sent by the network administrator

C) signaling used by the main voice port of the PBX switch

D) a disconnect process overseen by the network administrator

Q6) When connecting an FXO port to a device that supports pulse dialing, which command is optional?

A) **signal**

B) **dial-type**

C) **description**

D) **supervisory disconnect**

Q7) The wrong cable scheme is specified during E&M port configuration. What is the likely outcome?

A) The signal will not be transmitted.

B) The voice traffic will go in one direction only.

C) The voice ports will not be activated.

D) The switch will not recognize a request for service.

Q8) A router provides current on the E-lead as soon as it sees current on the M-lead. What type of signaling is being used?

A) delay-start signaling

B) immediate-start signaling

C) Wink-Start signaling

D) QSIG

Q9) It is desired to set a number of seconds to wait for a subsequent digit to be dialed after an initial digit. Which parameter should be configured?

A) **timeouts initial**

B) **timeouts interdigit**

C) **timing digit**

D) **timing interdigit**

Q10) What is the default setting for the **timeouts ringing** configuration parameter?

A) 15 seconds

B) 60 seconds

C) 100 seconds

D) 180 seconds

Q11) What are the two options for the **framing** command on a T1 connection? (Choose two.)

A) SF

B) ESF

C) CRC4

D) AMI

E) B8ZS

F) HDB3

Q12) What are the two options for the **linecode** command on an E1 connection? (Choose two.)

A) SF

B) ESF

C) CRC4

D) AMI

E) B8ZS

F) HDB3

Q13) For E1 connections, which timeslot is allocated to the D channel?

A) 1

B) 16

C) 23

D) 31

Q14) Which of the following is a configuration command required for ISDN voice functionality?

A) **dial-type**

B) **disconnect-ack**

C) **busyout**

D) **pri-group**

E) **ds0-group**

Q15) What is the purpose of T-CCS?

A) to route calls between PBXs

B) to provide point-to-point connections between PBXs

C) to pass proprietary signaling between PBXs

D) to specify a channel for call signaling

Q16) The **trunk** *number* entered in configuring T-CSS in a VoIP network must match the **trunk** *number* entered in which command?

A) **timeslots**

B) **dial-peer**

C) **destination-pattern**

D) **session target**

Q17) After you enter the **test voice port switch fax** command, which command can you use to check whether the voice port can operate in fax mode?

A) **show voice port**

B) **show voice call**

C) **show fax port**

D) none of the above

Q18) Which two conditions can be checked by using the **show voice port** command? (Choose two.)

A) The data that is configured is correct.

B) The port is enabled.

C) The E&M interfaces are configured correctly.

D) The PBX setup values are correct.

Quiz Answer Key

Q1) 1-C, 2-A, 3-D, 4-B

Relates to: Voice Port Applications

Q2) E

Relates to: Voice Port Applications

Q3) A

Relates to: FXS Ports

Q4) B

Relates to: FXS Ports

Q5) A

Relates to: FXO Ports

Q6) B

Relates to: FXO Ports

Q7) B

Relates to: E&M Ports

Q8) B

Relates to: E&M Ports

Q9) B

Relates to: Timers and Timing

Q10) D

Relates to: Timers and Timing

Q11) A, B

Relates to: Digital Voice Ports

Q12) D, F

Relates to: Digital Voice Ports

Q13) B

Relates to: ISDN

Q14) D

Relates to: ISDN

Q15) C

Relates to: CCS Options

Q16) C

Relates to: CCS Options

Q17) B

Relates to: Monitoring and Troubleshooting

Q18) A, B

Relates to: Monitoring and Troubleshooting

Adjusting Voice Quality

Overview

Voice port settings affect voice quality. There are a number of settings that you can configure to enhance the quality of voice traffic on voice ports.

Relevance

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

Objectives

Upon completing this lesson, you will be able to configure analog and digital voice ports for optimal voice quality. This includes being able to meet these objectives:

- Describe the electrical characteristics of analog voice and the factors affecting voice quality
- Configure voice port parameters to fine-tune voice quality
- Configure echo cancellation on the voice ports to improve voice quality

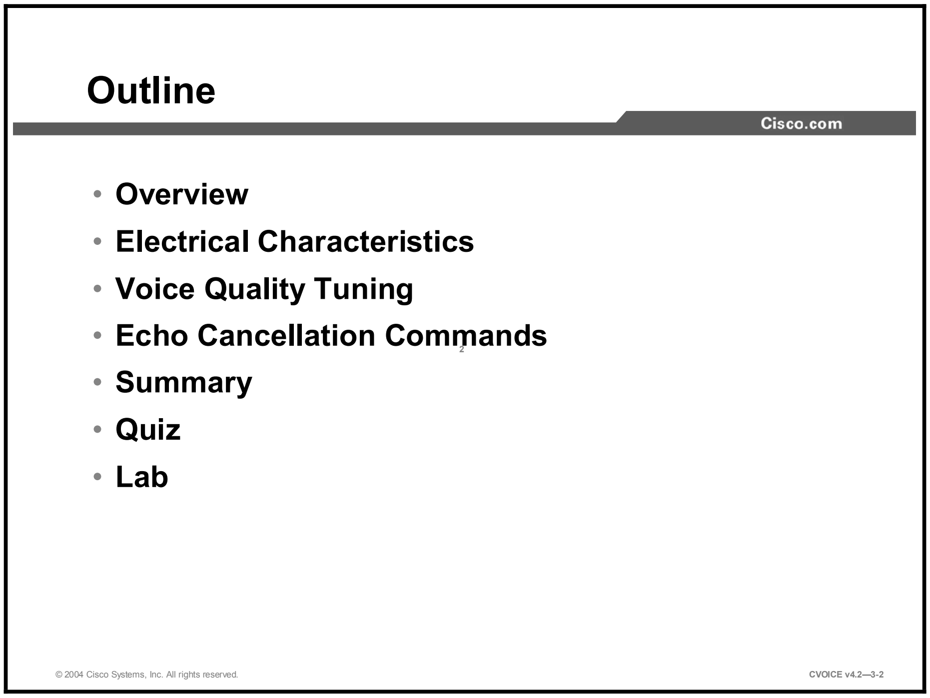
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with Cisco IOS configuration modes
- Familiarity with analog and digital voice port usage

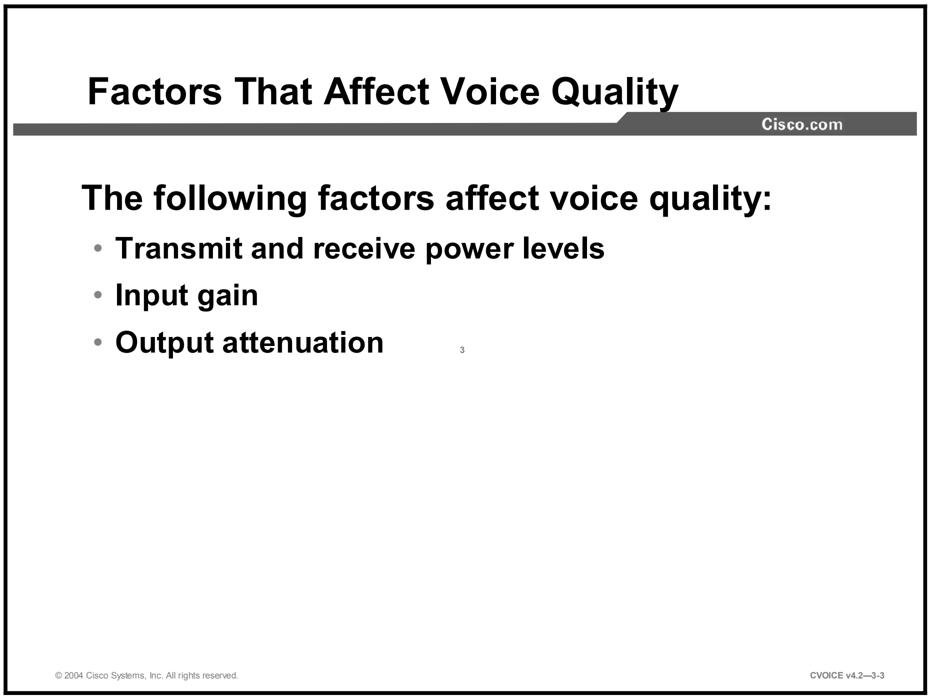
Outline

The outline lists the topics included in this lesson.



Electrical Characteristics

This topic describes the electrical characteristics of analog voice and the factors affecting voice quality.



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

Most initial voice signals enter the network through a two-wire local loop. Most switches connect to other switches through a four-wire connection. As voice travels through the network for delivery to the remote telephone, the voice signal must be passed from the two-wire local loop to the four-wire connection at the first switch, and from the four-wire connection at the switch to a two-wire local loop at the remote end. If the impedance at these two-wire to four-wire connections is not matched exactly, some of the voice signal reflects back in the direction of the source. As a result, originating callers hear their own voice reflected back. Sometimes, the reflected signal is reflected again, causing the destination to hear the same conversation twice.

In a traditional voice network, voice can reflect back; it usually goes unnoticed, however, because the delay is so low. In a VoIP network, echo is more noticeable because both packetization and compression contribute to delay.

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

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

Calculating Decibel Levels

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

Calculating network dB levels is often an exercise in simple number line arithmetic. The table provides common dB levels.

Calculating Decibel Levels

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Source 1 Out/In	Router 1 Adjustment	Net at Router 1	WAN __	Net at M Router 2	Router 2 Adjustment	Destination 1 In/Out
0 dB -->	–3 dB -->	–3 dB	__	–3 dB	± 6 dB -->	--> –9 dB
–9 dB <--	<--±6 dB	–3 dB	__	–3 dB	–3 dB	<-- 0 dB

CVOICE v4.2—3-4

Baselining Input and Output Power Levels

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

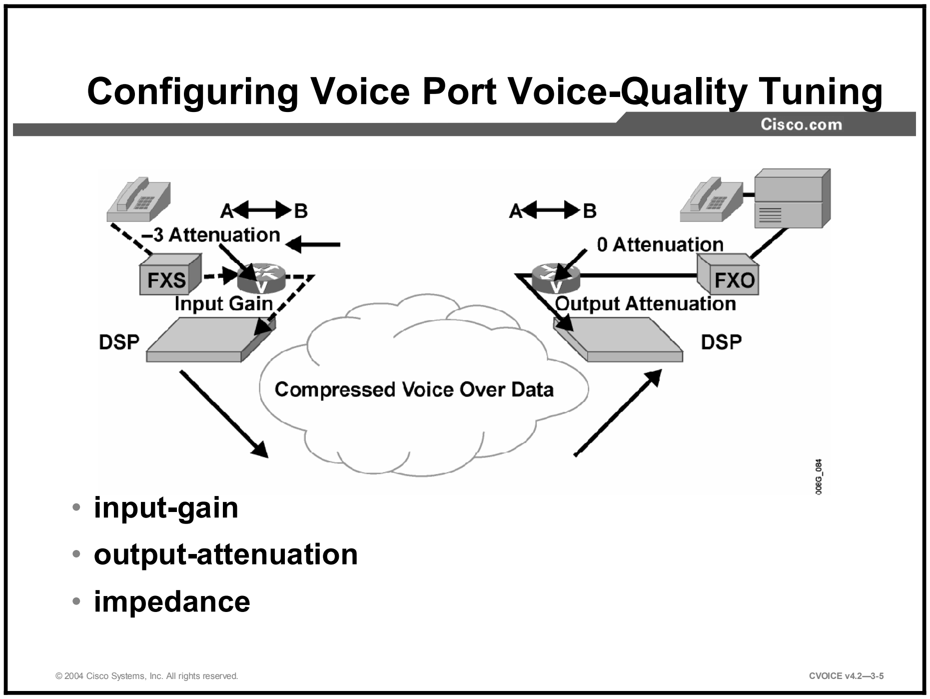
- Analog voice routers operate best when the receive level from an analog source is set at approximately –3 dB.
- In the United States and most of Europe, the receive (transmit) level that is normally expected for an analog telephone is approximately –9 dB. In Asian and South American countries, receive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

Example: Decibel Levels

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

Voice Quality Tuning

This topic describes voice-quality tuning configuration.



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

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

A variety of different factors, including input gain and output attenuation, can affect voice quality.

A loss plan looks at the required dB levels at specific interfaces, such as an analog FXS port connecting to a telephone, or an FXO port connecting to the PSTN. An analog voice router works best with a receive level of –3 dB. An analog telephone in North America and Europe works best with a receive level of –9 dB. Therefore, if the device connecting to that router provides a different level than the expected –3 dB, then input gain can be set to equalize it to –3 dB. If the output at the other end is a telephone that expects –9 dB, then the output voice port has to provide –6 dB output attenuation in addition to the –3 dB to send signaling to the telephone at the expected –9 dB levels. A systemwide loss plan looks at the dB levels of the initial input and the remote output ports and plans for the appropriate adjustments for end-to-end signal levels. You must consider other equipment (including PBXs) in the system when creating a loss plan.

Configuration Parameters

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

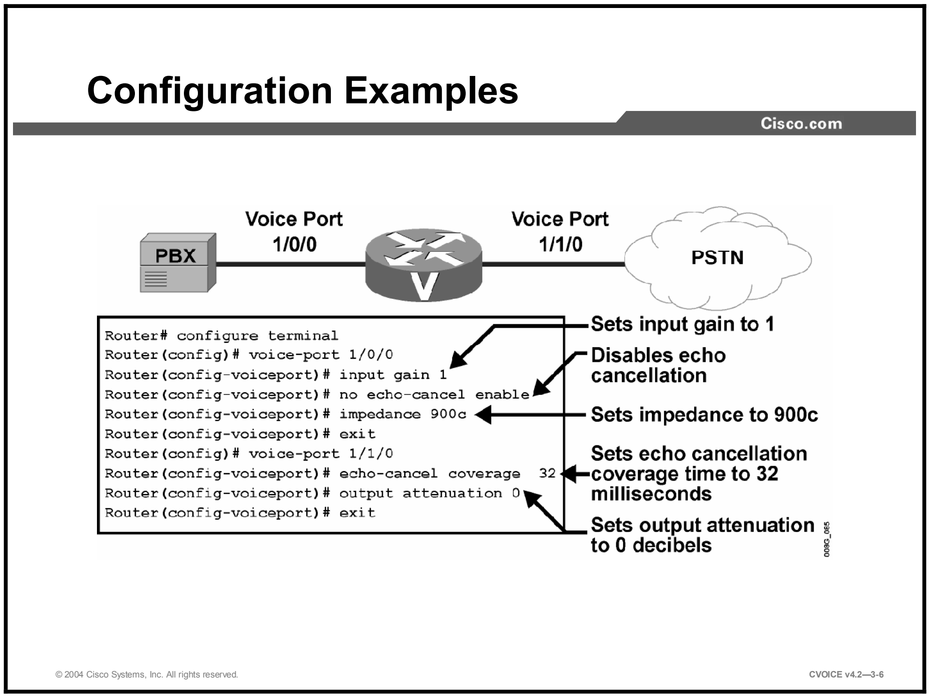
- **input-gain:** Configures a specific input gain, in decibels, to insert into the receiver side of the interface. The default value for this command assumes that a standard transmission loss plan is in effect, meaning that there must be an attenuation of –6 dB between telephones. The standard transmission plan defines country-specific dB levels and assumes that interfaces already provide the expected dB levels; for example, there must be attenuation of –6 dB between two telephones so that the input gain and output attenuation is 0, if the interfaces provide the required –6 dB attenuation.

  The gain of a signal to the PSTN can only be decreased. The gain of a signal coming into the router can be increased.
- **output-attenuation:** Configures the output attenuation value in decibels for the transmit side. The value represents the amount of loss to be inserted at the transmit side of the interface.
- **impedance:** Configures the terminating impedance of a voice port interface. The impedance value that is selected must match the setting from the specific telephony system to which it is connected. You must verify the impedance settings in the technical specifications document of the device. Impedance standards vary between countries. CO switches in the United States are predominantly 600 ohms real (600r). PBXs in the United States are normally 600r or 900 ohms complex (900c).

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

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

Example: Voice Port Tuning



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

E&M voice port parameters include:

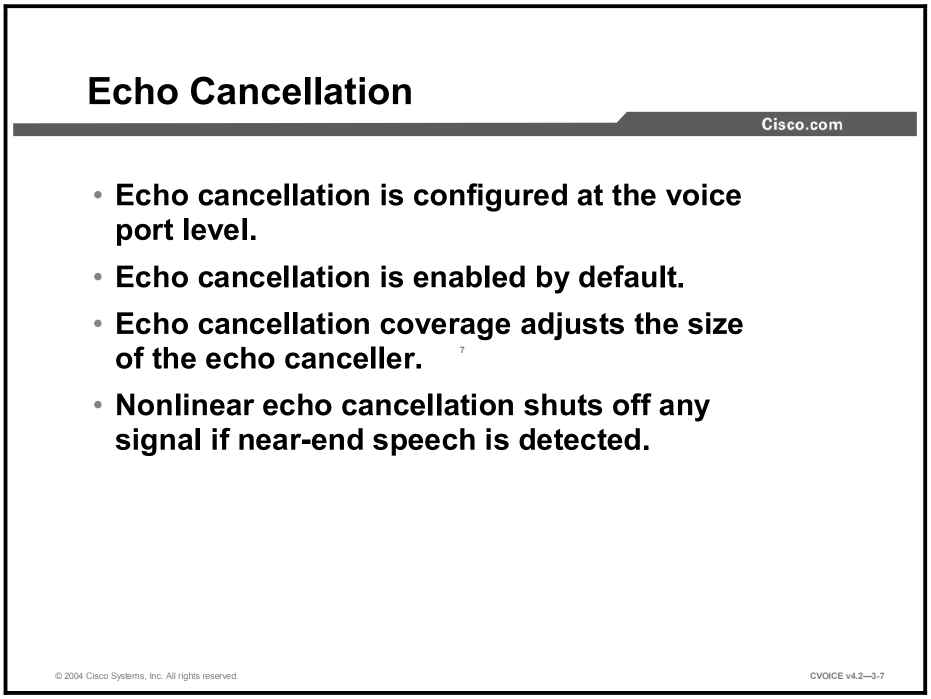
- **input-gain:** Increases the inbound voice level by 1 dB before the voice is transmitted across the network
- **no echo-cancel enable:** Disables echo cancellation
- **impedance:** Sets the impedance to match the connecting hardware

FXO voice port parameters include:

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

Echo Cancellation Commands

This topic describes echo cancellation configuration parameters.



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

- **echo-cancel enable:** Enables cancellation of voice that is sent out through the interface and received back on the same interface. Sound that is received back in this manner is perceived by the listener as echo. Echo cancellation keeps a certain-sized sample of the outbound voice and calculates what that same signal looks like when it returns as an echo. Echo cancellation then attenuates the inbound signal by that amount to cancel the echo signal. If you disable echo cancellation, it will cause the remote side of a connection to hear echo. Because echo cancellation is an invasive process that can minimally degrade voice quality, you should disable this command if it is not needed. There is no echo path for a four-wire E&M interface. The echo canceller should be disabled for this interface type.

Note This command is valid only if the **echo-cancel coverage** command has been configured.

- **echo-cancel coverage:** Adjusts the coverage size of the echo canceller. This command enables cancellation of voice that is sent out through the interface and received back on the same interface within the configured amount of time. If the local loop (the distance from the interface to the connected equipment that is producing the echo) is longer, the configured value of this command should be extended.

  If you configure a longer value for this command, it takes the echo canceller longer to converge; in this case, the user may hear a slight echo when the connection is initially set up. If the configured value for this command is too short, the user may hear some echo for the duration of the call, because the echo canceller is not canceling the longer-delay echoes.

There is no echo or echo cancellation on the network side; for example, the non-POTS side of the connection.

Note This command is valid only if the echo cancel feature has been enabled.

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

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

Caution Do not use the echo cancellation commands or adjust voice quality unless you are experienced in doing so. Arbitrarily adjusting these parameters could adversely affect voice quality.

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

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

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

Example: Echo Suppression Applied

If you speak into your telephone and hear your own voice a short time later, you are experiencing talker echo. Talker echo is caused by the remote telephony circuitry’s two-wire to four-wire hybrid circuit. Enabling echo-cancellation on your voice port will eliminate the problem. Depending on the return time of the echoed voice, you can further adjust using the **echo-cancel coverage** command.

Echo Canceller Comparison

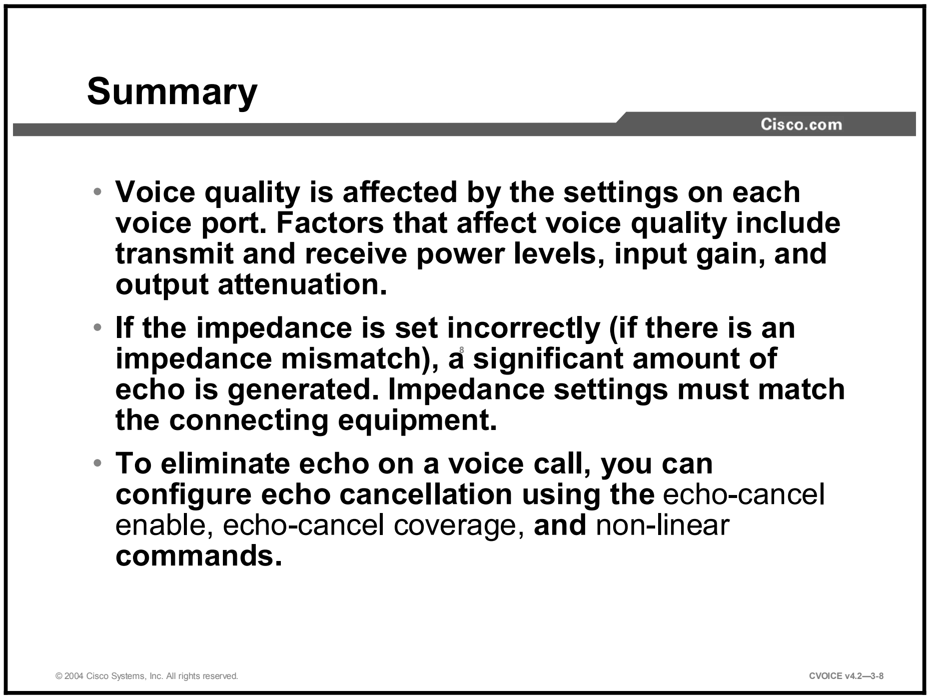
This table contains echo canceller comparison information.

Echo Canceller Comparison

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

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to this resource:

- *Voice Quality Tuning Commands*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/vvfport.htm#56672

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 3-1: Voice Port Configuration

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which factor can cause echo during transmission of voice over an IP network?

A) The input level is too high.

B) The output level is too high at the remote router.

C) The local router voice port input is too low.

D) The input filter is configured improperly.

Q2) At which receive level will analog voice routers operate best?

A) –2 dB

B) –3 dB

C) –9 dB

D) –14 dB

Q3) Which two parameters can be configured on an FXO voice port? (Choose two.)

A) **input-gain**

B) **echo-cancel enable**

C) **impedance**

D) **echo-cancel coverage**

E) **output-attenuation**

Q4) What is the best time to change default voice port configurations?

A) before you set up the network

B) after the network is up and running

C) after two dial peers experience poor quality

D) when there is a network failure

Q5) Which command is used to enable residual echo suppression?

A) **echo-cancel enable**

B) **echo-cancel coverage**

C) **non-linear**

D) **no echo-cancel enable**

Q6) Which of these commands is enabled by default?

A) **echo-cancel enable**

B) **echo-cancel coverage**

C) **non-linear**

D) **no echo-cancel enable**

Q7) If the echo coverage interval is set too long, what is the impact?

A) There may be residual echo.

B) The user may hear a slight echo when the connection is initially set up.

C) The user may hear some echo for the duration of the call.

D) The user may experience clipping at the end of sentences.

Quiz Answer Key

Q1) B

Relates to: Electrical Characteristics

Q2) B

Relates to: Electrical Characteristics

Q3) D, E

Relates to: Voice Quality Tuning

Q4) B

Relates to: Voice Quality Tuning

Q5) C

Relates to: Echo Cancellation Commands

Q6) A

Relates to: Echo Cancellation Commands

Q7) A

Relates to: Echo Cancellation Commands

Module 4

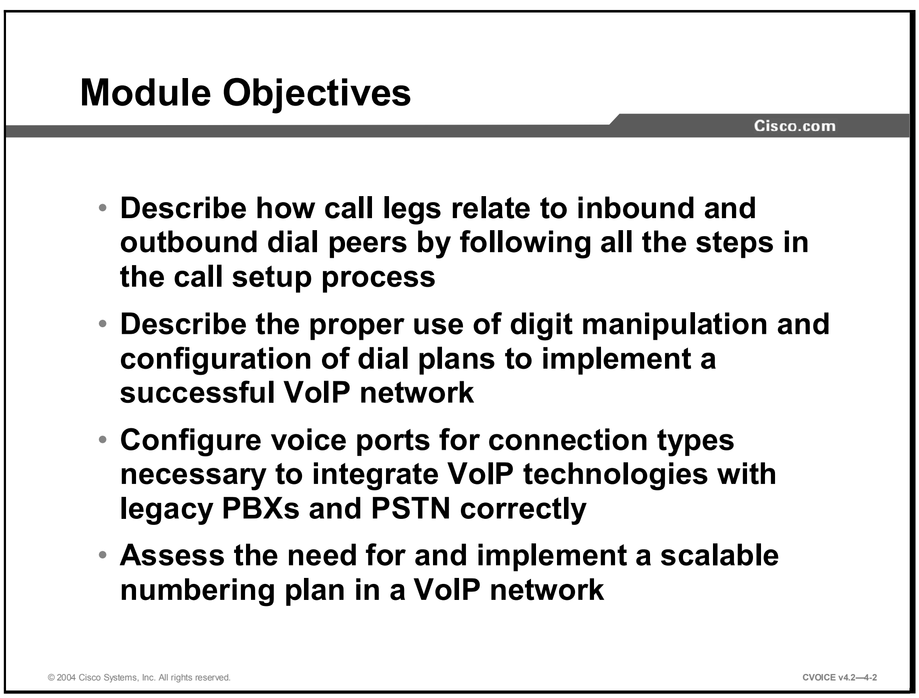
Voice Dial Plans

Overview

Configuring dial peers is the key to setting up dial plans and implementing voice over a packet network. A router may need to manipulate digits in a dial string before it passes the dial string to a telephony device. This module discusses dial-peer configuration, hunt groups, digit manipulation, and special-purpose connections.

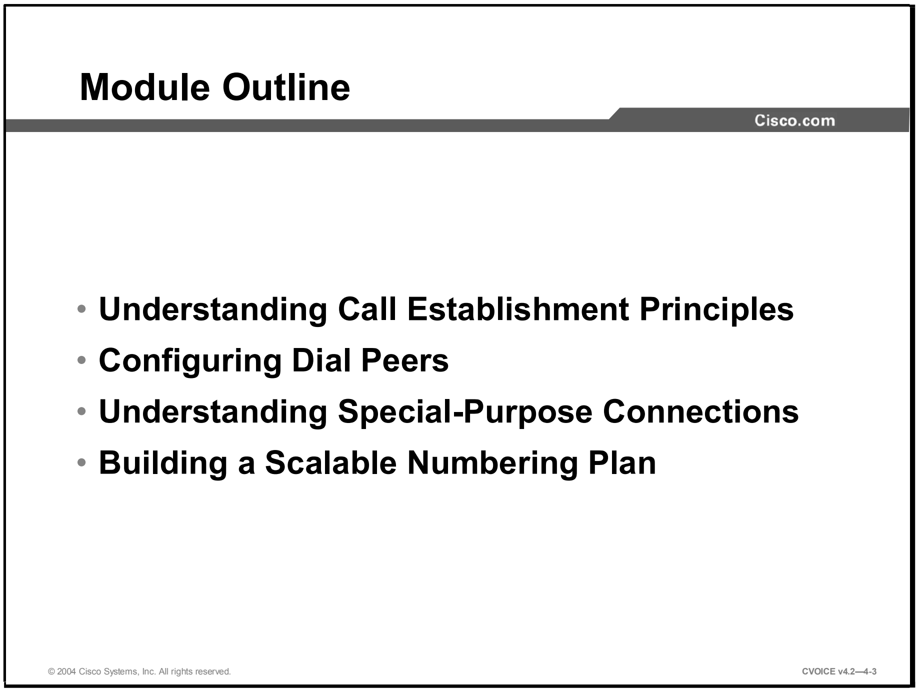
Module Objectives

Upon completing this module, you will be able to configure the call flows for plain old telephone service (POTS), Voice over IP (VoIP), and default dial peers.



Module Outline

The outline lists the components of this module.



Understanding Call Establishment Principles

Overview

This lesson describes call flows as they relate to inbound and outbound dial peers.

Relevance

As a call is set up across the network, the existence of various parameters is checked and negotiated. A mismatch in parameters can cause call failure. It is important to understand how routers interpret call legs and how call legs relate to inbound and outbound dial peers.

Objectives

Upon completing this lesson, you will be able to describe how call legs relate to inbound and outbound dial peers by following all the steps in the call setup process. This includes being able to meet these objectives:

- Describe call legs and their relationships to other components
- Describe how call legs are interpreted by routers to establish end-to-end calls

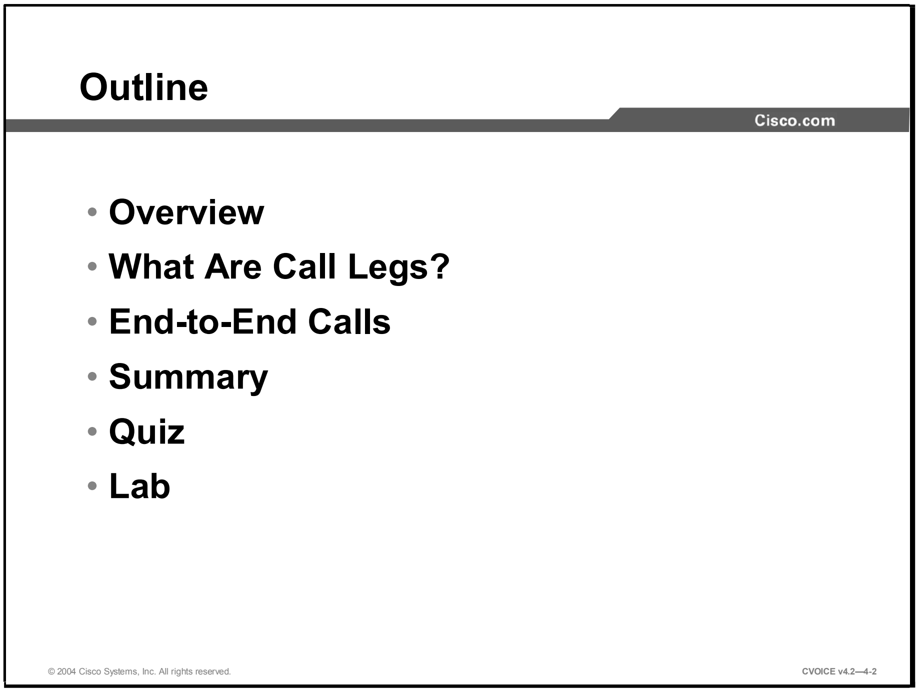
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with Cisco IOS configuration modes
- Familiarity with call-routing concepts and voice ports

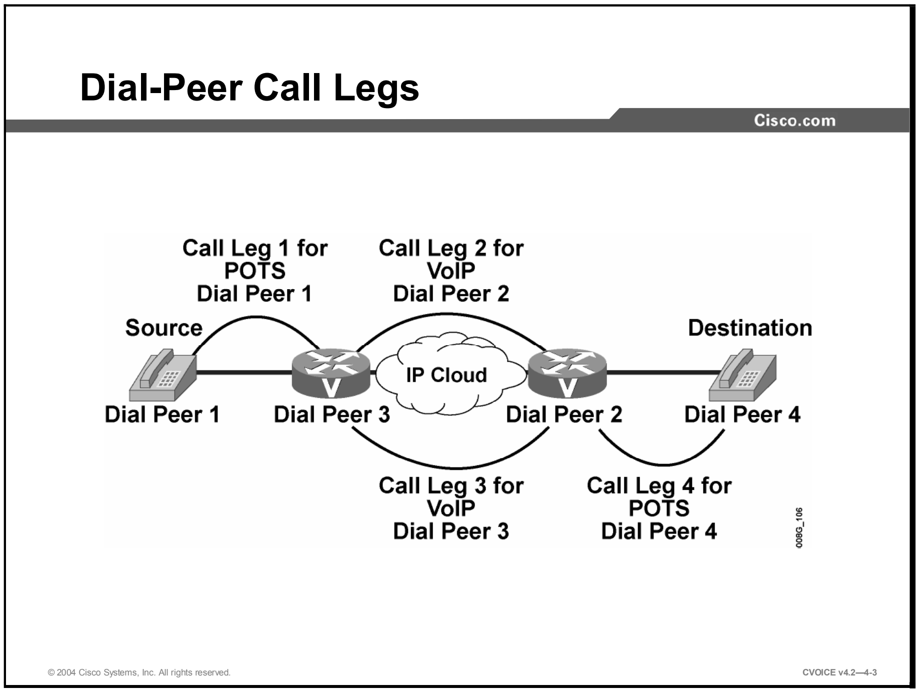
Outline

The outline lists the topics included in this lesson.



What Are Call Legs?

This topic describes call legs and their relationship to other components.



Call legs are logical connections between any two telephony devices, such as gateways, routers, Cisco CallManagers, or telephony endpoint devices.

Call legs are router-centric. When an inbound call arrives, it is processed separately until the destination is determined. Then a second outbound call leg is established, and the inbound call leg is switched to the outbound voice port.

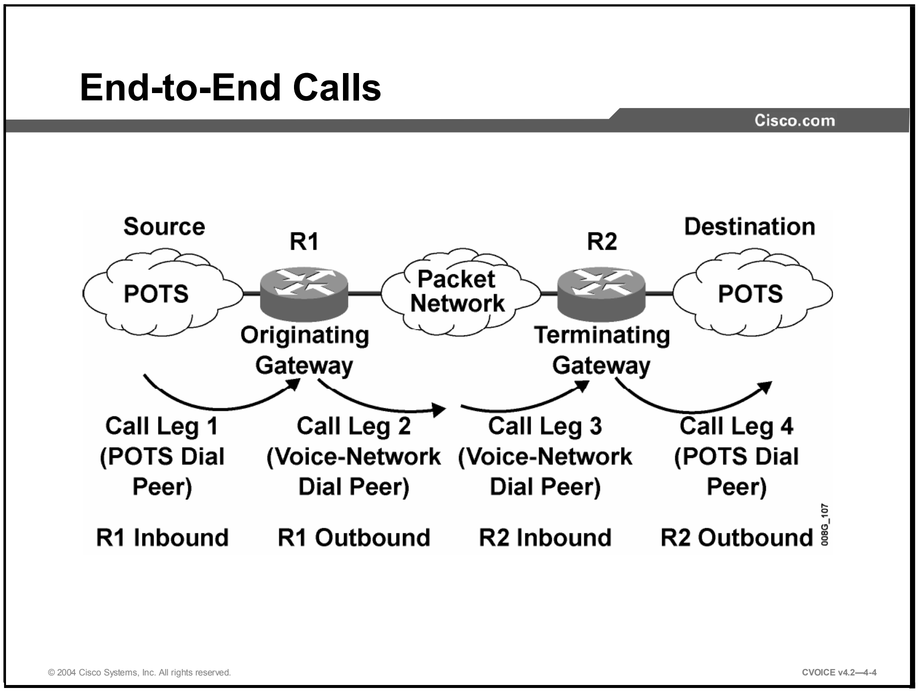
Example: Call Legs Defined

The connections are made when you configure dial peers on each interface. An end-to-end call consists of four call legs: two from the source router perspective (as shown in the figure), and two from the destination router perspective. To complete an end-to-end call from either side and send voice packets back and forth, you must configure all four dial peers.

Dial peers are used only to set up calls. When the call is established, dial peers are no longer used.

End-to-End Calls

This topic explains how routers interpret call legs to establish end-to-end calls.



An end-to-end voice call consists of four call legs: two from the originating router (R1) or gateway perspective, and two from the terminating router (R2) or gateway perspective. An inbound call leg originates when an incoming call comes *into* the router or gateway. An outbound call leg originates when a call is placed *from* the router or gateway.

A call is segmented into call legs and a dial peer is associated with each call leg. The process for call setup is listed below:

1. The plain old telephone service (POTS) call arrives at R1 and an inbound POTS dial peer is matched.
2. After associating the incoming call to an inbound POTS dial peer, R1 creates an inbound POTS call leg and assigns it a Call ID (call leg 1).
3. R1 uses the dialed string to match an outbound voice network dial peer.
4. After associating the dialed string to an outbound voice network dial peer, R1 creates an outbound voice network call leg and assigns it a Call ID (call leg 2).
5. The voice network call request arrives at R2 and an inbound voice network dial peer is matched.
6. After R2 associates the incoming call to an inbound voice network dial peer, R2 creates the inbound voice network call leg and assigns it a Call ID (call leg 3). At this point, both R1 and R2 negotiate voice network capabilities and applications, if required.

When the originating router or gateway requests nondefault capabilities or applications, the terminating router or gateway must match an inbound voice network dial peer that is configured for such capabilities or applications.

7. R2 uses the dialed string to match an outbound POTS dial peer.
8. After associating the incoming call setup with an outbound POTS dial peer, R2 creates an outbound POTS call leg, assigns it a Call ID, and completes the call (call leg 4).

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- **A call is segmented into call legs with a dial peer associated with each call leg.**
- **A call leg is a logical connection between two gateways or routers or between a gateway or router and a telephony endpoint.**
- **An end-to-end call comprises four call legs: two from the voice router perspective, and two from the destination router perspective.**

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References

For additional information, refer to this resource:

- *Understanding Inbound and Outbound Dial Peers on Cisco IOS Platforms*
  http://www.cisco.com/warp/public/788/voip/in_out_dial_peers.html

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 4-1: POTS Dial Peers

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) When an end-to-end call is established, how many inbound call legs are associated with the call?

A) one

B) two

C) three

D) four

Q2) Which dial peers should you configure to complete an end-to-end call?

A) the inbound dial peers only

B) the outbound dial peers only

C) one inbound and one outbound dial peer

D) all four dial peers

Q3) Arrange the steps in the call setup process in the correct order.

_____ 1. Router 2 creates the inbound voice network call leg and assigns it a Call ID.

_____ 2. The POTS call arrives at router 1 and an inbound POTS dial peer is matched.

_____ 3. Router 1 creates an outbound voice network call leg and assigns it a Call ID.

_____ 4. Router 1 creates an inbound POTS call leg and assigns it a Call ID.

_____ 5. The voice network call request arrives at router 2 and an inbound voice network dial peer is matched.

_____ 6. Router 2 creates an outbound POTS call leg and assigns it a Call ID.

_____ 7. Router 2 uses the dialed string to match an outbound POTS dial peer.

_____ 8. At this point, both router 1 and router 2 negotiate voice network capabilities and applications, if required.

_____ 9. Router 1 uses the dialed string to match an outbound voice network dial peer.

Q4) What is the role of the terminating router if the originating router requests nondefault voice capabilities?

A) It negotiates with the originating router until the default capabilities are accepted.

B) It reconfigures the ports to meet the requested capabilities.

C) It matches an inbound voice network dial peer that has the requested capabilities.

D) It terminates the call.

Quiz Answer Key

Q1) B

Relates to: What Are Call Legs?

Q2) D

Relates to: What Are Call Legs?

Q3)

Step 1 The POTS call arrives at router 1, and an inbound POTS dial peer is matched. (2.)

Step 2 Router 1 creates an inbound POTS call leg and assigns it a Call ID. (4.)

Step 3 Router 1 uses the dialed string to match an outbound voice network dial peer. (9.)

Step 4 Router 1 creates an outbound voice network call leg and assigns it a Call ID. (3.)

Step 5 The voice network call request arrives at router 2 and an inbound voice network dial peer is matched. (5.)

Step 6 Router 2 creates the inbound voice network call leg and assigns it a Call ID. (1.)

Step 7 At this point, both router 1 and router 2 negotiate voice network capabilities and applications, if required. (8.)

Step 8 Router 2 uses the dialed string to match an outbound POTS dial peer. (7.)

Step 9 Router 2 creates an outbound POTS call leg and assigns it a Call ID. (6.)

Relates to: End-to-End Calls

Q4) C

Relates to: End-to-End Calls

Configuring Dial Peers

Overview

This lesson describes voice dial peers, hunt groups, digit manipulation, and the matching of calls to dial peers.

Relevance

Successful implementation of a Voice over IP (VoIP) network relies heavily on the proper application of dial peers, the digits they match, and the services they specify. The network engineer must have in-depth knowledge of dial-peer configuration options and their uses. This lesson discusses the proper use of digit manipulation and the configuration of dial peers.

Objectives

Upon completing this lesson, you will be able to describe the proper use of digit manipulation and configuration of dial peers to implement a successful VoIP network. This includes being able to meet these objectives:

- Describe dial peers and their application
- Configure POTS dial peers
- Configure VoIP dial peers
- Describe destination-pattern options and the applicable shortcuts
- Describe the default dial peer
- Describe how the router matches inbound dial peers
- Describe how the router matches outbound dial peers
- List hunt-group commands
- Configure hunt groups
- Describe how the router and the attached telephony equipment collect and consume digits, and apply them to the dial peer
- Describe digit manipulation and the commands that are used to connect to a specified destination

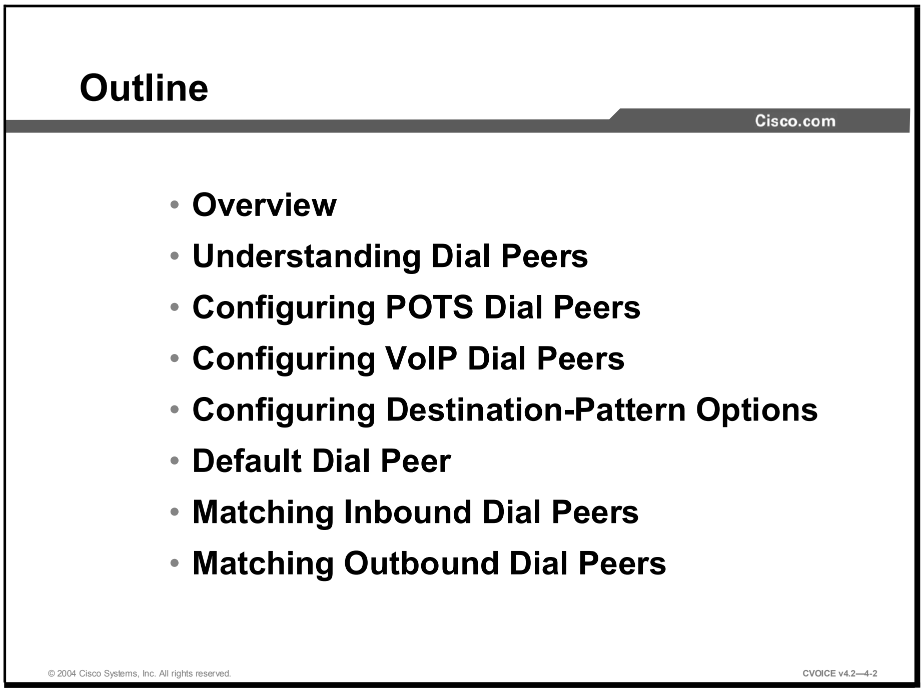
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with Cisco IOS commands
- Familiarity with telephony concepts

Outline

The outline lists the topics included in this lesson.



Outline (Cont.)

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- Hunt-Group Commands
- Configuring Hunt Groups
- Digit Collection and Consumption
- Understanding Digit Manipulation
- Summary
- Quiz
- Lab

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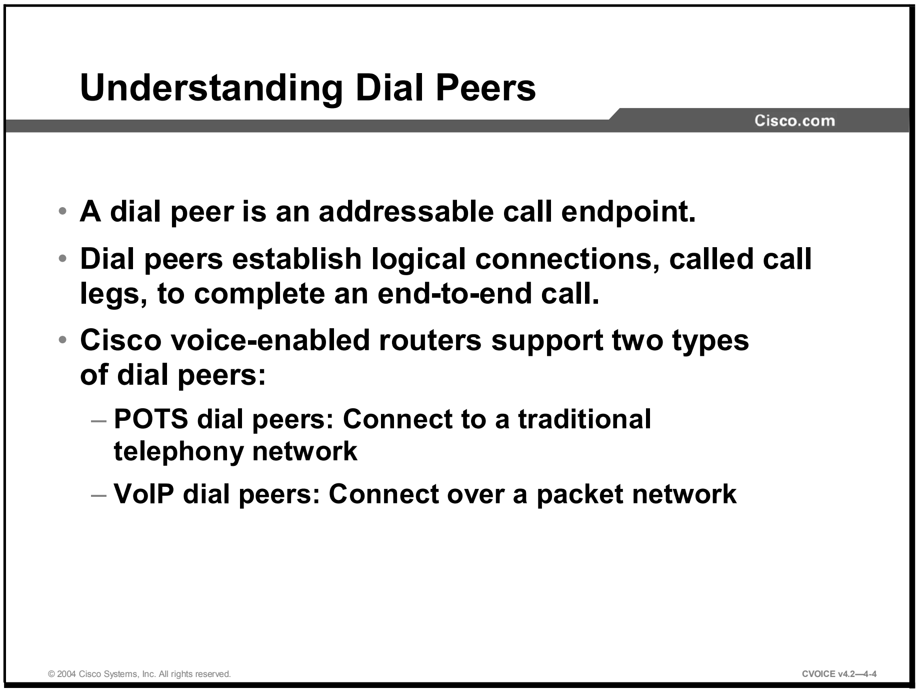
Understanding Dial Peers

This topic describes dial peers and their applications.



When a call is placed, an edge device generates dialed digits as a way of signaling where the call should terminate. When these digits enter a router voice port, the router must have a way to decide whether the call can be routed, and where the call can be sent. The router does this by looking through a list of dial peers.

A dial peer is an addressable call endpoint. The address is called a *destination pattern* and is configured in every dial peer. Destination patterns can point to one telephone number only or to a range of telephone numbers. Destination patterns use both explicit digits and wildcard variables to define a telephone number or range of numbers.

The router uses dial peers to establish logical connections. These logical connections, known as call legs, are established in either an inbound or outbound direction.

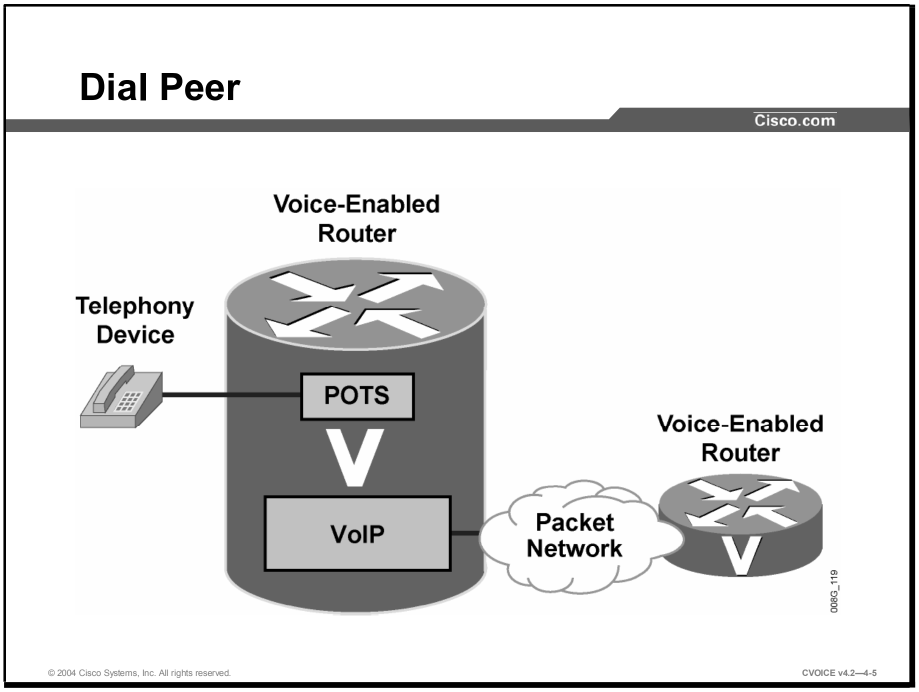
Dial peers define the parameters for the calls that they match; for example, if a call is originating and terminating at the same site, and is not crossing through slow-speed WAN links, then the call can cross the local network uncompressed and without special priority. A call that originates locally and crosses the WAN link to a remote site may require compression with a specific coder-decoder (codec). In addition, this call may require that voice activity detection (VAD) be turned on, and will need to receive preferential treatment by specifying a higher priority level.

Cisco Systems voice-enabled routers support two types of dial peers:

- **POTS dial peers:** Connect to a traditional telephony network, such as the public switched telephone network (PSTN) or a PBX, or to a telephony edge device such as a telephone or fax machine. POTS dial peers perform these functions:
  - Provide an address (telephone number or range of numbers) for the edge network or device
  - Point to the specific voice port that connects the edge network or device
- **VoIP dial peers:** Connect over a packet network. VoIP dial peers perform these functions:
  - Provide a destination address (telephone number or range of numbers) for the edge device that is located across the network
  - Associate the destination address with the next-hop router or destination router, depending on the technology used

Example: Dial-Peer Configuration

This figure shows a dial-peer configuration.



In the figure, the telephony device connects to the Cisco Systems voice-enabled router. The POTS dial-peer configuration includes the telephone number of the telephony device and the voice port to which it is attached. The router knows where to forward incoming calls for that telephone number.

The Cisco voice-enabled router VoIP dial peer is connected to the packet network. The VoIP dial-peer configuration includes the destination telephone number (or range of numbers) and the next-hop or destination voice-enabled router network address.

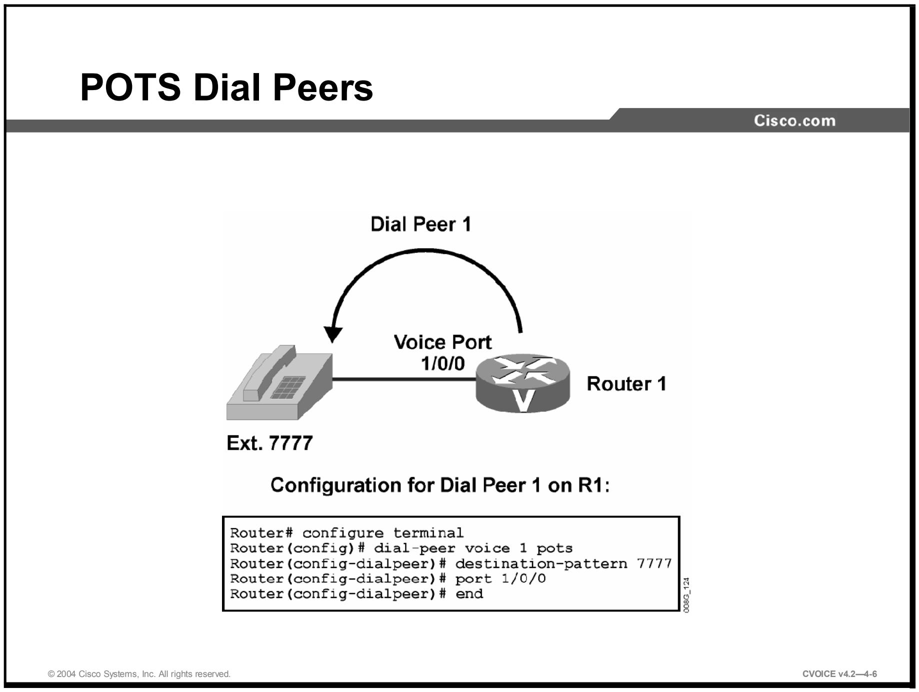
Follow these steps to place a VoIP call:

How to Place a VoIP Call

Step	Action
1.	Configure the source router with a compatible dial peer that specifies the recipient destination address.
2.	Configure the recipient router with a POTS dial peer that specifies which voice port the router uses to forward the voice call.

Configuring POTS Dial Peers

This topic describes how to configure POTS dial peers.



Before the configuration of Cisco IOS dial peers can begin, the user must have a good understanding of where the edge devices reside, what type of connections need to be made between these devices, and what telephone numbering scheme is applied to the devices.

Follow these steps to configure POTS dial peers:

How to Configure POTS Dial Peers

Step	Action
1.	Configure a POTS dial peer at each router or gateway where edge telephony devices connect to the network.
2.	Use the **destination-pattern** command in the dial peer to configure the telephone number.
3.	Use the **port** command to specify the physical voice port that the POTS telephone is connected to.

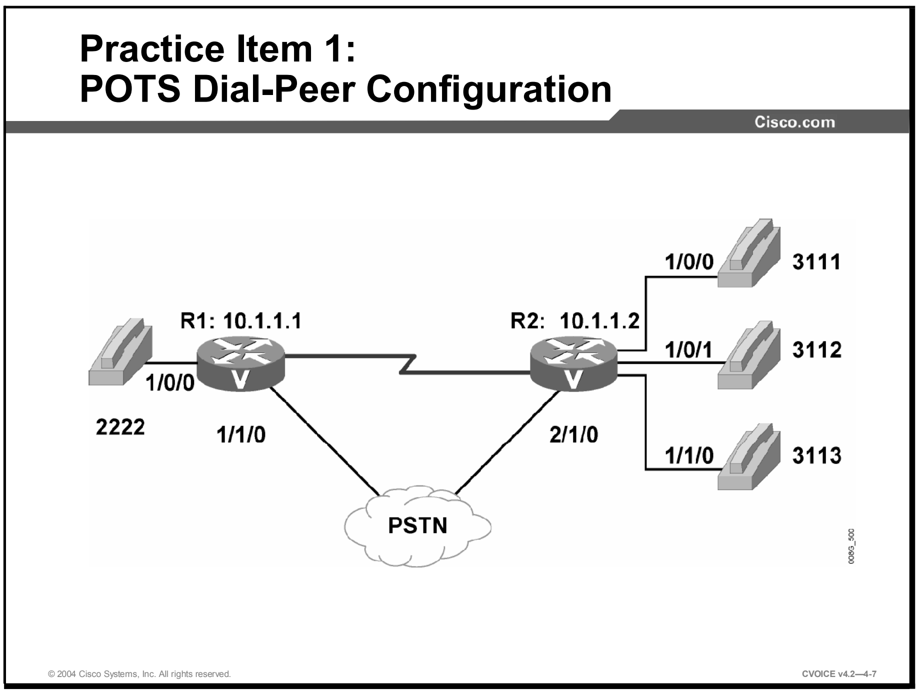
The dial-peer type will be specified as POTS because the edge device is directly connected to a voice port and the signaling must be sent from this port to reach the device. There are two basic parameters that need to be specified for the device: the telephone number and the voice port. When a PBX is connecting to the voice port, a range of telephone numbers can be specified.

Example: POTS Dial-Peer Configuration

The figure illustrates proper POTS dial-peer configuration on a Cisco voice-enabled router. The **dial-peer voice 1 pots** command notifies the router that dial peer 1 is a POTS dial peer with a tag of 1. The **destination-pattern 7777** command notifies the router that the attached telephony device terminates calls destined for telephone number 7777. The **port 1/0/0** command notifies the router that the telephony device is plugged into module 1, voice interface card (VIC) slot 0, voice port 0.

Practice Item 1: POTS Dial-Peer Configuration

Throughout this lesson, you will use practice items to practice what you have learned. In this scenario, assume that there is a data center at the R1 site, and executive offices at the R2 site. Using the diagram, create POTS dial peers for the four telephones shown.



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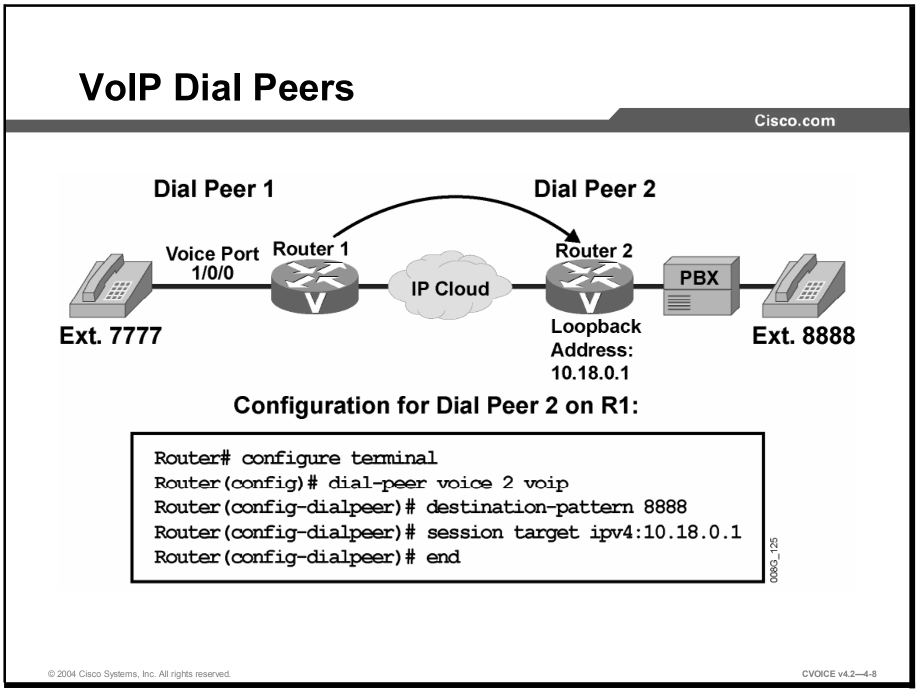
Configuring VoIP Dial Peers

This topic describes how to configure VoIP dial peers.



The administrator must know how to identify the far-end voice-enabled device that will terminate the call. In a small network environment, the device may be the IP address of the remote device. In a large environment, identifying the device may mean pointing to a Cisco CallManager or gatekeeper for address resolution and Call Admission Control (CAC) to complete the call.

You must follow these steps to configure VoIP dial peers:

How to Configure VoIP Dial Peers

Step	Action
1.	Configure the path across the network for voice data.
2.	Specify the dial peer as a VoIP dial peer.
3.	Use the **destination-pattern** command to configure a range of numbers reachable by the remote router or gateway.
4.	Use the **session target** command to specify an IP address of the terminating router or gateway.
5.	Use the remote device loopback address as the IP address.

The dial peer is specified as a VoIP dial peer, which alerts the router that it must process a call according to the various parameters that are specified in the dial peer. The dial peer must then package it as an IP packet for transport across the network. Specified parameters may include the codec used for compression (VAD, for example), or marking the packet for priority service.

The **destination-pattern** parameter configured for this dial peer is typically a range of numbers that are reachable via the remote router or gateway.

Because this dial peer points to a device across the network, the router needs a destination IP address to put in the IP packet. The **session target** parameter allows the administrator to specify either an IP address of the terminating router or gateway, or another device; for example, a gatekeeper or Cisco CallManager that can return an IP address of that remote terminating device.

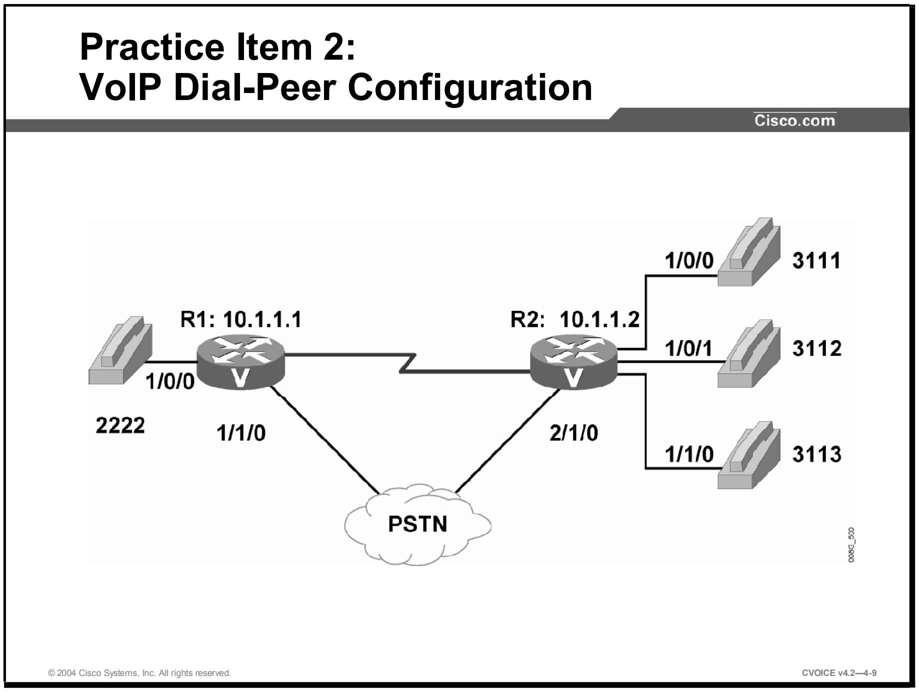
To determine which IP address a dial peer should point to, it is recommended that you use a loopback address. The loopback address is always up on a router, as long as the router is powered on and the interface is not administratively shut down. If an interface IP address is used instead of the loopback, and that interface goes down, the call will fail even if there is an alternate path to the router.

Example: VoIP Dial-Peer Configuration

The figure illustrates the proper VoIP dial-peer configuration on a Cisco voice-enabled router. The **dial-peer voice 2 voip** command notifies the router that dial peer 2 is a VoIP dial peer with a tag of 2. The **destination-pattern 8888** command notifies the router that this dial peer defines an IP voice path across the network for telephone number 8888. The **session target ipv4:10.18.0.1** command defines the IP address of the router that is connected to the remote telephony device.

Practice Item 2: VoIP Dial-Peer Configuration

Using the diagram, create VoIP dial peers for each of the R1 and R2 sites.



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Configuring Destination-Pattern Options

This topic describes destination-pattern options and the applicable shortcuts.

Common Destination-Pattern Options

Cisco.com

Command Syntax: destination-pattern [+] *string* [T]

+	(Optional) Character indicating an E.164 standard number
string	Series of digits that specify the E.164 or private dial-plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: • The *asterisk (*)* and *pound sign (#)* that appear on standard touch-tone dial pads. • *Comma (,)* which inserts a pause between digits. • *Period (.)* which matches any single entered digit (this character is used as a wildcard). • *Brackets ([ ])*, which indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.
T	(Optional) Control character indicating that the destination-pattern value is a variable-length dial string

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The destination pattern associates a telephone number with a given dial peer. The destination pattern also determines the dialed digits that the router collects and forwards to the remote telephony interface, such as a PBX, Cisco CallManager, or the PSTN. You must configure a destination pattern for each POTS and VoIP dial peer that you define on the router.

The destination pattern can indicate a complete telephone number, a partial telephone number with wildcard digits, or it can point to a range of numbers defined in a variety of ways.

Destination-pattern options include the following:

- **Plus sign (+):** An optional character that indicates an E.164 standard number. E.164 is the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) recommendation for the international public telecommunication numbering plan. The plus sign in front of a destination-pattern string specifies that the string must conform to E.164.
- ***string:*** A series of digits specifying the E.164 or private dial-plan telephone number. The examples below show the use of special characters that are often found in destination pattern strings:
  - An asterisk (*) and pound sign (#) appear on standard touch-tone dial pads. These characters may need to be used when passing a call to an automated application that requires these characters to signal the use of a special feature. For example, when calling an interactive voice response (IVR) system that requires a code for access, the number dialed might be “5551212888#”, which would initially dial the telephone number “5551212” and input a code of “888” followed by the pound key to terminate the IVR input query.

— A comma (,) inserts a one-second pause between digits. The comma can be used, for example, where a "9" is dialed to signal a PBX that the call should be processed by the PSTN. The "9" is followed by a comma to give the PBX time to open a call path to the PSTN, after which the remaining digits will be played out. An example of this string is "9,5551212".

— A period (.) matches any single entered digit from 0 to 9, and is used as a wildcard. The wildcard can be used to specify a group of numbers that may be accessible via a single destination router, gateway, PBX, or Cisco CallManager. A pattern of "200." allows for ten uniquely addressed devices, while a pattern of "20.." can point to 100 devices. If one site has the numbers 2000 through 2049, and another site has the numbers 2050 through 2099, then the bracket notation would be more efficient.

— Brackets ([ ]) indicate a range. A range is a sequence of characters that are enclosed in the brackets. Only single numeric characters from 0 to 9 are allowed in the range. In the previous example, the bracket notation could be used to specify exactly which range of numbers is accessible through each dial peer. For example, the first site pattern would be "20[0 – 4].", and the second site pattern would be "20[5-9].". Note that in both cases, a dot is used in the last digit position to represent any single digit from 0 to 9. The bracket notation offers much more flexibility in how numbers can be assigned.

- **T:** An optional control character indicating that the destination-pattern value is a variable-length dial string. In cases where callers may be dialing local, national, or international numbers, the destination pattern must provide for a variable-length dial plan. If a particular voice gateway has access to the PSTN for local calls and access to a transatlantic connection for international calls, then calls being routed to that gateway will have a varying number of dialed digits. A single dial peer with a destination pattern of ".T" could support the different call types. The interdigit timeout determines when a string of dialed digits is complete. The router continues to collect digits until there is an interdigit pause longer than the configured value, which by default is 10 seconds.

When the calling party finishes entering dialed digits, there is a pause equal to the interdigit timeout value *before* the router processes the call. The calling party can immediately terminate the interdigit timeout by entering the pound character (#), which is the default termination character. Because the default interdigit timer is set to 10 seconds, users may experience a long call setup delay.

Note	Cisco IOS software does not check the validity of the E.164 telephone number. It accepts any series of digits as a valid number.

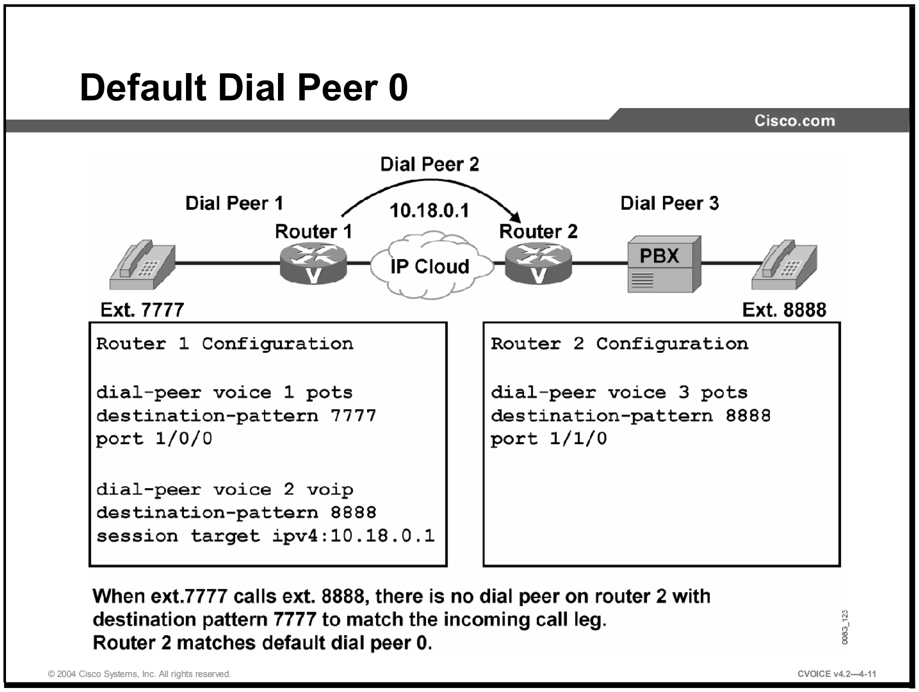
Example: Matching Destination Patterns

Destination-Pattern Options

Destination Pattern	Matching Telephone Numbers
5550124	Matches one telephone number exactly, 5550124. This is typically used when there is a single device, such as a telephone or fax, connected to a voice port.
55501[1-3].	Matches a seven-digit telephone number where the first five digits are 55501, the sixth digit can be a 1, 2, or 3, and the last digit can be any valid digit. This type of destination pattern is used when telephone number ranges are assigned to specific sites. In this example, the destination pattern is used in a small site that does not need more than 30 numbers assigned.
.T	Matches any telephone number that has at least one digit and can vary in length from 1 to 32 digits total. This destination pattern is used for a dial peer that services a variable-length dial plan, such as local, national, and international calls. It can also be used as a default destination pattern so that any calls that do not match a more specific pattern will match this pattern and can be directed to an operator.

Default Dial Peer

This topic describes the default dial peer.



When a matching inbound dial peer is not found, the router resorts to the default dial peer.

Note Default dial peers are used for inbound matches only. They are not used to match outbound calls that do not have a dial peer configured.

The default dial peer is referred to as *dial peer 0*.

Example: Use of Default Dial Peer

In the figure, only one-way dialing is configured. The caller at extension 7777 can call extension 8888 because there is a VoIP dial peer configured on router 1 to route the call across the network. There is no VoIP dial peer configured on router 2 to point calls across the network toward router 1. Therefore, there is no dial peer on router 2 that will match the calling number of extension 7777 on the inbound call leg. If no incoming dial peer matches the calling number, the inbound call leg automatically matches to a default dial peer (POTS or VoIP).

Note There is an exception to the previous statement. Cisco voice and dial platforms, such as the AS53*xx* and AS5800, require that a configured inbound dial peer be matched for incoming POTS calls to be accepted as voice calls. If there is no inbound dial-peer match, the call is treated and processed as a dialup (modem) call.

Dial peer 0 for inbound VoIP peers has the following configuration:

- **any codec**
- **ip precedence 0**
- **vad enabled**
- **no rsvp support**
- **fax-rate service**

Dial peer 0 for inbound POTS peers has the following configuration:

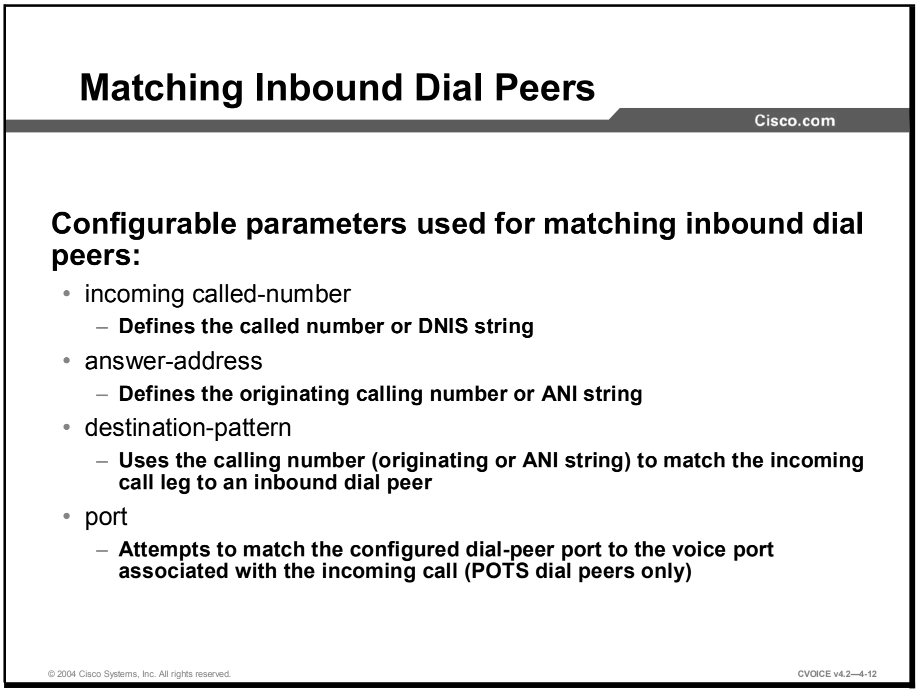
- **no ivr application**

You cannot change the default configuration for dial peer 0. Default dial peer 0 fails to negotiate nondefault capabilities or services. When the default dial peer is matched on a VoIP call, the call leg that is set up in the inbound direction uses any supported codec for voice compression, based on the requested codec capability coming from the source router. When a default dial peer is matched, the voice path in one direction may have different parameters than the voice in the return direction. This may cause one side of the connection to report good-quality voice while the other side reports poor-quality voice; for example, the outbound dial peer has VAD disabled, but the inbound call leg is matched against the default dial peer, which has VAD enabled. In this example, VAD is on in one direction and off in the return direction.

When the default dial peer is matched on an inbound POTS call leg, there is no default IVR application with the port. As a result, the user gets a dial tone and proceeds with dialed digits.

Matching Inbound Dial Peers

This topic describes how the router matches inbound dial peers.



When determining how inbound dial peers are matched on a router, it is important to note whether the inbound call leg is matched to a POTS or VoIP dial peer. Matching occurs in the following manner:

- Inbound POTS dial peers are associated with the incoming POTS call legs of the originating router or gateway.
- Inbound VoIP dial peers are associated with the incoming VoIP call legs of the terminating router or gateway.

Three information elements sent in the call setup message are matched against four configurable dial-peer command attributes.

The table describes the three call setup information elements.

Call Setup Information Elements

Call Setup Element	Description
Called Number Dialed Number Identification Service	This is the call-destination dial string, and it is derived from the ISDN setup message or channel associated signaling Dialed Number Identification Service (DNIS).
Calling Number Automatic Number Identification	This is a number string that represents the origin, and it is derived from the ISDN setup message or channel associated signaling (CAS) automatic number identification (ANI). The ANI is also referred to as the calling line ID.
Voice Port	This represents the POTS physical voice port.

When the Cisco IOS router or gateway receives a call setup request, it makes a dial-peer match for the incoming call. This is not digit-by-digit matching; instead, the router uses the full digit string received in the setup request for matching against the configured dial peers.

The router or gateway matches call setup element parameters in the following order:

How the Router or Gateway Matches Inbound Dial Peers

Step	Action
1.	The router or gateway attempts to match the called number of the call setup request with the configured **incoming called-number** of each dial peer.
2.	If a match is not found, the router or gateway attempts to match the calling number of the call setup request with **answer-address** of each dial peer.
3.	If a match is not found, the router or gateway attempts to match the calling number of the call setup request to the **destination-pattern** of each dial peer.
4.	The voice port uses the voice port number associated with the incoming call setup request to match the inbound call leg to the configured dial peer **port** parameter.
5.	If multiple dial peers have the same port configured, then the router or gateway matches the first dial peer added to the configuration.
6.	If a match is not found in the previous steps, then the default is dial peer 0.

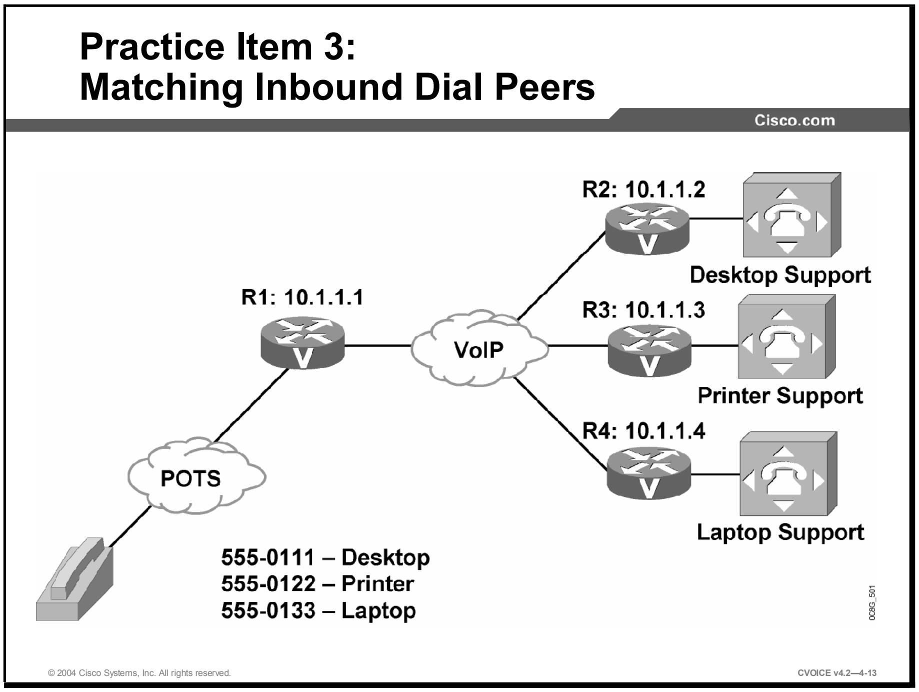
Because call setups always include DNIS information, it is recommended that you use the **incoming called-number** command for inbound dial-peer matching. Configuring **incoming called-number** is useful for a company that has a central call center providing support for a number of different products. Purchasers of each product get a unique 1-800 number to call for support. All support calls are routed to the same trunk group destined for the call center. When a call comes in, the computer telephony system uses the DNIS to flash the appropriate message on the computer screen of the agent to whom the call is routed. The agent will then know how to customize the greeting when answering the call.

The calling number ANI with **answer-address** is useful when you want to match calls based on the originating calling number. For example, when a company has international customers who require foreign-language-speaking agents to answer the call, the call can be routed to the appropriate agent based on the country of call origin.

You must use the calling number ANI with **destination-pattern** when the dial peers are set up for two-way calling. In a corporate environment, the head office and the remote sites must be connected. As long as each site has a VoIP dial peer configured to point to each site, inbound calls from the remote site will match against that dial peer.

Practice Item 3: Matching Inbound Dial Peers

In this practice item, assume that you are setting up a technical support center for desktop PCs, printers, and laptops. Customers who dial specific numbers reach the appropriate technical support staff. Using the diagram, create dial peers on R1 to route incoming calls per the incoming called number to the appropriate site.



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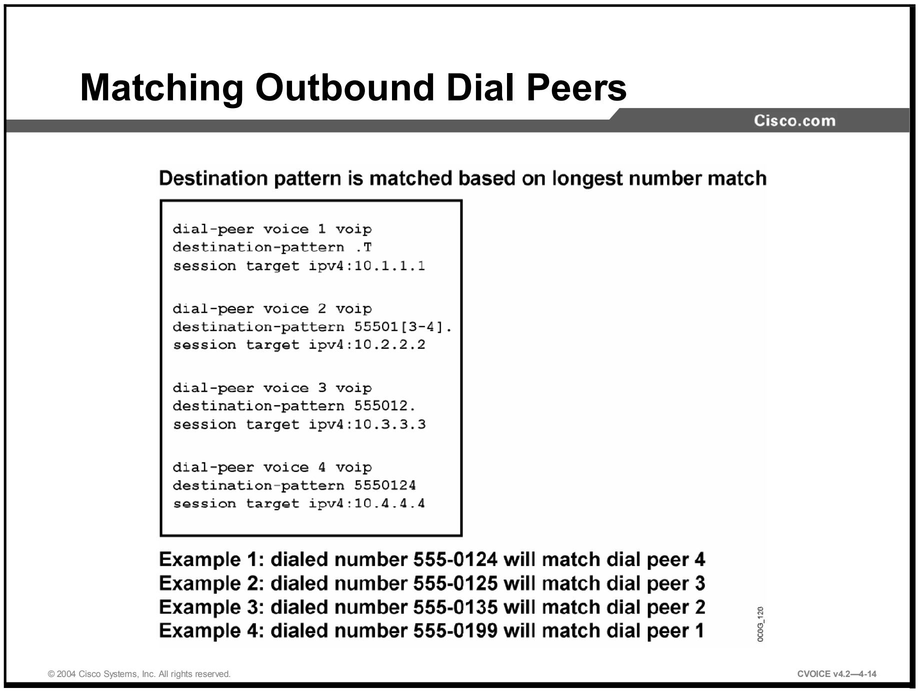
Matching Outbound Dial Peers

This topic describes how the router matches outbound dial peers.



Outbound dial-peer matching is completed on a digit-by-digit basis. Therefore, the router or gateway checks for dial-peer matches after receiving each digit and then routes the call when a full match is made.

The router or gateway matches outbound dial peers in the following order:

How the Router or Gateway Matches Outbound Dial Peers

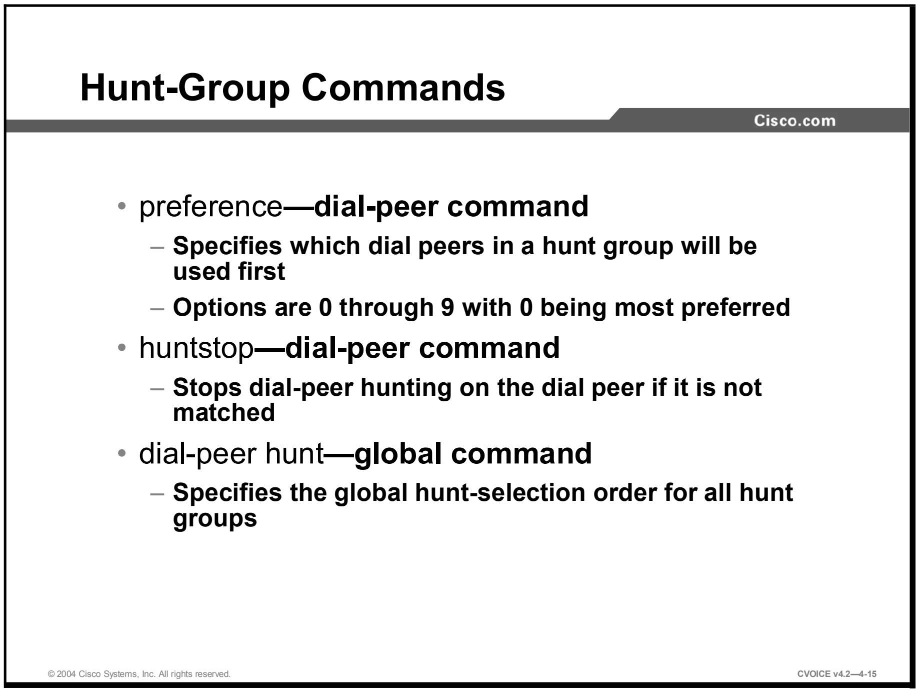
Step	Action
1.	The router or gateway uses the dial peer **destination-pattern** command to determine how to route the call.
2.	The **destination-pattern** command routes the call in the following manner: ■ On POTS dial peers, the **port** command forwards the call. ■ On VoIP dial peers, the **session target** command forwards the call.
3.	Use the **show dialplan number *string*** command to determine which dial peer is matched to a specific dialed string. This command displays all matching dial peers in the order that they are used.

Example: Matching Outbound Dial Peers

In the figure, dial peer 1 matches any digit string that has not matched other dial peers more specifically. Dial peer 2 matches any seven-digit number in the 30 and 40 range of numbers starting with 55501. Dial peer 3 matches any seven-digit number in the 20 range of numbers starting with 55501. Dial peer 4 matches the specific number 5550124 only. When the number 5550124 is dialed, dial peers 1, 3, and 4 all match that number, but dial peer 4 places that call because it has the most specific destination pattern.

Hunt-Group Commands

This topic describes hunt-group commands.



Cisco voice-enabled routers support the concept of *hunt groups*, sometimes called *rotary groups,* in which multiple dial peers are configured with the same destination pattern. Because the destination of each POTS dial peer is a single voice port to a telephony interface, hunt groups help ensure that calls get through even when a specific voice port is busy. If the router is configured to hunt, it can forward a call to another voice port when one voice port is busy.

The following is a list of hunt-group commands:

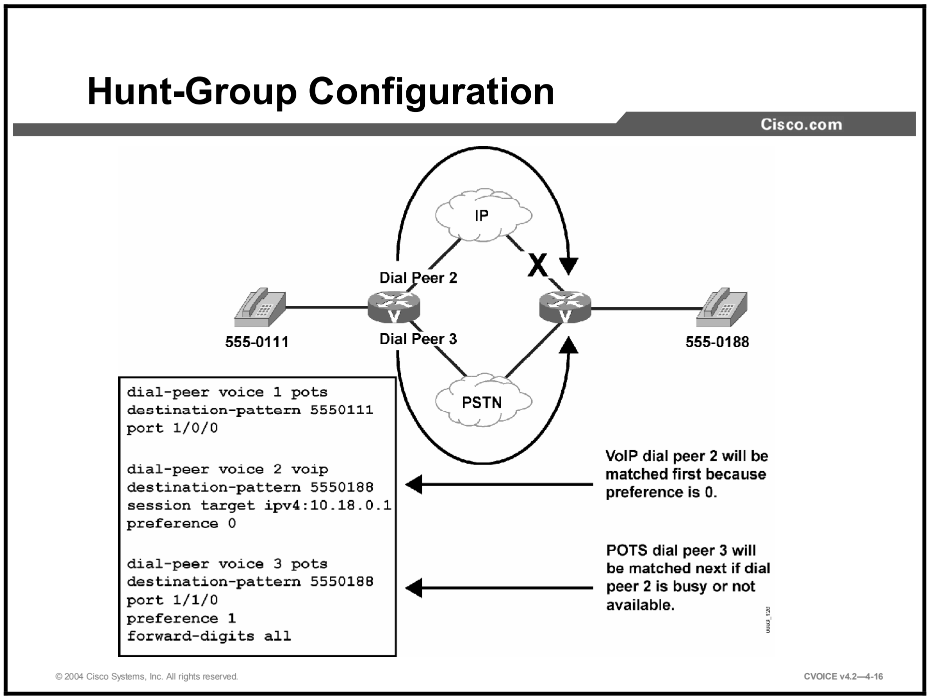
- **preference:** Sets priority for dial peers. The destination with the lowest setting has the highest priority.
- **huntstop:** Disables dial-peer hunting on the dial peer.
- **dial-peer hunt:** Changes the default selection order for hunting through dial peers.

You can also use the following command to view dial-peer hunt current settings:

- **show dial-peer voice summary**: Shows the current settings for dial-peer hunt.

Configuring Hunt Groups

This topic describes how to configure hunt groups.



In some business environments, such as call centers or sales departments, there may be a group of agents available to answer calls coming in to a single number. Scenario 1 may randomly distribute the calls between all agents. Scenario 2 may send calls to the senior agents first, and send calls to the junior agents only when all senior agents are busy. Both of these scenarios can be serviced by configuring a hunt group with specific commands to control the hunt actions.

Follow these steps to configure hunt groups:

How to Configure Hunt Groups

Step	Action
1.	Configure the same destination pattern across multiple dial peers.
2.	The destination pattern matches the greatest number of dialed digits.
3.	Use the **preference** command if the destination pattern of the dial peer is the same for several dial peers.
4.	If the preference does not act as the tiebreaker, then the router picks the matching dial peer randomly.

You must use the **dial-peer hunt** global configuration command to change the default selection order of the procedure or to choose different methods for hunting through dial peers. To view the current setting for **dial-peer hunt**, use the **show dial-peer voice summary** command.

If the desired action is not hunting through a range of dial peers, the **huntstop** command disables dial-peer hunting on the dial peer. After you enter this command, no further hunting is allowed if a call fails on the selected dial peer. This is useful in situations where it is undesirable to hunt to a less-specific dial peer if the more specific call fails; for example, if a call is destined for a particular staff member and the person is on the phone, the router searches for any other dial peer that may match the dialed number. If there is a more generic destination pattern in another dial peer that also matches, the call is routed to the generic destination pattern. If this is not the desired action, then configuring the **huntstop** command in the more specific dial peer will send the caller a busy signal.

You can mix POTS and VoIP dial peers when creating hunt groups. This is useful if you want incoming calls sent over the packet network; however, if that network connectivity fails, you want to reroute the calls back through the PBX, or through the router, to the PSTN.

By default, the router selects dial peers in a hunt group according to the following criteria, in the order listed:

How the Router Selects Dial Peers in a Hunt Group

Step	Action
1.	The router matches the most specific telephone number.
2.	The router matches according to the preference setting.
3.	The router matches randomly.

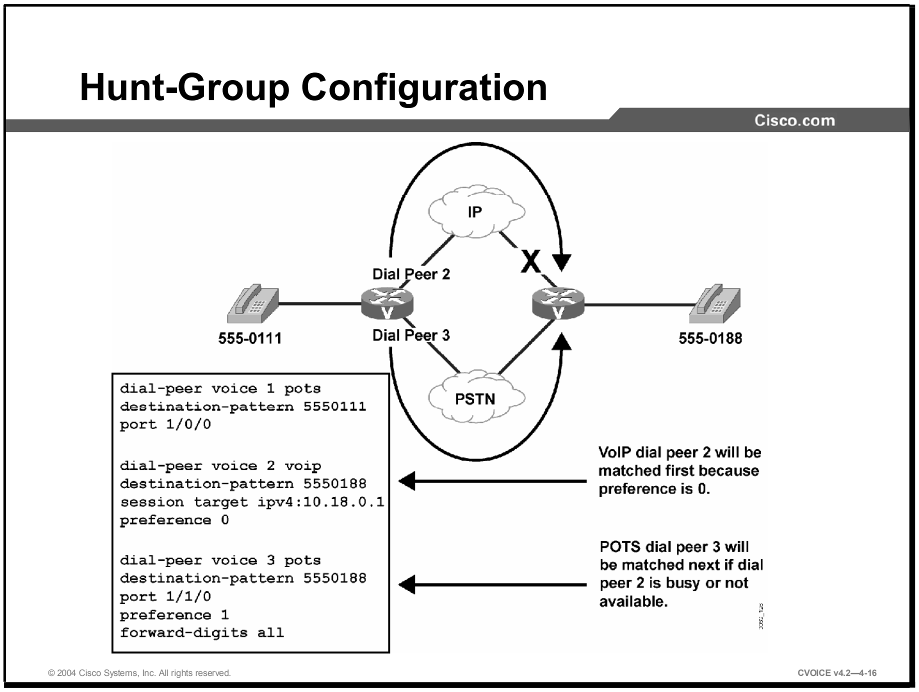
The destination pattern that matches the greatest number of dialed digits is the first dial peer selected by the router. For example, if one dial peer is configured with a dial string of "345…." and a second dial peer is configured with "3456789", the router selects "3456789" first because it has the longest explicit match of the two dial peers. Without a PBX, if the line is currently in use, the desired action is to send a call to a voice-mail system or a secretary, instead of giving the caller a busy signal.

If the destination pattern is the same for several dial peers, you can configure the priority by using the **preference** dial-peer command. You would use the **preference** command to configure service for scenario 2, where the dial peers connecting to the senior agents would have the preference 0 and the dial peers connecting to the junior agents would have the preference 1. The lower the preference setting, the higher the priority for that dial peer to handle the call.

If all destination patterns are equal, by default, the preference is set to 0 on all dial peers. If the preference does not act as the tiebreaker, then a dial peer matching the called number will be picked randomly. This configuration would service scenario 1.

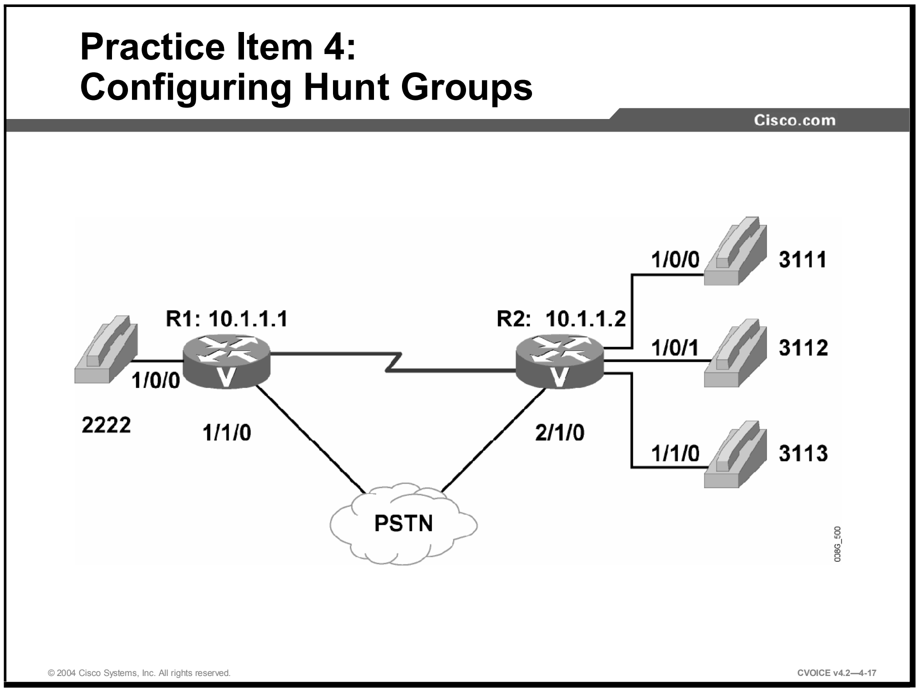
Example: Hunt-Group Application

The figure shows an example of configuring a hunt group to send calls to the PSTN if the IP network fails. For all calls going to 555-0188, VoIP dial peer 2 is matched first because the preference is set to zero. If the path through the IP network fails, POTS dial peer 3 is matched and the call is forwarded through the PSTN. The **forward-digits** command forwards all digits to the PSTN to automatically complete the call without a secondary dial tone.



Practice Item 4: Configuring Hunt Groups

Using the diagram, configure a hunt group using the preference command on R2 such that if extension 3111 is busy, the call rings extension 3112. Assume that you have POTS dial peers for all three extensions already configured.



R2: Already configured

```
dial-peer voice 1 pots
destination-pattern 3111
port 1/0/0

dial-peer voice 2 pots
destination-pattern 3112
port 1/0/1

dial-peer voice 3 pots
destination-pattern 3113
port 1/1/0
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R2: Hunt group dial peer

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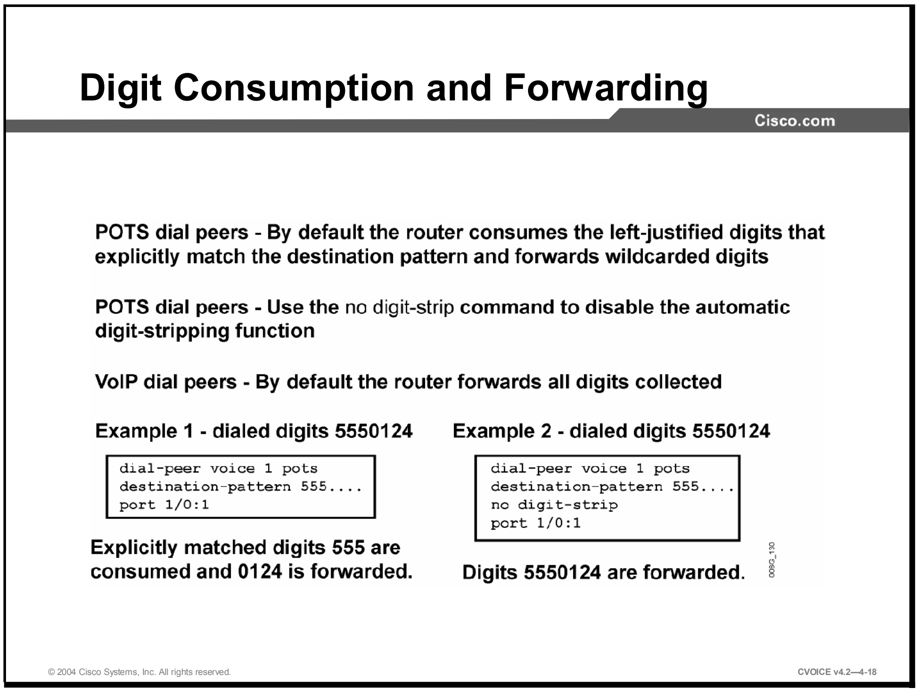
Digit Collection and Consumption

This topic describes how the router collects and consumes digits and applies them to the dial-peer statements.

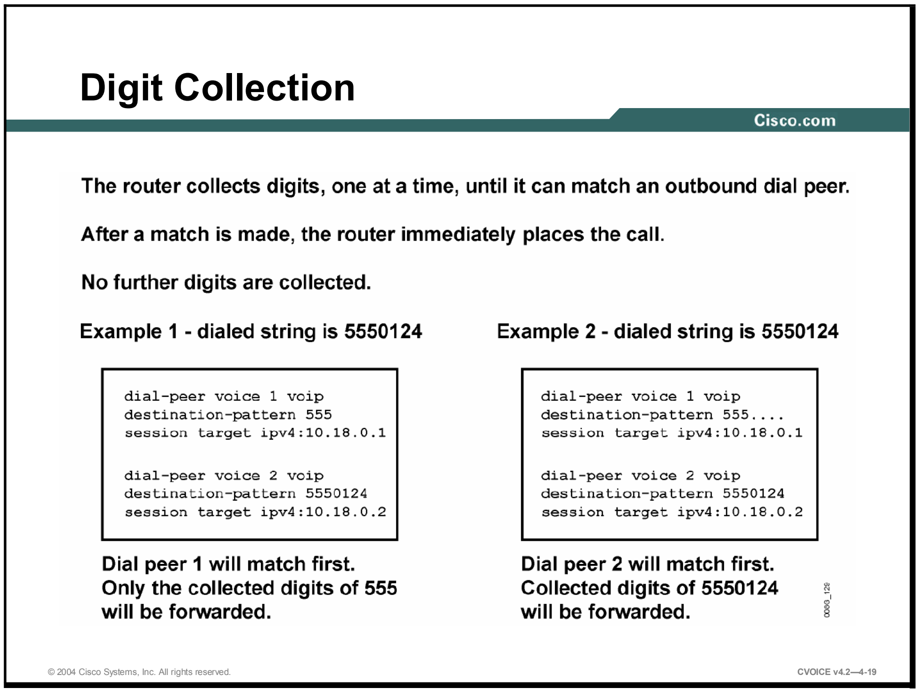


Use the **no digit-strip** command to disable the automatic digit-stripping function. This allows the router to match digits and pass them to the telephony interface.

By default, when the terminating router matches a dial string to an outbound POTS dial peer, the router strips off the left-justified digits that explicitly match the destination pattern. The remaining digits, or *wildcard digits,* are forwarded to the telephony interface, which connects devices such as a PBX or the PSTN.

Digit stripping is the desired action in some situations. There is no need to forward digits out of a POTS dial peer if it is pointing to a Foreign Exchange Station (FXS) port that connects a telephone or fax machine. If digit stripping is turned off on this type of port, the user may hear tones after answering the call because any unconsumed and unmatched digits are passed through the voice path after the call is answered.

In other situations, where a PBX or the PSTN is connected through the POTS dial peer, digit stripping is not desired because these devices need additional digits to further direct the call. In these situations, the administrator must assess the number of digits that need to be forwarded for the remote device to correctly process the call. With a VoIP dial peer, all digits are passed across the network to the terminating voice-enabled router.



When a voice call enters the network, the router collects digits as follows:

How the Router Collects Digits

Step	Action
1.	The originating router collects dialed digits until it matches an outbound dial peer.
2.	The router immediately places the call and forwards the associated dial string.
3.	The router collects no additional dialed digits.

Example: Digit Collection

The figure demonstrates the impact that overlapping destination patterns have on the call-routing decision. In example 1, the destination pattern in dial peer 1 is a subset of the destination pattern in dial peer 2. Because the router matches one digit at a time against available dial peers, an exact match will always occur on dial peer 1, and dial peer 2 will never be matched.

In example 2, the length of the destination patterns in both dial peers is the same. Dial peer 2 has a more specific value than dial peer 1, so it will be matched first. If the path to IP address 10.18.0.2 is unavailable, dial peer 1 will be used.

Destination patterns are matched based on the longest explicit number match. Digits collected are dependant on the configured destination pattern. The table describes how different number combinations are matched and collected.

Matching Destination Patterns

Dialed Digits	Destination Pattern	Dialed Digits Collected
5550124	5……	5550124
5550124	555….	5550124
5550124	555	555
5550124	555T	5550124

In the first row of the table, the destination pattern specifies a seven-digit string. The first digit must be a five, and the remaining six digits can be any valid digits. All seven digits must be entered before the destination pattern is matched.

In the second row, the destination pattern specifies a seven-digit string. The first three digits must be 555, and the remaining four digits can be any valid digits. All seven digits must be entered before the destination pattern is matched.

In the third row, the destination pattern specifies a three-digit string. The dialed digits must be exactly 555. When the user begins to dial the seven-digit number, the destination pattern matches after the first three digits are entered. The router then stops collecting digits and places the call. If the call is set up quickly, the answering party at the other end may hear the remaining four digits as the user finishes dialing the string. After a call is set up, any dual tone multifrequency (DTMF) tones are sent through the voice path and played out at the other end.

In the last row, the destination pattern specifies a variable-length digit string that is at least three digits long. The first three digits must be exactly 555, and the remaining digits can be any valid digits. The "T" tells the router to continue collecting digits until the interdigit timer expires. The router stops collecting digits when the timer expires, or when the user presses the pound (#) key.

Understanding Digit Manipulation

This topic describes digit manipulation and the commands that are used to connect to a specified destination.

Digit Manipulation Commands

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- prefix
  - Dial-peer command
  - Adds digits to the front of the dial string before it is forwarded to the telephony interface
- forward-digits
  - Dial-peer command
  - Controls the number of digits forwarded to the telephony interface
- num-exp
  - Global command
  - Expands an extension into a full telephone number or replaces one number with another
- translation-rule
  - Global and dial-peer command
  - Digit translation rules used to manipulate the calling number digits, or ANI, or the called number digits, or DNIS, for a voice call

 CVOICE v4.2—4-20

Digit manipulation is the task of adding or subtracting digits from the original dialed number to accommodate user dialing habits or gateway needs. The digits can be manipulated before matching an inbound or outbound dial peer. The following is a list of digit manipulation commands and their uses:

- **prefix:** This dial-peer command adds digits to the front of the dial string before it is forwarded to the telephony interface. This occurs after the outbound dial peer is matched, but before digits get sent out of the telephony interface. Use the **prefix** command when the dialed digits leaving the router must be changed from the dialed digits that had originally matched the dial peer; for example, a call is dialed using a four-digit extension such as 0123, but the call needs to be routed to the PSTN, which requires ten-digit dialing. If the four-digit extension matches the last four digits of the actual PSTN telephone number, then you can use the **prefix 902555** command to prepend the six additional digits needed for the PSTN to route the call to 902-555-0123. After the POTS dial peer is matched with the destination pattern of 0123, the **prefix** command prepends the additional digits, and the string “9025550123” is sent out of the voice port to the PSTN.
- **forward-digits:** This dial-peer command specifies the number of digits that must be forwarded to the telephony interface, regardless of whether they are explicitly matched or wildcard matched. This command occurs after the outbound dial peer is matched, but before the digits are sent out of the telephony interface. When a specific number of digits are configured for forwarding, the count is right justified. For example, if the POTS dial peer has a destination pattern configured to match all extensions in the 1000 range (**destination-pattern 1…**), by default, only the last three digits are forwarded to the PBX that is connected to the specified voice port. If the PBX needs all four digits to route the call, you must use the command **forward-digits 4**, or **forward-digits all**, so that the appropriate number of digits are forwarded. To restore the **forward-digits** command to its

default setting, use the **default forward-digits** command. Using the **no forward-digits** command specifies that no digits are to be forwarded.

- **num-exp:** The **num-exp** global command expands an extension into a full telephone number or replaces one number with another. The number expansion table manipulates the called number. This command occurs before the outbound dial peer is matched; therefore, you must configure a dial peer with the expanded number in the destination pattern for the call to go through. The number expansion table is useful, for example, where the PSTN changes the dialing requirements from seven-digit dialing to ten-digit dialing. In this scenario, you can do one of the following:
  - Make all the users dial all ten digits to match the new POTS dial peer that is pointing to the PSTN.
  - Allow the users to continue dialing the seven-digit number as they have before, but expand the number to include the area code before the ten-digit outbound dial peer is matched.

Note You must use the **show num-exp** command to view the configured number-expansion table. You must use the **show dialplan number** *number* command to confirm the presence of a valid dial peer to match the newly expanded number.

- **translation-rule:** Digit translation is a two-step configuration process. First, the translation rule is defined at the global level. Then, the rule is applied at the dial-peer level either as inbound or outbound translation on either the called or calling number. Translation rules manipulate the ANI or DNIS digits for a voice call. Translation rules convert a telephone number into a different number before the call is matched to an inbound dial peer, or before the outbound dial peer forwards the call; for example, an employee may dial a five-digit extension to reach another employee of the same company at another site. If the call is routed through the PSTN to reach the other site, the originating gateway may use translation rules to convert the five-digit extension into the ten-digit format that is recognized by the central office (CO) switch.

You can also use translation rules to change the numbering type for a call. For example, some gateways may tag a number with more than 11 digits as an international number, even when the user must dial “9” to reach an outside line. In this case, the number that is tagged as an international number needs to be translated into a national number—without the 9—before it is sent to the PSTN.

As illustrated in this topic, there are numerous ways to manipulate digits at various stages of call completion. The administrator needs to determine which command will be most suitable and the requirements that are necessary for manipulation.

Note To test configured translation rules, you must use the **test translation** command.

Example: Using Digit Manipulation Tools

The following is a sample configuration using the **prefix** command:

```
dial-peer voice 1 pots
destination-pattern 555....
prefix 555
port 1/0/0
```

In the sample configuration using the **prefix** command, the device attached to port 1/0/0 needs all seven digits to process the call. On a POTS dial peer, only wildcard-matched digits are forwarded by default. Use the **prefix** command to send the prefix numbers 555 before forwarding the four wildcard-matched digits.

The following is a sample configuration using the **forward-digits** command:

```
dial-peer voice 1 pots
destination-pattern 555....
forward-digits 7
port 1/0/0
```

In the sample configuration using the **forward-digits** command, the device attached to port 1/0/0 needs all seven digits to process the call. On a POTS dial peer, only wildcard-matched digits are forwarded by default. The **forward-digits** command allows the user to specify the total number of digits to forward.

The following is a sample configuration using the number expansion table (**num-exp**) command:

```
num-exp 2... 5552...
dial-peer voice 1 pots
destination-pattern 5552...
port 1/1/0
```

In the sample configuration using the **num-exp** command, the extension number 2… is expanded to 5552… before an outbound dial peer is matched; for example, the user dials 2401, but the outbound dial peer 1 is configured to match 5552401.

The following is a sample configuration using the digit translation (**translation-rule**) command:

```
translation-rule 5
rule 1 2401 5552401
dial-peer voice 1 pots
translate-outgoing called-number 5
```

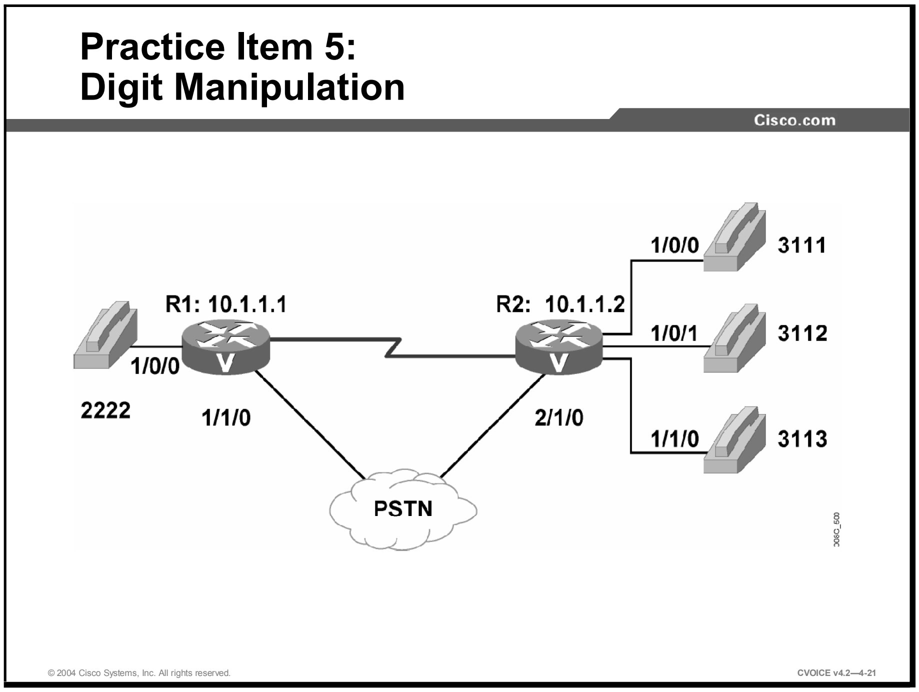
In the sample configuration using the **translation-rule** command, the rule is defined to translate 2401 into 5552401. The dial peer **translate-outgoing called-number 5** command notifies the router to use the globally defined translation rule 5 to translate the number before sending the string out the port. It is applied as an outbound translation from the POTS dial peer.

The following example shows a translation rule that converts any called number that starts with 91 and is tagged as an international number into a national number without the 9 before sending it to the PSTN.

```
translation-rule 20
rule 1 91 1 international national
!
!
dial-peer voice 10 pots
destination-pattern 91..........
translate-outgoing called 20
port 1/1:5
forward-digits all
```


Practice Item 5: Digit Manipulation

Assuming that all POTS and VoIP dial peers are configured, create a dial peer to divert calls from R1 to R2 across the PSTN in the event of failure of the VoIP network. Assume that digits must be forwarded to the PSTN, and a prefix of 555 is necessary.



R1

____

____

____

____

____

____

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- A dial peer is an addressable endpoint.
- Cisco voice-enabled routers support POTS dial peers and VoIP dial peers.
- Basic POTS dial-peer configuration consists of defining the dial peer with a tag number and POTS designation, defining the destination-pattern, and defining the voice port to which the device is connected.
- Basic VoIP dial-peer configuration consists of defining the dial peer with a tag number and VoIP designation, defining the destination pattern, and defining the remote voice-enabled router through the session target command.

 CVOICE v4.2—4-22

Summary (Cont.)

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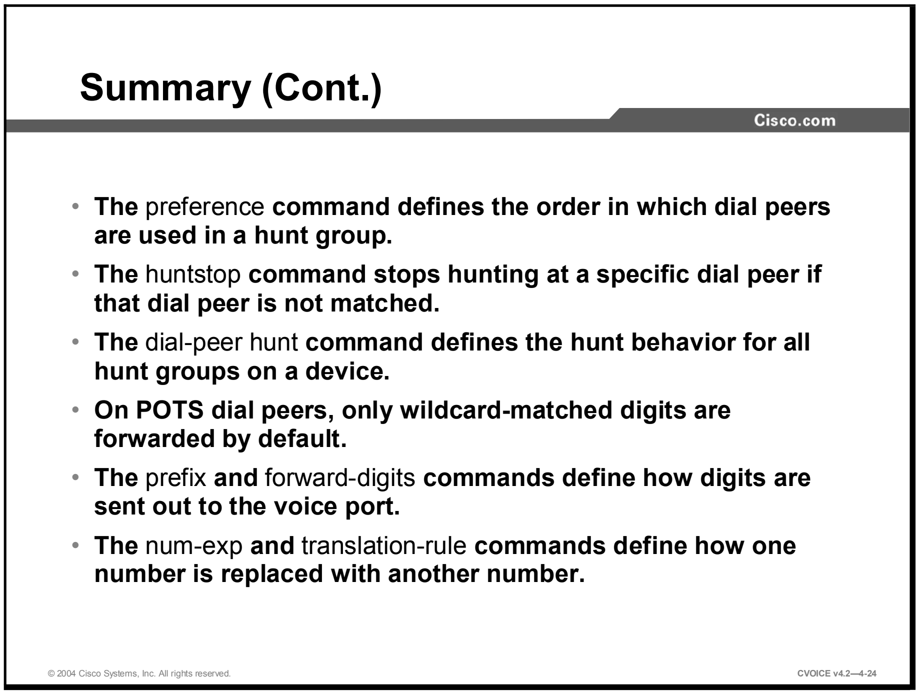
- Destination patterns can define specific telephone numbers or use wildcards to define a range of numbers.
- If no matching inbound dial peer is configured for a call, the default dial peer is used.
- Inbound dial-peer matching uses the incoming called-number, answer-address, destination pattern, and port—in that order—to match inbound dial peers.
- Outbound dial-peer matching uses the longest number match in the destination pattern to match an outbound dial peer.
- Hunt groups are created when more than one dial peer has the same destination pattern, but points to a different voice port or session target.

 CVOICE v4.2—4-23



References

For additional information, refer to these resources:

- *Dial Peer Overview*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/vvfpeers.htm#xtocid2
- *Case Study: Understanding Inbound Matching and Default Dial-Peer 0*
  http://www.cisco.com/warp/public/788/voip/in_dial_peer_match.html#fifth
- *Voice—Understanding Inbound and Outbound Dial Peers on Cisco IOS*
  http://www.cisco.com/warp/public/788/voip/in_out_dial_peers.html
- *VR: Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_r/
- *Cisco IP Telephony*, Cisco Press; ISBN: 1587050501; 1st edition (December 17, 2001)

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 4-1: POTS Dial Peers

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which two functions are performed by a POTS dial peer? (Choose two.)

A) provides an address for the edge network or device

B) provides a destination address for the edge device that is located across the network

C) routes the call across the network

D) identifies the specific voice port that connects the edge network or device

E) associates the destination address with the next-hop router or destination router, depending on the technology used

Q2) The address is configured on each dial peer and is called a _____.

A) telephone number

B) number range

C) destination pattern

D) call endpoint

Q3) Which two parameters must be specified on a router that is connected to a telephone? (Choose two.)

A) voice port

B) dial type

C) calling plan

D) telephone number

Q4) At which router must you configure a POTS dial peer?

A) one inbound router on the network

B) one outbound router on the network

C) one inbound and one outbound router on the network

D) each router where edge telephony devices connect to the network

Q5) Which command is used to specify the address of the terminating router or gateway?

A) **destination-port**

B) **destination-pattern**

C) **session target**

D) **destination address**

E) **dial-peer terminal**

Q6) Why should the loopback address be used in the **session target** command?

A) The call fails if the interface goes down.

B) The interface will never shut down.

C) The call will use an alternate path if the interface shuts down.

D) The call will never fail as long as the router is operating.

Q7) What does a plus sign (+) before the telephone number indicate?

A) The telephone number must conform to ITU-T Recommendation E.164.

B) The number is an extension of a telephone number.

C) An additional digit must be dialed before the telephone number.

D) The telephone number can vary in length.

Q8) Which special character in a destination-pattern string is used as a wildcard?

A) asterisk (*)

B) pound sign (#)

C) comma (,)

D) period (.)

E) brackets ([])

Q9) What happens if there is no matching dial peer for an outbound call?

A) The default dial peer is used.

B) Dial peer 0 is used.

C) The POTS dial peer is used.

D) The call is dropped.

Q10) What is the default dial-peer configuration for inbound POTS peers? (Choose four.)

A) any codec

B) no ivr application

C) vad enabled

D) no rsvp support

E) ip precedence 0

Q11) Which configurable parameter is set for POTS dial peers only?

A) answer-address

B) destination-pattern

C) incoming called-number

D) port

Q12) In what order does the router attempt to match the called number of the call setup request with a dial-peer attribute?

_____ 1. **answer-address**

_____ 2. **destination-pattern**

_____ 3. **incoming called-number**

_____ 4. **port**

Q13) Match the dial-peer configuration to the specified destination. (There may be more than one answer for each destination.).

A) **dial-peer voice 1 pots**
destination-pattern .T
port 1/0:1

B) **dial-peer voice 2 pots**
destination-pattern 555[0-2,5]…
port 1/1/0

C) **dial-peer voice 1 pots**
destination-pattern 5553…
port 1/0:1

D) **dial-peer voice 1 pots**
destination-pattern 5553216
port 1/0.1

_____ 1. dialed number is 5551234

_____ 2. dialed number is 5550000

_____ 3. dialed number is 5553216

Q14) When the router is matching outbound VoIP dial peers, which command is used to forward the call?

A) **destination-pattern**

B) **port**

C) **session target**

D) **show dialplan number *string***

Q15) Match the hunt-group commands with their functions.

A) **preference**

B) **dial-peer hunt**

C) **show dial-peer voice summary**

D) **huntstop**

_____ 1. shows the current settings for dial-peer hunt

_____ 2. sets priority for dial peers

_____ 3. stops dial-peer hunting on the dial peer

_____ 4. changes the default selection order for hunting through dial peers

Q16) Which destination has the highest priority?

A) **dial-peer voice 1 pots**
destination-pattern 5551000
port 1/0/0
preference 0

B) **dial-peer voice 1 voip**
destination-pattern 5551200
port 1/1/1
preference 9

C) **dial-peer voice 1 pots**
destination-pattern 5551234
port 1/0/0
preference 1

Q17) By default, which is the first criterion that a router uses to select dial peers in a hunt group?

A) The router matches the most specific phone number.

B) The router matches according to the preference setting.

C) The router matches the POTS dial peer first.

D) The router matches randomly.

Q18) When you are configuring a hunt group, which command should you use if the destination pattern of the dial peer is the same for several dial peers?

A) **dial-peer hunt**

B) **huntstop**

C) **preference**

D) **priority**

Q19) The user is dialing the number 5550124. In the space below each dial peer configuration, specify the digits that are passed out of the telephony interface.

A) **dial-peer voice 1 pots**
destination-pattern 5550124
port 1/0/0

____

B) **dial-peer voice 2 pots**
destination-pattern 555…
port 1/0/0

____

C) **dial-peer voice 3 voip**
destination-pattern 555….
session target ipv4: 10.0.0.1

____

D) **dial-peer voice 4 pots**
destination-pattern 555….
port 1/0/0
no digit-strip

____

E) **dial-peer voice 5 pots**
destination-pattern 555….
port 1/0/0
forward-digits 6

____

Q20) What is the default behavior of a terminating router when it is matching a dial string to an outbound POTS dial peer?

A) The router removes the far-left digits that match the destination pattern, and forwards the remaining digits.

B) The router strips off the wildcard digits and forwards the matching digits only.

C) The router forwards all the digits without attempting a match.

D) The router does not forward any digits if it cannot match them.

Q21) A **number expansion table** command is used on a dial peer:

num-exp 1… 5551…

!
dial-peer voice 1 pots
destination-pattern 5551…
port 1/1/0

If a user dials the number 1825, what numbers will be matched for an outbound dial peer?

A) 1825

B) 555

C) 5551

D) 5551825

Q22) When a dial peer is configured with the **prefix** command, when are digits added to the front of the dial string?

A) before the outbound dial peer is matched

B) after the outbound dial peer is matched

C) after the digits are sent out of the telephony interface

D) when the digits are received on the dial peer

Quiz Answer Key

Q1) A, D

Relates to: Understanding Dial Peers

Q2) C

Relates to: Understanding Dial Peers

Q3) A, D

Relates to: Configuring POTS Dial Peers

Q4) D

Relates to: Configuring POTS Dial Peers

Q5) C

Relates to: Configuring VoIP Dial Peers

Q6) B

Relates to: Configuring VoIP Dial Peers

Q7) A

Relates to: Configuring Destination-Pattern Options

Q8) D

Relates to: Configuring Destination-Pattern Options

Q9) D

Relates to: Default Dial Peer

Q10) A, C, D, E

Relates to: Default Dial Peer

Q11) D

Relates to: Matching Inbound Dial Peers

Q12)

1. **incoming called-number**
2. **answer-address**
3. **destination-pattern**
4. **port**

Relates to: Matching Inbound Dial Peers

Q13) 1-A, B; 2-A, B; 3-D

Relates to: Matching Outbound Dial Peers

Q14) C

Relates to: Matching Outbound Dial Peers

Q15) 1-C, 2-A, 3-D, 4-B

Relates to: Hunt-Group Commands

Q16) A

Relates to: Hunt-Group Commands

Q17) A

Relates to: Configuring Hunt Groups

Q18) C

Relates to: Configuring Hunt Groups

Q19) A-None
B-012
C-5550124
D-5550124
E-555012

Relates to: Digit Collection and Consumption

Q20) A

Relates to: Digit Collection and Consumption

Q21) D

Relates to: Understanding Digit Manipulation

Q22) B

Relates to: Understanding Digit Manipulation

Practice Item Answer Key

Practice Item 1: POTS Dial Peers

R1:

```
dial-peer voice 1 pots
destination-pattern 2222
port 1/0/0
```

R2:

```
dial-peer voice 1 pots
destination-pattern 3111
port 1/0/0

dial-peer voice 2 pots
destination-pattern 3112
port 1/0/1

dial-peer voice 3 pots
destination-pattern 3113
port 1/1/0
```

Practice Item 2: VoIP Dial-Peer Configuration

R1:

```
dial-peer voice 101 voip
destination-pattern 3...
session target ipv4:10.1.1.2
```

R2:

```
dial-peer voice 101 voip
destination-pattern 2...
session target ipv4:10.1.1.1
```


Practice Item 3: Matching Inbound Dial Peers

R1:

```
dial-peer voice 202 voip
incoming called-number 5550111
session target ipv4:10.1.1.2

dial-peer voice 203 voip
incoming called-number 5550122
session target ipv4:10.1.1.3

dial-peer voice 204 voip
incoming called-number 5550133
session target ipv4:10.1.1.4
```

Practice Item 4: Configuring Hunt Groups

R2:

```
dial-peer voice 4 pots
destination-pattern 3111
port 1/0/1
preference 2
```

Practice Item 5: Digit Manipulation

R1:

```
dial-peer voice 300 pots
destination-pattern 3...
port 1/1/0
forward-digits all
prefix 555
preference 2
```

Understanding Special-Purpose Connections

Overview

This lesson explores the uses and applications of various special-purpose connections on Cisco Systems telephony equipment.

Relevance

Integrating VoIP technologies to legacy PBXs and PSTNs often requires voice port configuration for certain connection types. The original design often calls for tie-lines between PBXs. When replacing tie-lines with a VoIP solution, special configuration at the voice port level can emulate the original tie-line design. In many cases, telecommuters require access to PBX services that resemble other extensions of the PBX, regardless of where they actually reside. In other instances, telephones, such as lobby customer-service telephones, need to be connected directly to customer-service staff. It is important to understand how to provide these services through voice port configuration.

Objectives

Upon completing this lesson, you will be able to correctly configure voice ports for connection types necessary to integrate VoIP technologies with legacy PBXs and PSTN. This includes being able to meet these objectives:

- Identify different special-purpose connection commands
- Describe how the network establishes PLAR and PLAR-OPX connections
- Configure trunk connections
- Describe how the network establishes tie-line connections

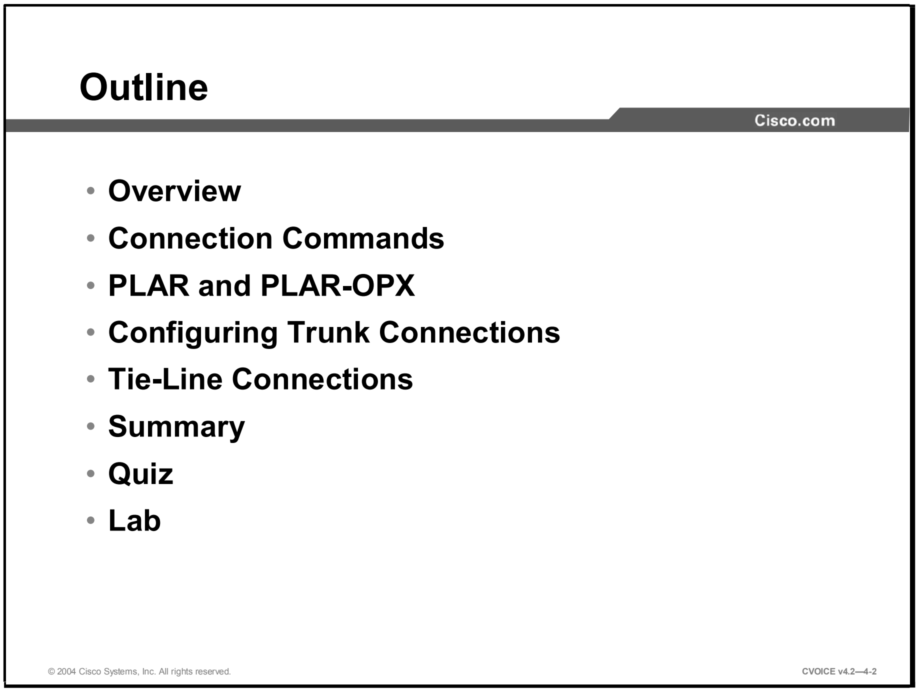
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with trunk and tie-line concepts
- Familiarity with voice port configuration

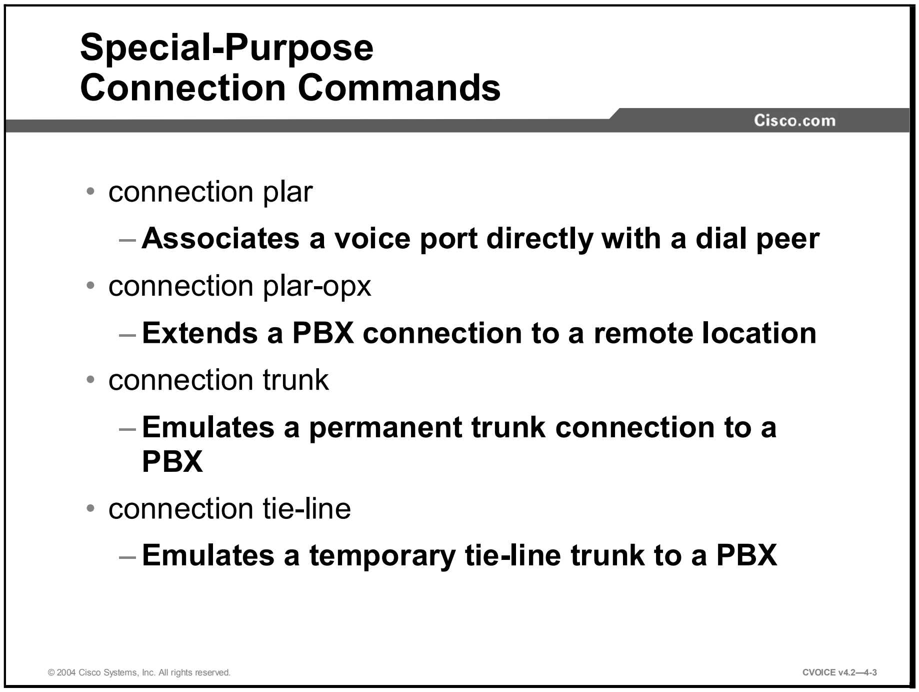
Outline

The outline lists the topics included in this lesson.



Connection Commands

This topic identifies different special-purpose connection commands.



You can configure voice ports to support special connection requirements. These requirements usually reflect the needs of a specific business environment that must connect to the network in a special way. The following is a list of available connection commands and their application:

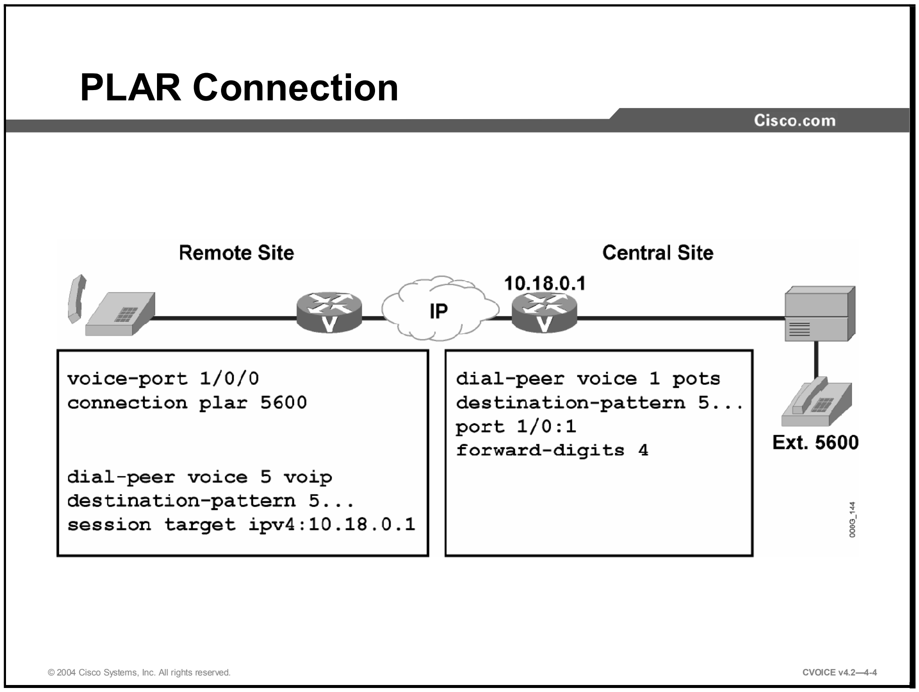
- **connection plar:** Private line, automatic ringdown (PLAR) is an autodialing mechanism that permanently associates a voice port with a far-end voice port, allowing call completion to a specific telephone number or PBX. When the calling telephone goes off hook, a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX. The caller does not hear a dial tone and does not have to dial a number. PLAR connections are widely used in the business world. One common use is to connect stockbrokers with trading floors. Timing is critical when dealing with stock transactions—the amount of time it may take to dial a number and get a connection can be costly in some cases. Another common use is in the travel sector, directly connecting travelers with services. Often, at places like airports, the traveler will see display boards advertising taxi companies, car rental companies, and local hotels. These displays often have telephones that will connect the traveler directly with the service of choice; the device is preconfigured with the telephone number of the desired service. One obvious difference between these telephones and a normal telephone is that they do not have a dial mechanism.
- **connection plar-opx:** Most frequently, a PLAR-Off Premises eXtension (OPX) is a PBX extension that is not located on the business site even though it operates as though it is directly connected to the PBX. Company staff can dial an extension and reach the remote telephone as though it were on site. The remote telephone has access to PBX services such as voice mail and extension dialing. This functionality is most often used when onsite staff turns into telecommuters. Many companies are cutting back on office space in expensive locations and are setting up their staff with home offices. A PLAR-OPX connection is configured between the office and the remote site so that the telecommuter can continue to

access all the corporate telephony services in the same manner as before. This allows the telecommuter to dial the same extensions to reach other staff, and to have access to long-distance dialing and other voice services via the same calling codes. From the office perspective, onsite staff can reach the telecommuter by dialing the same extension as before. One OPX connection feature is that when a call is being attempted, the voice-enabled router or gateway that takes the call from the PBX or Cisco CallManager will not report a call completion until the far end has answered the call. Without the OPX configuration, the PBX or Cisco CallManager passes the call to the local gateway or router. Then, the gateway or router routes the call to the PSTN. After the PSTN sends ringing to the telephone, the router will report call completion back to the PBX or Cisco CallManager. At this point, the call is completed. The problem is that if the call is not answered, there is no way to reroute the call to the corporate voice-mail server. From the PBX or Cisco CallManager perspective, the call is completed. When you configure the OPX, however, the gateway or router will not report call completion unless the telephone is actually answered.

- **connection trunk:** The **connection trunk** command specifies a connection that emulates a permanent trunk connection between two PBXs, a PBX and a local extension, or some combination of telephony interfaces with signaling passed transparently through the packet data network. A trunk connection remains permanent in the absence of active calls and is established immediately after configuration. The ports on either end of the connection are dedicated until you disable trunking for that connection. If, for some reason, the link between the two voice ports goes down, the virtual trunk reestablishes itself after the link comes back up. This configuration is useful when a permanent connection is desired between two devices. In this scenario, a caller at one end of the trunk connection can pick up the telephone and speak into it without dialing any digits or waiting for call setup. This is analogous to the red telephone to the Kremlin that is depicted in vintage movies. With a trunk connection, there is no digit manipulation performed by the gateway or router. Because this is a permanent connection, digit manipulation is not necessary.
- **connection tie-line:** The **connection tie-**line command specifies a connection that emulates a temporary tie-line trunk to a PBX. Although a tie-line connection is similar to a trunk connection, it is automatically set up for each call and torn down when the call ends. Another difference is that digits are added to the dial string *before* matching an outbound dial peer; for example, if a user were to dial extension 8000, which terminates at a remote office, the voice port is configured with an identifying number for that remote office. If that office ID is the number 7, then the digits that are sent to be matched against the outbound dial peer would be 78000. This new five-digit number would be carried across the network to the remote site. At the remote site, the number 7 can be stripped off or, if necessary, passed to the destination device.

PLAR and PLAR-OPX

This topic describes the use of PLAR and PLAR-OPX connections.



As demonstrated in the figure, the following actions must occur to establish a PLAR connection:

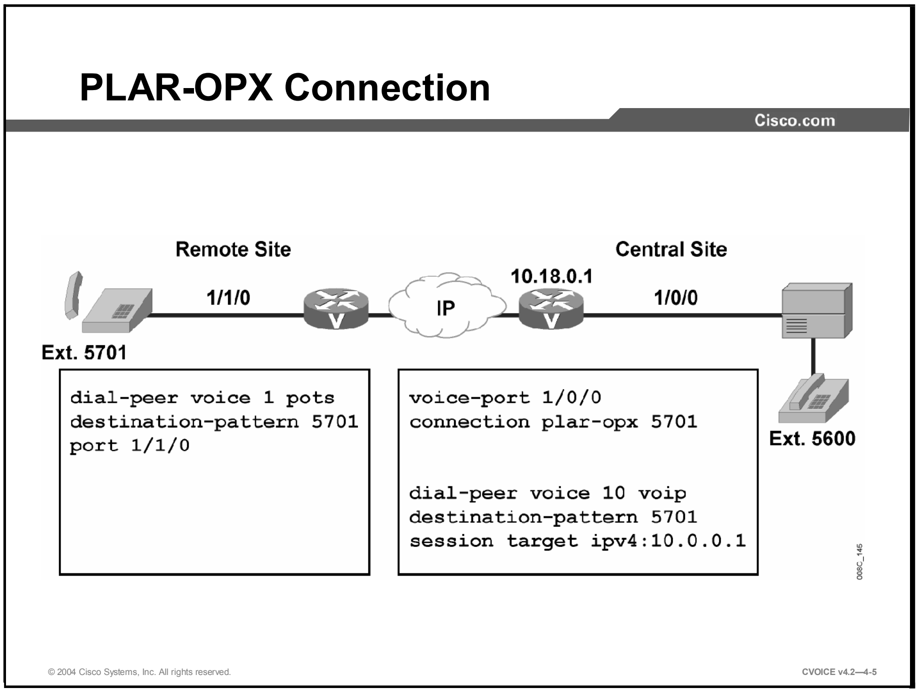
1. A user at the remote site lifts the handset.

2. A voice port at the remote site router automatically generates digits 5600 for a dial-peer lookup.

3. The router at the remote site matches digits 5600 to VoIP dial peer 5 and sends the setup message with the digits 5600 to IP address 10.18.0.1 as designated in the **session target** statement.

4. The router at the central site matches received digits 5600 to POTS dial peer 1 and forwards digits 5600 out voice port 1/0:1. At the same time, it sends a call-complete setup message to the router at the remote site because both the inbound and outbound call legs on the central-site router were processed correctly.

5. The PBX receives digits 5600 and rings the appropriate telephone.



As demonstrated in the figure, the following actions must occur to establish a PLAR-OPX connection:

1. A user at the central site calls a user at a remote site using the extension 5701.

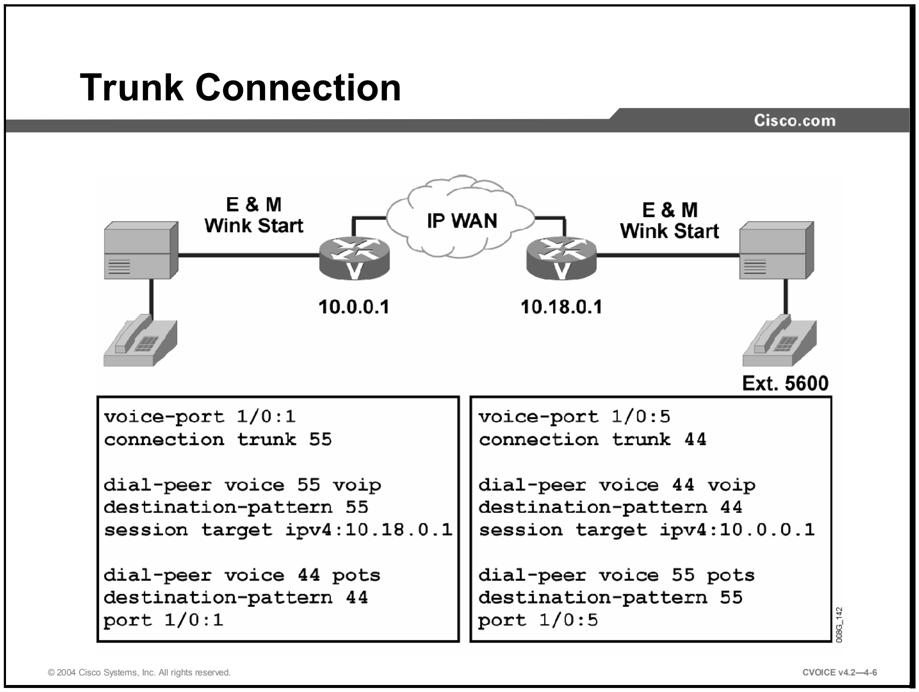
2. PBX routes the call to the central-site router port 1/0/0, which is configured for PLAR-OPX and pointing to extension 5701.

3. The central-site router matches VoIP dial peer 10 and sends a setup message to the corresponding IP address. In the meantime, port 1/0/0 does not respond immediately to the PBX with a seizure or off hook indication, but waits for the remote site call setup complete message.

4. After the remote router sends the call setup complete message, the central-site router sends a trunk seizure indication to the PBX and opens a voice path.

Configuring Trunk Connections

This topic describes how to configure trunk connections.



As demonstrated in the figure, the following must occur to establish a trunk connection:

1. Use the **connection trunk** command to establish a two-way permanent connection between two voice ports across the IP network.

2. Configure the **connection trunk** parameter on the voice ports connecting the two PBXs and configure the session target for each IP address.

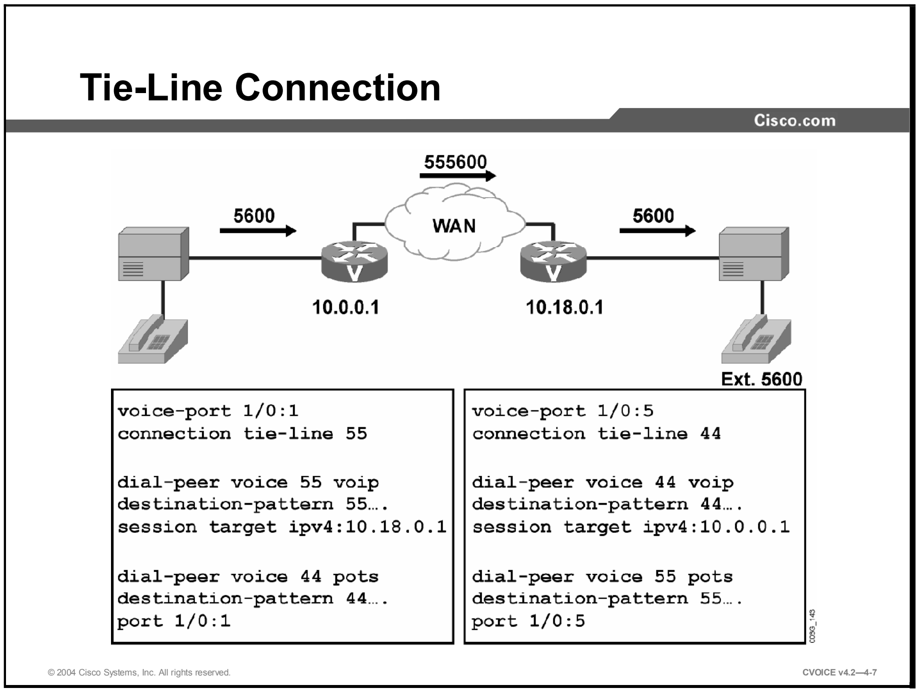
In the example, the router on the left is configured to set up a trunk connection from voice port 1/0:1 to a remote voice-enabled router with the IP address of 10.18.0.1 (the router on the right). This is done by specifying the same number in the **connection trunk** voice port command as in the appropriate dial peer **destination-pattern** command. In this example, the router on the left uses **connection trunk 55**, which matches VoIP dial peer 55. The call is routed to the router on the right, which matches the 55 in a POTS dial peer. The router on the right is also configured to set up a trunk connection from its voice port 1/0:5 to a remote voice-enabled router with the IP address of 10.0.0.1 (the router on the left). The router on the right uses 44 as its connection trunk number. These trunk connections are set up when the routers power on and remain up until the router is powered down or the ports are shut down.

The following conditions must be met for VoIP to support virtual trunk connections:

- You must use the following voice port combinations:
  - recEive and transMit or ear and mouth (E&M) to E&M (same type)
  - FXS to Foreign Exchange Office (FXO)
  - FXS to FXS (with no signaling)
- You must not perform number expansion on the destination-pattern telephone numbers configured for trunk connection.
- You must configure both end routers for trunk connections.

Tie-Line Connections

This topic describes the use and application of tie-lines.



In traditional telephony networks, companies often had dedicated circuits called tie-lines connecting two PBXs. This, in effect, allowed callers at one site to reach callers at the remote site only through that tie-line connection. Now that the IP network is replacing the traditional telephony connection, the two sites are logically "tied" together through the use of the **connection tie-line** command at both sites. Callers at one site can still reach callers at the remote site only, but the call goes over the IP network. The **connection tie-line** command emulates tie-lines between PBXs.

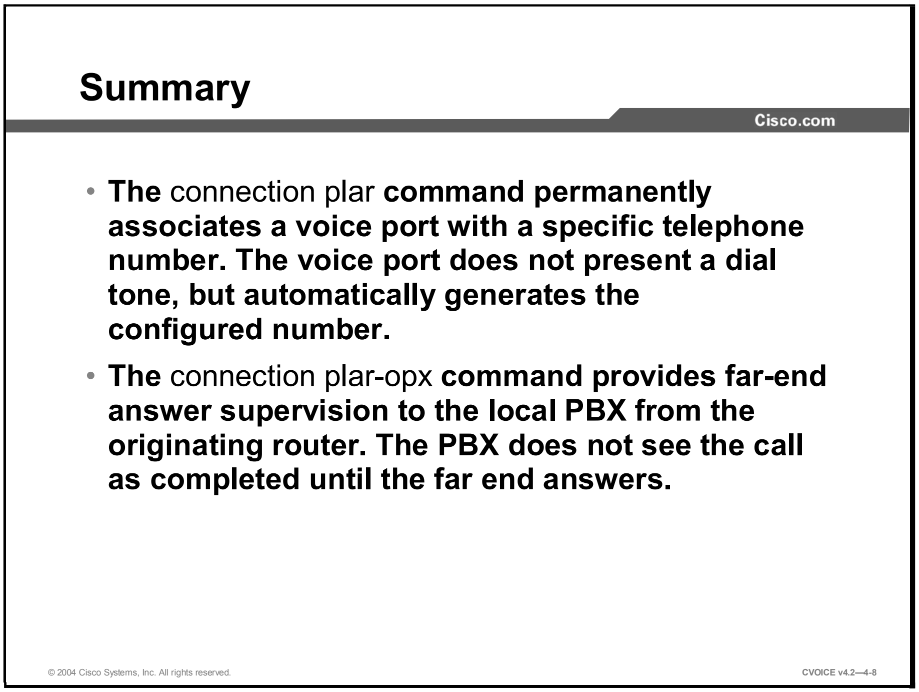
As demonstrated in the figure, you must complete the following procedure to establish a tie-line connection:

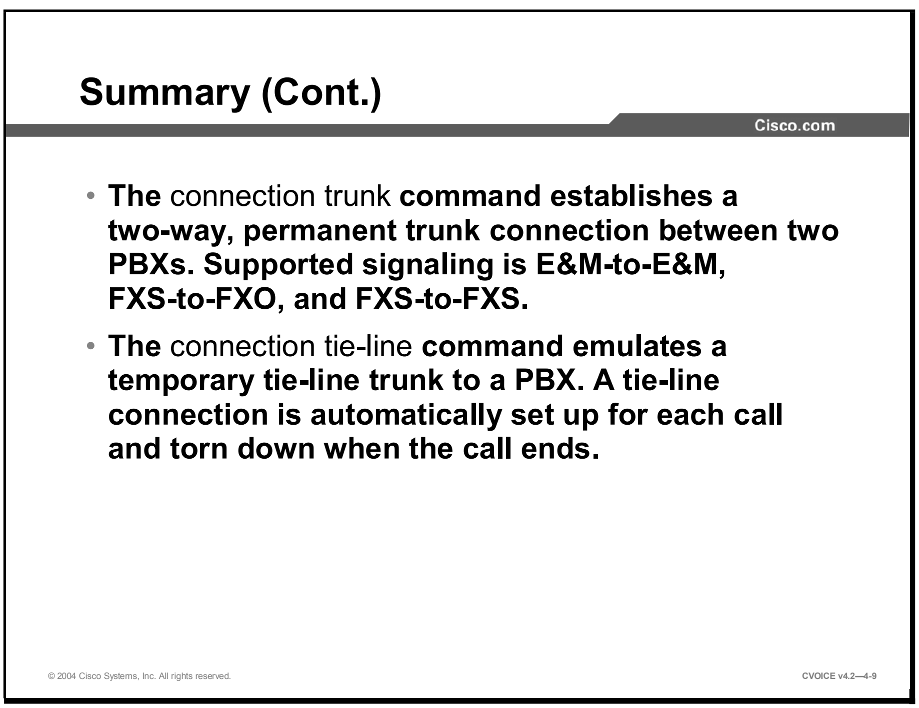
1. Use the **connection tie-line** command when the dial plan requires the addition of digits in front of any digits dialed by the PBX.
2. Use the combined set of digits to route the call onto the network.
3. The tie-line port waits to collect digits from the PBX.
4. The terminating router automatically strips the tie-line digits.

In the figure, the caller on the left picks up the telephone and dials the four-digit extension, 5600. Because the voice port on the left router is configured for **connection tie-line**, the router collects the four digits and prepends the tie-line digits "55" to make a six-digit number, 555600. That number is then matched to a VoIP dial peer and sent to the appropriate IP address. After the call reaches the far-end router, it is matched against a POTS dial peer with the destination pattern "55….". Because POTS dial peers, by default, forward only wildcard digits, only the four-digit extension 5600 is passed to the PBX.

Summary

This topic summarizes the key points discussed in this lesson.





References

For additional information, refer to this resource:

- *Cisco IOS Voice, Video, and Fax Command Reference*
  (A user ID and password are required to access this site.)
  http://www.cisco.com/en/US/partner/products/sw/iosswrel/ps1835/products_configuration_guide_book09186a0080080ada.html

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 4-2: Connections

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which Cisco IOS command would you use if you want a site ID number to be prepended to the dialed digits before they are trunked across the network?

A) **connection plar**

B) **connection plar-opx**

C) **connection tie-line**

D) **connection trunk**

Q2) What happens when a dial peer is configured with Cisco IOS command **connection plar**?

A) The caller hears a dial tone and the number is automatically dialed.

B) The caller does not hear a dial tone and the call is automatically set up.

C) The caller dials an extension and reaches a telephone on a remote site.

D) The caller does not hear a dial tone and the call is set up after dialing.

Q3) The voice port at the remote site is configured with this command:

voice-port 1/0/0
connection plar 5678

What is the next step to make a call after the user at the remote site lifts the handset?

A) The user must dial extension 5678 to make the call.

B) The telephone will automatically dial 5678 and the user need only dial the extension.

C) The voice port will automatically generate digits 5678 for a dial-peer lookup.

D) The voice port has been permanently associated with dial peer 5678 and the call is already established.

Q4) The voice port at the remote site is configured with this command:

voice-port 1/0/0
connection plar-opx 5678

What is the next step to make a call after the user at the remote site lifts the handset?

A) The user must dial extension 5678 to make the call.

B) The telephone will automatically dial 5678 and the user need only dial the extension.

C) The voice port will automatically generate digits 5678 for a dial-peer lookup.

D) The voice port has been permanently associated with dial peer 5678 and the call is already established.

Q5) Which two conditions are necessary for VoIP to support virtual trunk connections? (Choose two.)

A) Both end routers are configured for trunk connections.

B) Number expansion is used for the telephone numbers configured for the trunk connection.

C) E&M voice ports are connected to E&M voice ports.

D) FXO voice ports are connected to FXO voice ports.

E) FXS voice ports are connected to FXO voice ports with no signaling.

Q6) On which POTS voice ports must the **connection trunk** parameter be configured?

A) the voice ports connecting the two FXS trunks

B) the voice ports connecting the FXS trunk to the FXO trunk

C) the voice ports connecting the two PBX trunks

D) the voice ports connecting the E&M trunk to the FXS trunk

Q7) A dial peer is configured with this command:

voice-port 1/0:1
connection tie-line 35

If the caller dials 7901, what number does the router at the caller end try to match to a dial peer?

A) 35

B) 7901

C) 790135

D) 357901

Q8) How do tie-line connections over IP networks differ from tie-line connections over traditional telephony networks?

A) The calls go over the IP network.

B) Callers can dial a shorter number.

C) Callers at one site can reach callers at any other site.

D) Callers at one site can reach callers at the remote site only.

Quiz Answer Key

Q1) C

Relates to: Connection Commands

Q2) B

Relates to: Connection Commands

Q3) C

Relates to: PLAR and PLAR-OPX

Q4) C

Relates to: PLAR and PLAR-OPX

Q5) A, C

Relates to: Configuring Trunk Connections

Q6) C

Relates to: Configuring Trunk Connections

Q7) D

Relates to: Tie-Line Connections

Q8) A

Relates to: Tie-Line Connections

Building a Scalable Numbering Plan

Overview

This lesson describes the attributes of a scalable numbering plan for voice networks, addresses the challenges of designing these networks, and identifies the methods of implementing numbering plans.

Relevance

To integrate VoIP networks into existing voice networks, network administrators must have the skills and knowledge to implement a comprehensive, scalable, and logical numbering plan.

Objectives

Upon completing this lesson, you will be able to assess the need for and implement a scalable numbering plan in a VoIP network. This includes being able to meet these objectives:

- List four customer components that must be considered when implementing a scalable numbering plan and explain why they are important
- Describe the required attributes of a scalable numbering plan and list the benefits provided
- Describe the five attributes and advantages of a hierarchical numbering plan and list the benefits provided
- Describe the challenges associated with the integration of internal numbering with the public numbering plan
- Describe two methods that are used to integrate existing dial plans into a VoIP network

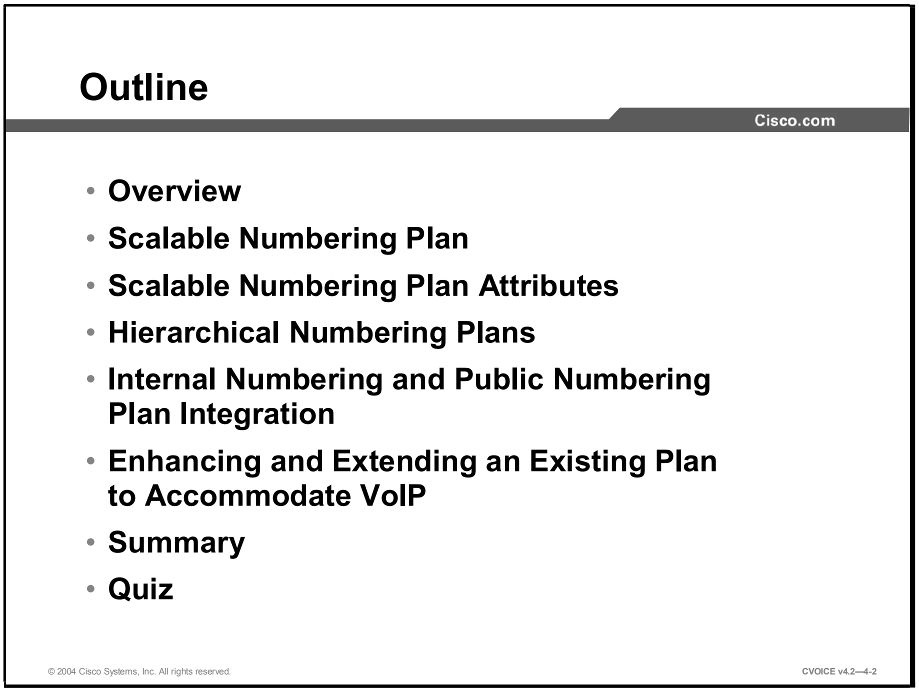
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of VoIP networks
- Basic knowledge of telephone numbering plans, such as E.164

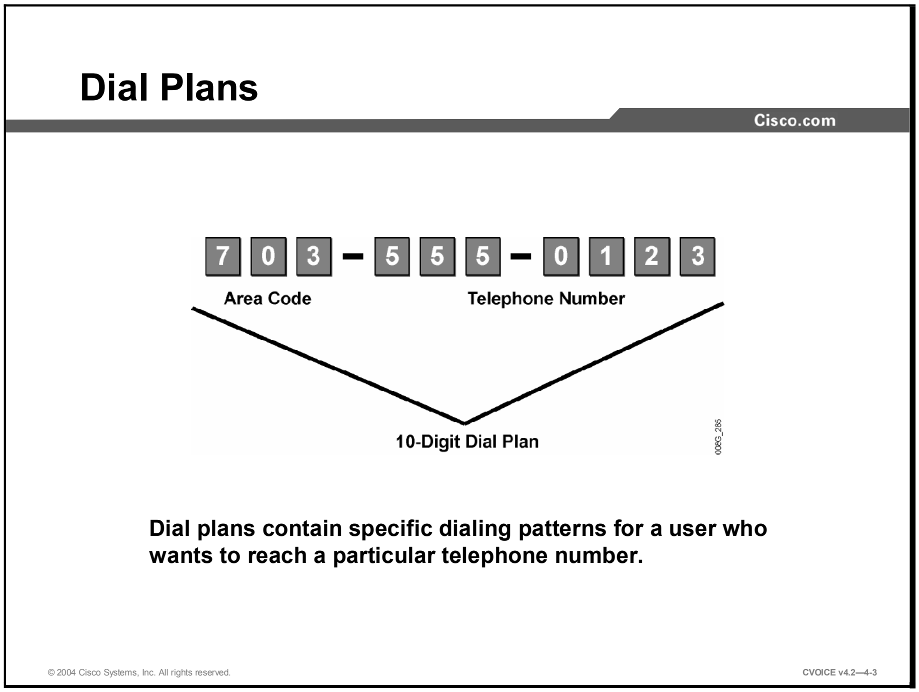
Outline

The outline lists the topics included in this lesson.



Scalable Numbering Plan

This topic describes the need for a scalable numbering plan in a VoIP network.



Although most people are not acquainted with dial plans by name, they use them daily.

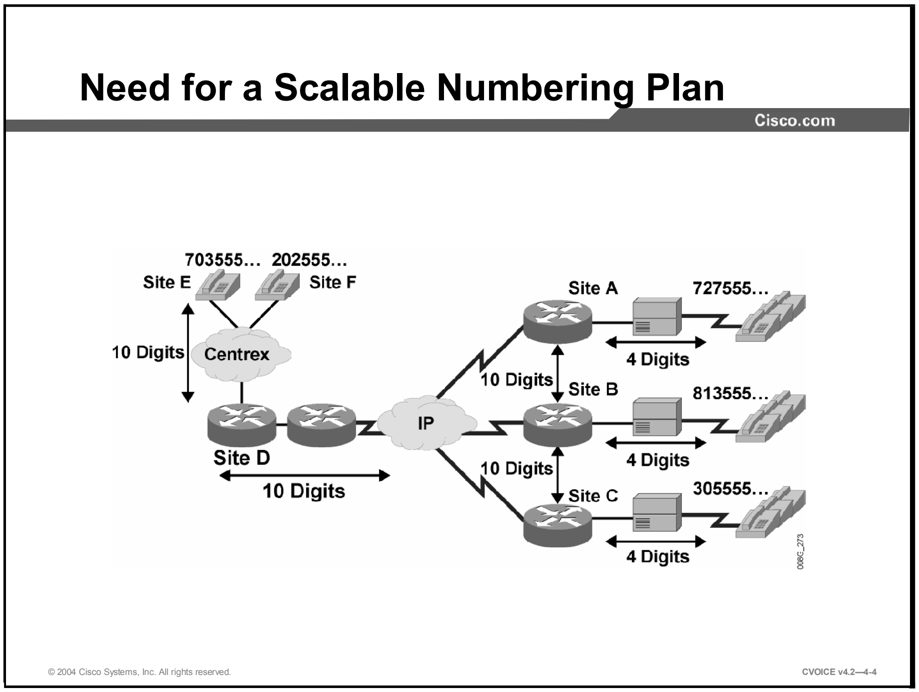
Example: Dial Plan Implementations

The North American telephone network is designed around a 10-digit dial plan that consists of 3-digit area codes and 7-digit telephone numbers. For telephone numbers that are located within an area code, the PSTN uses a 7-digit dial plan. Features within a CO-based PBX, such as Centrex, allow the use of a custom 5-digit dial plan for customers who subscribe to that service. PBXs are more flexible and allow for variable-length dial plans containing 3 to 11 digits.

Dial plans contain specific dialing patterns for a user who wants to reach a particular telephone number. Dial plans also contain access codes, area codes, specialized codes, and combinations of the numbers of digits dialed.

Dial plans require knowledge of the customer network topology, current telephone number dialing patterns, proposed router and gateway locations, and traffic-routing requirements. If the dial plans are for a private internal voice network that is not accessed by the outside voice network, the telephone numbers can be any number of digits.

Typically, companies that implement VoIP networks carry voice traffic within the least expensive systems and paths. Implementing this type of system involves routing calls through IP networks, private trunks, PBXs, key systems, and the PSTN. The numbering plan to support the system is scalable, easily understood by the user, and transportable between all of the system components. The use of alternate path components reduces instances of call failure. Finally, the numbering plan conforms to all applicable standards and formats for all of the systems involved.



This figure illustrates a complex voice network that consists of the components discussed in this topic. A comprehensive and scalable numbering plan must be well-planned and well-implemented on networks such as this. The Centrex service requires 7-digit dialing between itself and site D; the IP network requires 7-digit dialing toward sites A, B, and C; and each of the PBXs requires 3-digit dialing.

Scalable Numbering Plan Attributes

This topic describes the attributes of a scalable numbering plan.

Attributes of a Scalable Numbering Plan

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- **Logic distribution**
- **Hierarchical design**
- **Simplicity in provisioning**
- **Reduction in postdial delay**
- **Availability and fault tolerance**

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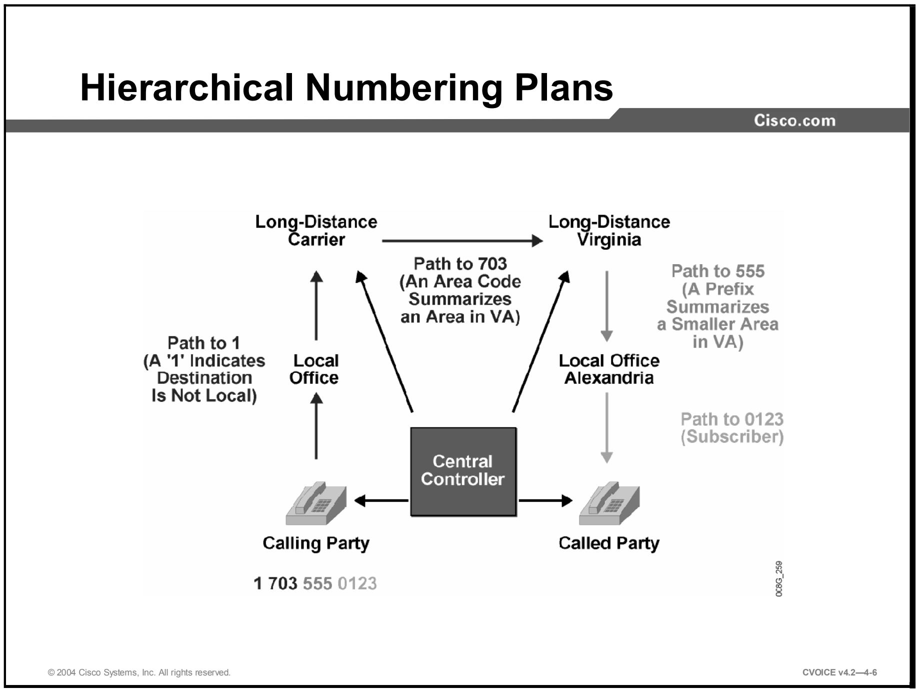
When designing a large-scale numbering plan, you must adhere to the following attributes:

- **Logic distribution:** Good dial plan architecture relies on the effective distribution of the dial plan logic among the various components. Devices that are isolated to a specific portion of the dial plan reduce the complexity of the configuration. Each component focuses on a specific task accomplishment. Generally, the local switch or gateway handles details that are specific to the local point of presence (POP). Higher-level routing decisions are passed along to the gatekeepers and PBXs. A well-designed network places the majority of the dial plan logic at the gatekeeper devices.
- **Hierarchical design (scalability):** You must strive to keep the majority of the dial plan logic (routing decisions and failover) at the highest-component level. Maintaining a hierarchical design makes the addition and deletion of number groups more manageable. Scaling the overall network is much easier when configuration changes are made to a single component.
- **Simplicity in provisioning:** Keep the dial plan simple and symmetrical when designing a network. Try to keep consistent dial plans on the network by using translation rules to manipulate the local digit dialing patterns. These number patterns are normalized into a standard format or pattern before the digits enter the VoIP core. Putting digits into a standard format simplifies provisioning and dial-peer management.
- **Reduction in postdial delay:** Consider the effects of postdial delay in the network when you design a large-scale dial plan. Postdial delay is the time between the last digit dialed and the moment the phone rings at the receiving location. In the PSTN, people expect a short postdial delay and to hear ringback within seconds. The more translations and lookups that take place, the longer the postdial delay becomes. Overall network design, translation rules, and alternate pathing affect postdial delay. You must strive to use these tools most efficiently to reduce postdial delay.

- **Availability and fault tolerance:** Consider overall network availability and call success rate when you design a dial plan. Fault tolerance and redundancy within VoIP networks are most important at the gatekeeper level. By using an alternate path you help provide redundancy and fault tolerance in the network.

Hierarchical Numbering Plans

This topic describes the advantages and attributes of hierarchical numbering plans.



Scalable telephony networks require telephone-numbering plans that are hierarchical. A hierarchical design has the following advantages:

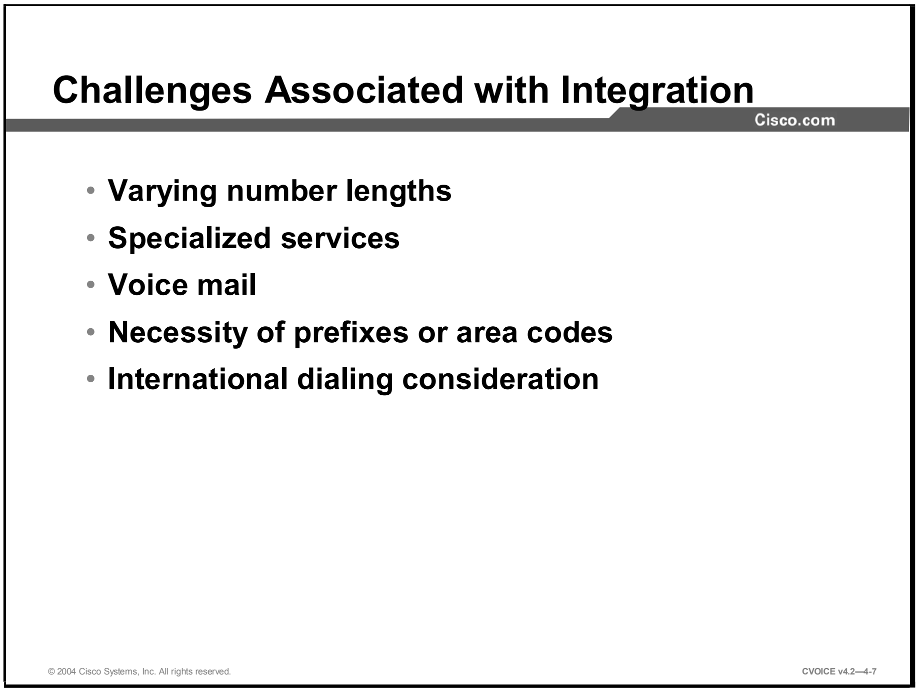
- **Simplified provisioning:** Provides the ability to easily add new groups and modify existing groups
- **Simplified routing:** Keeps local calls local and uses a specialized number key, such as an area code, for long-distance calls
- **Summarization:** Establishes groups of numbers in a specific geographical area or functional group
- **Scalability:** Adds scalability to the number plan by adding additional high-level number groups
- **Management:** Controls number groups from a single point in the overall network

It is not easy to design a hierarchical numbering plan. Existing numbering plans in the network (such as proprietary PBXs, key systems, and telephony services such as Centrex), and the necessity to conform to the PSTN at gateways, all contribute to the complexity of the design. Translation between these systems is a difficult task. If possible, avoid retraining system users. The goal is to design a numbering plan that has the following attributes:

- Minimal impact on existing systems
- Minimal impact on users of the system
- Minimal translation configuration
- Consideration of anticipated growth
- Conformance to public standards, where applicable

Internal Numbering and Public Numbering Plan Integration

This topic describes the challenges associated with integrating internal numbering with the public numbering plan.



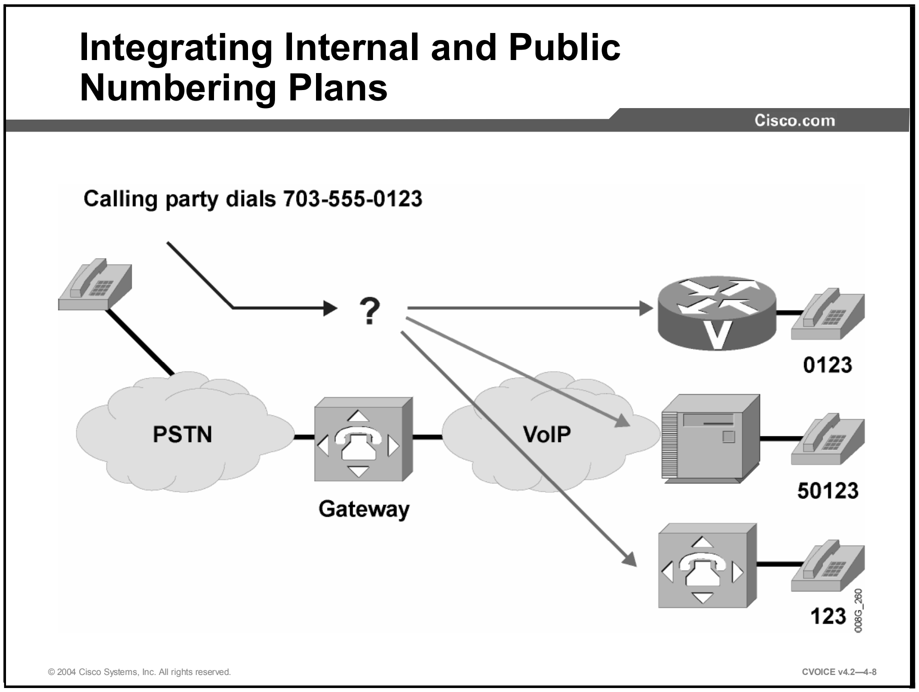
Numbering plans vary greatly throughout the world. Different countries use different number lengths and hierarchical plans within their borders. Telephony equipment manufacturers and service providers use nonstandard numbering. In an attempt to standardize numbering plans, the International Telecommunication Union (ITU) developed the E.164 worldwide prefix scheme.

Number plan integration from an internal system such as a VoIP and PBX system to the PSTN requires careful planning. The hierarchical structure of the number plan and the problems associated with varying number lengths in different systems make number plan integration complex.

The challenges that you face with number plan integration include the following:

- **Varying number lengths:** Within the IP network, consideration is given to varying number lengths that exist outside the IP network. Local, long-distance, key system, and Centrex dialing from within the IP network may require digit manipulation.
- **Specialized services:** Services such as Centrex and their equivalents typically have 4- or 5-digit numbers. Dialing from the PSTN into a private VoIP network and then out to a Centrex extension can also require extensive digit manipulation.
- **Voice mail:** When a called party cannot be reached, the network may have to redirect the call to voice mail. Since the voice-mail system can require a completely different number plan than the endpoint telephones, translation is necessary.

- **Necessity of prefixes or area codes:** It can be necessary to strip or add area codes, or prepend or replace prefixes. Rerouting calls from the IP network to the PSTN for failure recovery can require extra digits.
- **International dialing consideration:** Country codes and number plans vary in length within countries. Dialing through an IP network to another country requires careful consideration.

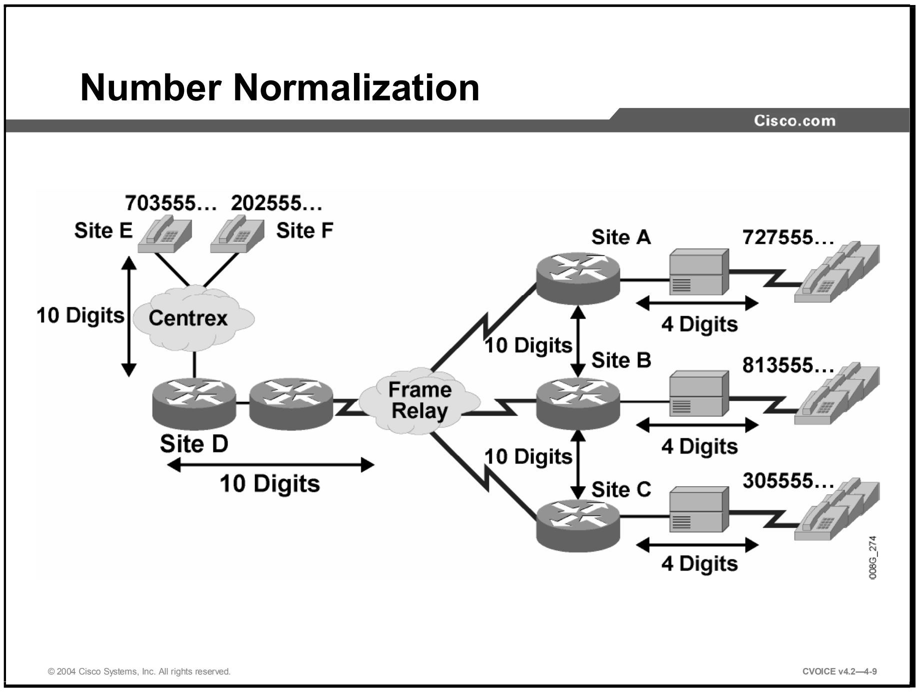


This figure shows a call from the PSTN destined for 1-703-555-0123. The gateway must realize the true destination. All endpoints end with the correct digit sequence, but which is the correct endpoint? Should the gateway append or prepend digits to the dialed number? Should it strip and omit digits?

Different PBXs support variable-length dial plans that contain 3 to 11 digits. The length variations present challenges when private plans merge with public number plans. Issues also arise when no one answers a number and the call is forwarded to a voice-mail system.

Enhancing and Extending an Existing Plan to Accommodate VoIP

This topic describes methods for integrating existing number plans into a VoIP network.



There are many ways that you can enhance and extend an existing number plan to accommodate the VoIP network; all of them require careful planning and consideration. This lesson will discuss two of these ways: number normalization and technology prefixes.

Example: Number Normalization

When site E (703555….) dials 7275550199, the full 10-digit dialed string is passed through the Centrex to the router at site D. Router D matches the destination pattern 7275550199 and forwards the 10-digit dial string to router A. Router A matches the destination pattern 727555…., strips off the matching 727555, and forwards the remaining 4-digit dial string to the PBX. The PBX matches the correct station and completes the call to the proper extension.

Calls in the reverse direction are handled similarly. However, because the Centrex service requires the full 10-digit dial string to complete calls, the POTS dial peer at router D is configured with digit stripping disabled. An alternate solution involves enabling digit stripping and configuring the dial peer with a 6-digit prefix (in this case 703555), which results in forwarding the full dial string to the Centrex service.

Router Digit Stripping Comparison

Router A	Router D
dial-peer voice 1 pots	dial-peer voice 4 pots
destination-pattern 727555….	destination-pattern 703555….
port 1/0:1	**no digit-strip**
!	port 1/0:1
dial-peer voice 4 voip	!
destination-pattern 703555….	dial-peer voice 5 pots
session target ipv4:10.10.10.2	destination-pattern 202555….
!	**no digit-strip**
dial-peer voice 5 voip	port 1/0:1
destination-pattern 202555….	!
session target ipv4:10.10.10.3	dial-peer voice 1 voip
!	destination-pattern 727555….
	session target ipv4:10.10.10.1
	!

Another method, called "technology prefixes," allows you to include special characters in the called number. These special characters (most commonly designated as 1#, 2#, 3#, etc.) are prepended to the called number on the outgoing VoIP dial peer. The gatekeeper then checks its gateway technology prefix table for gateways that are registered with that particular technology prefix. Technology prefixes also identify a type, class, or pool of gateways.

Example: Technology Prefixes Applied

Voice gateways can register with technology prefix 1#; H.320 gateways with technology prefix 2#; and voice-mail gateways with technology prefix 3#. Multiple gateways can register with the same type prefix. When this happens, the gatekeeper makes a random selection among gateways of the same type.

If the callers know the type of device that they are trying to reach, they can include the technology prefix in the destination address to indicate the type of gateway to use to get to the destination. For example, if a caller knows that address 7275550111 belongs to a regular telephone, the caller can use the destination address of 1#7275550111, where 1# indicates that the address should be resolved by a voice gateway. When the voice gateway receives the call for 1#7275550111, it strips off the technology prefix and routes the next leg of the call to the telephone at 7275550111.

You can enter technology prefix commands on gateways and gatekeepers in two places, depending on how you want to design the technology prefix decision intelligence: the gateway VoIP interface or the gateway dial peer.

You can implement this type of digit manipulation and management of dialed numbers in various ways, depending on the infrastructure of the network. All of the components, including the gatekeepers, gateways, Cisco CallManagers, PBXs, key systems, and other systems, may need to be included in the process.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Since implementing a scalable numbering plan involves extensive call routing, it is important to understand a customer network topology, number dialing patterns, router and gateway locations, and traffic requirements.
- The required design-simplification attributes of a scalable numbering plan provide increased manageability and increased call service and delivery.
- The associated attributes of a hierarchical numbering plan are simplified provisioning and routing, summarization, scalability, and management. They provide minimal system and configuration impact, anticipated growth consideration, and conformance to public standards, where applicable.

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Summary (Cont.)

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- Varying number lengths, specialized services, voice mail, necessity of prefixes or area codes, and international dialing considerations are challenges associated with integrating an internal numbering plan with the public numbering plan.
- Digit manipulation and the addition of technology prefixes are methods to extend and enhance VoIP numbering plans.

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References

For additional information, refer to these resources:

- *Designing a Static Dial Plan*
  http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/dp3_isd.htm
- *Understanding One Stage and Two Stage Voice Dialing*
  http://www.cisco.com/warp/public/788/voip/1stage2stage.html
- *Voice Design and Implementation Guide*
  http://www.cisco.com/warp/public/788/pkt-voice-general/8.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) The North American telephone-numbering plan is based on _____ digits.

A) 3

B) 7

C) 10

D) 11

Q2) What three types of information do you need to design a dial plan for a customer? (Choose three.)

A) network topology

B) traffic-routing requirements

C) current dialing patterns

D) proposed router and gateway locations

E) type of PBX and PSTN connection

Q3) Match the attribute of a scalable numbering plan with its description.

A) Logic distribution

B) Hierarchical design

C) Simplicity in provisioning

D) Reduction in postdial delay

E) Availability and fault tolerance

_____ 1. uses alternate paths to make sure there is overall network availability and call success rates

_____ 2. makes the addition and deletion of number groups more manageable

_____ 3. gives the network components the ability to focus on specific tasks to complete calls

_____ 4. is affected by network design, translation rules, and alternate paths

_____ 5. keeps dial plans consistent on the network by using translation rules to manipulate the local digit dialing patterns

Q4) When distributing the dial plan logic, the majority of the dial plan logic should be placed at the _____.

A) dial peers

B) edge devices

C) gatekeepers

D) gateways

Q5) Match the advantage of a hierarchical numbering plan with its definition.

A) simplified provisioning

B) simplified routing

C) summarization

D) scalability

E) management

_____ 1. adds more high-level number groups

_____ 2. provides the ability to add new groups and modify existing groups easily

_____ 3. controls number groups from a single point in the overall network

_____ 4. establishes a group of numbers in a specific geographical area or functions group

_____ 5. keeps local calls local and uses a specialized number key such as an area code for long-distance calls

Q6) Which two factors make the design of hierarchical numbering plans complex? (Choose two.)

A) requirements of long-distance calls

B) varying number lengths

C) number of geographical areas to be included in the network

D) billing mechanisms

E) necessity of prefixes or area codes

Q7) Which of the challenges associated with integration requires translations?

A) varying number lengths

B) specialized services

C) voice mail

D) necessity of prefixes or area codes

E) international dialing

Q8) Which worldwide prefix scheme was developed by ITU to standardize numbering plans?

A) G.114

B) E.164

C) G.164

D) E.114

Q9) Choose two locations where you can enter technology prefix commands. (Choose two.)

A) PBXs

B) gatekeepers

C) gateways

D) key systems

Q10) Technology prefixes can be used to identify _____. (Choose three.)

A) type of gateway

B) class of gateway

C) pool of gateways

D) gatekeeper zone

E) gatekeeper class

F) gatekeeper location

Quiz Answer Key

Q1) C

Relates to: Scalable Numbering Plan

Q2) A, B, C

Relates to: Scalable Numbering Plan

Q3) 1-E, 2-B, 3-A, 4-D, 5-C

Relates to: Scalable Numbering Plan Attributes

Q4) C

Relates to: Scalable Numbering Plan Attributes

Q5) 1-D, 2-A, 3-E, 4-C, 5-B

Relates to: Hierarchical Numbering Plans

Q6) B, E

Relates to: Hierarchical Numbering Plans

Q7) C

Relates to: Internal Numbering and Public Numbering Plan Integration

Q8) B

Relates to: Internal Numbering and Public Numbering Plan Integration

Q9) B, C

Relates to: Enhancing and Extending an Existing Plan to Accommodate VoIP

Q10) A, B, C

Relates to: Enhancing and Extending an Existing Plan to Accommodate VoIP

Module 5

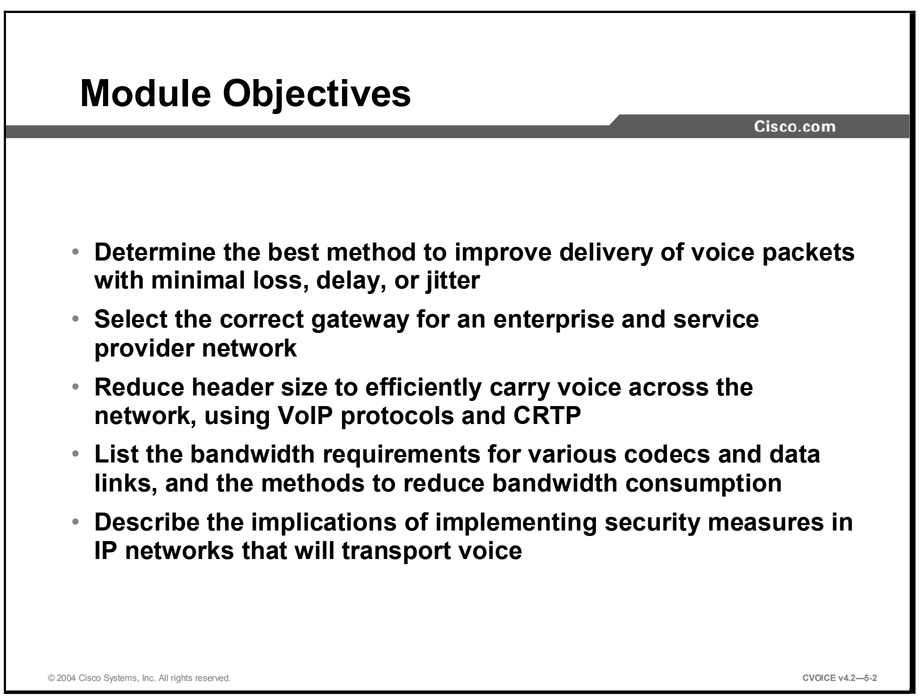
Introduction to VoIP

Overview

Voice over IP (VoIP) enables a voice-enabled router to carry voice traffic, such as telephone calls and faxes, over an IP network. This module introduces the fundamentals of VoIP, bandwidth requirements using different coder-decoders (codecs) and data links, and implementation solutions. The role of gateways and their use in integrating VoIP with traditional voice technologies is explained. Other voice over network technologies such as Frame Relay and ATM are discussed and compared to VoIP.

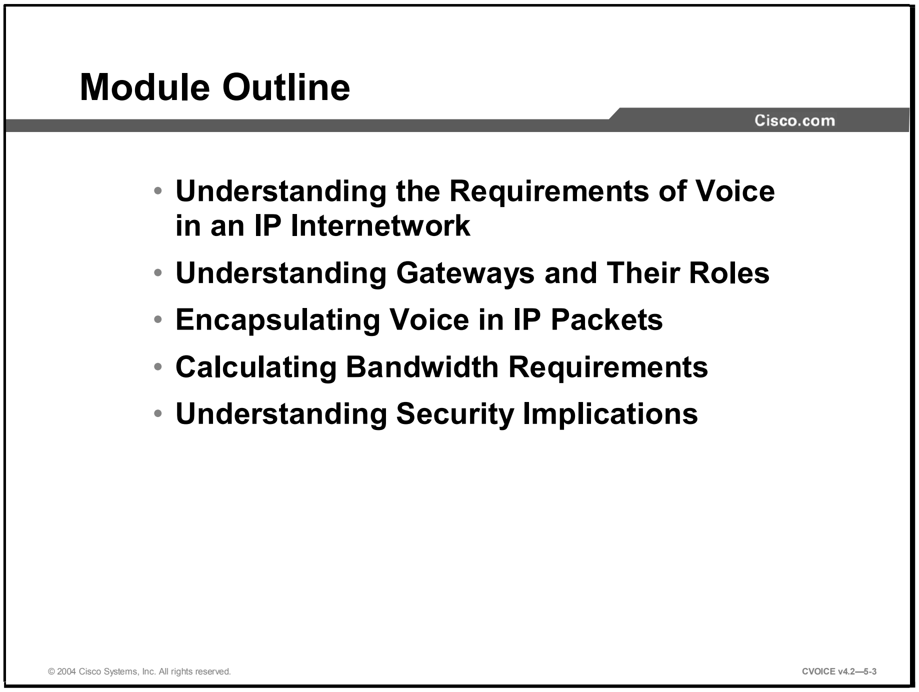
Module Objectives

Upon completing this module, you will be able to describe the fundamentals of VoIP and identify challenges and solutions regarding its implementation.



Module Outline

The outline lists the components of this module.



Understanding the Requirements of Voice in an IP Internetwork

Overview

This lesson describes the fundamentals of VoIP and identifies the challenges and solutions regarding its implementation.

Relevance

To implement VoIP solutions, you must understand how IP networks operate and what impact they have on voice traffic. Packet loss, delay, and jitter are some of the challenges voice encounters in an IP environment. This lesson will highlight these issues and discuss methods to deal with them.

Objectives

Upon completing this lesson, you will be able to determine the best method for improving delivery of voice packets with minimal loss, delay, or jitter. This includes being able to meet these objectives:

- Explain which characteristics of IP networks cause problems for real-time traffic
- Describe packet loss, delay, and jitter problems in IP networks and identify a solution for each problem
- Describe five techniques that are used by Cisco IOS software to ensure consistent delivery and throughput of voice packets in an IP network
- Illustrate the steps used by the RTP to reorder packets on IP networks
- Describe four methods for improving reliability and availability of the IP internetwork for the delivery of voice packets

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Familiarity with IP network concepts dealing with reliability, throughput, delay, and loss

Outline

The outline lists the topics included in this lesson.

Outline

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- Overview
- Real-Time Voice in a Best-Effort IP Internetwork
- Packet Loss, Delay, and Jitter
- Consistent Throughput
- Reordering of Voice Packets
- Reliability and Availability
- Summary
- Quiz
- Lab

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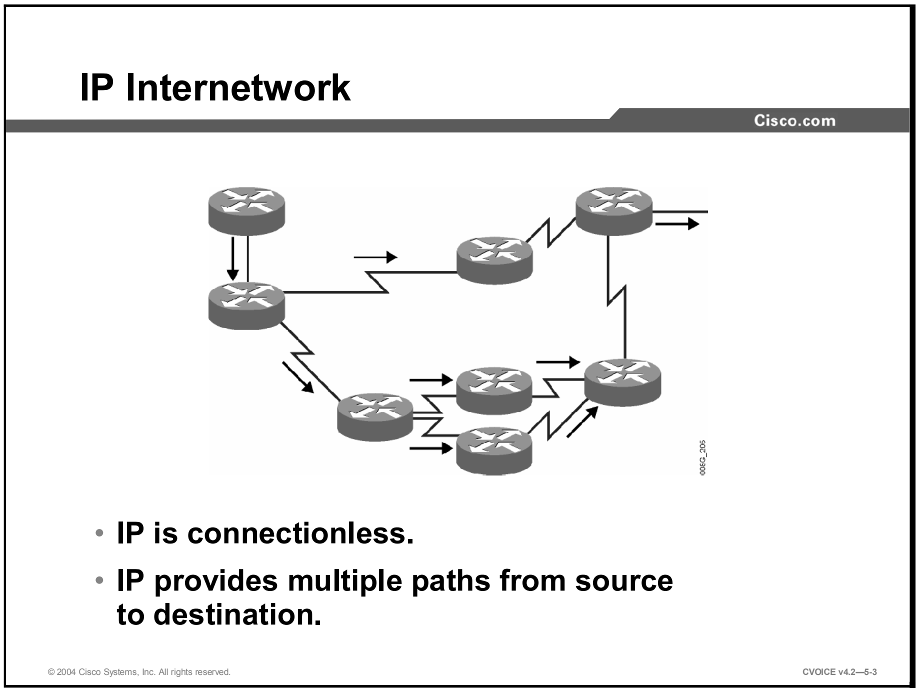
Real-Time Voice in a Best-Effort IP Internetwork

This topic lists problems associated with implementation of real-time voice traffic in a best-effort IP internetwork.



The traditional telephony network was originally designed to carry voice. The design of circuit-switched calls provides a guaranteed path and a delay threshold between source and destination. The IP network was originally designed to carry data. Data networks were not designed to carry voice traffic. Although data traffic is best-effort traffic and can withstand some amount of delay, jitter, and loss, voice traffic is real-time traffic that requires a certain quality of service (QoS). In the absence of any special QoS parameters, a voice packet is treated as just another data packet.

The user must have a well-engineered network, end to end, when running delay-sensitive applications such as VoIP. Fine-tuning the network to adequately support VoIP involves a series of protocols and features geared toward QoS.

Example: Real-Time Voice Delivery Issues

In the IP network shown in the figure, voice packets that enter the network at a constant rate can reach the intended destination by a number of routes. Because each of these routes may have different delay characteristics, the arrival rate of the packets may vary. This condition is called jitter.

Another effect of multiple routes is that voice packets can arrive out of order. The far-end voice-enabled router or gateway has to re-sort the packets and adjust the interpacket interval for a proper-sounding voice playout.

Network transmission adds corruptive effects like noise, delay, echo, jitter, and packet loss to the speech signal. VoIP is susceptible to these network behaviors, which can degrade the voice application.

If a VoIP network is to provide the same quality that users have come to expect from traditional telephony services, then the network must ensure that the delay in transmitting a voice packet across the network, and the associated jitter, does not exceed specific thresholds.

Packet Loss, Delay, and Jitter

This topic discusses the causes of packet loss, end-to-end delay, and jitter delay in an IP internetwork.

Packet Loss, Delay, and Jitter

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- **Packet loss**
  - **Loss of packets severely degrades the voice application.**
- **Delay**
  - **VoIP typically tolerates delays up to 150 ms before the quality of the call degrades.**
- **Jitter**
  - **Instantaneous buffer use causes delay variation in the same voice stream.**

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In traditional telephony networks, voice has a guaranteed delay across the network by strict bandwidth association with each voice stream. Configuring voice in a data network environment requires network services with low delay, minimal jitter, and minimal packet loss. Over the long term, packet loss, delay, and jitter will all affect voice quality, as follows:

- **Packet loss:** The IP network may drop voice packets if the network quality is poor, if the network is congested, or if there is too much variable delay in the network. Codec algorithms can correct small amounts of loss, but too much loss can cause voice clipping and skips. The chief cause of packet loss is network congestion.
- **Delay:** End-to-end delay is the time that it takes the sending endpoint to send the packet to the receiving endpoint. End-to-end delay consists of the following two components:
  - **Fixed network delay:** You should examine fixed network delay during the initial design of the VoIP network. The International Telecommunication Union (ITU) standard G.114 states that a one-way delay budget of 150 ms is acceptable for high-quality voice. Research at Cisco Systems has shown that there is a negligible difference in voice quality scores using networks built with 200-ms delay budgets. Examples of fixed network delay include propagation delay of signals between the sending and receiving endpoints, voice encoding delay, and voice packetization time for various VoIP codecs.

- **Variable network delay:** Congested egress queues and serialization delays on network interfaces can cause variable packet delays. Serialization delay is a constant function of link speed and packet size. The larger the packet and the slower the link-clocking speed, the greater the serialization delay. Although this ratio is known, it can be considered variable because a larger data packet can enter the egress queue at any time before a voice packet. If the voice packet must wait for the data packet to serialize, the delay incurred by the voice packet is its own serialization delay, plus the serialization delay of the data packet in front of it.

- **Jitter:** Jitter is the variation between the expected arrival of a packet and when it is actually received. To compensate for these delay variations between voice packets in a conversation, VoIP endpoints use jitter buffers to turn the delay variations into a constant value so that voice can be played out smoothly. Buffers can fill instantaneously, however, because network congestion can be encountered at any time within a network. This instantaneous buffer use can lead to a difference in delay times between packets in the same voice stream.

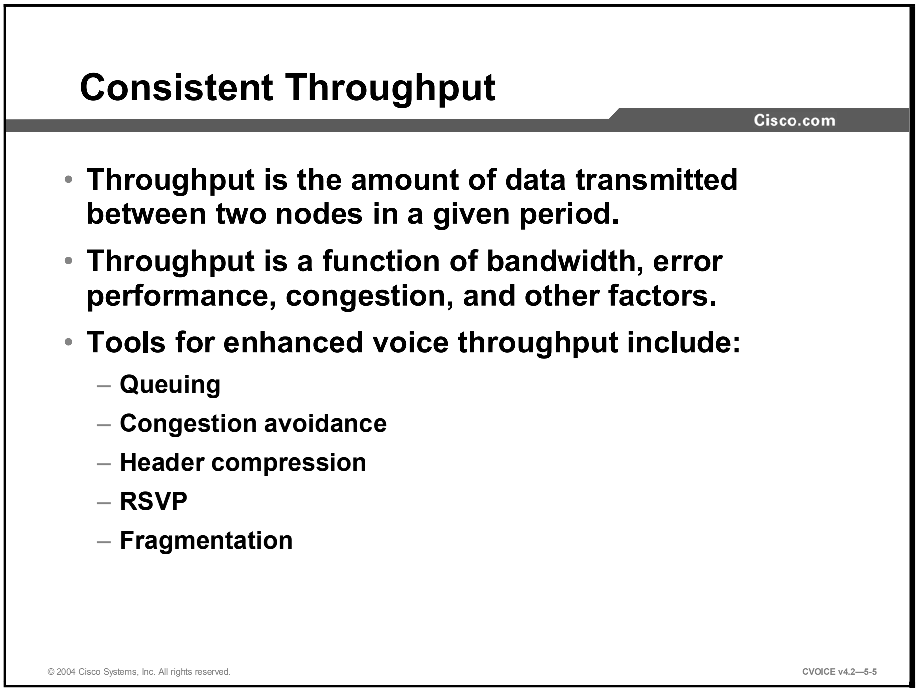
Example: Packet Loss, Delay, and Jitter Problems

The effect of end-to-end packet loss, delay, and jitter can be heard as follows:

- The calling party says, "Good morning, how are you?"
- With end-to-end delay, the called party hears, "……Good morning, how are you?"
- With jitter, the called party hears, "Good……morning, how……are you?"
- With packet loss, the called party hears, "Good m..ning, w are you?"

Consistent Throughput

This topic describes the methods that you can use to ensure consistent delivery and throughput of voice packets in an IP internetwork.



Throughput is the actual amount of useful data that is transmitted from a source to a destination. The amount of data that is placed in the pipe at the originating end is not necessarily the same amount of data that comes out at the destination. The data stream may be affected by error conditions in the network; for example, bits may be corrupted in transit, leaving the packet unusable. Packets may also be dropped during times of congestion, potentially forcing a retransmit, using twice the amount of bandwidth for that packet.

In the traditional telephony network, voice had guaranteed bandwidth associated with each voice stream. Cisco IOS software uses a number of techniques to reliably deliver real-time voice traffic across the modern data network. These techniques, which all work together to ensure consistent delivery and throughput of voice packets, include the following:

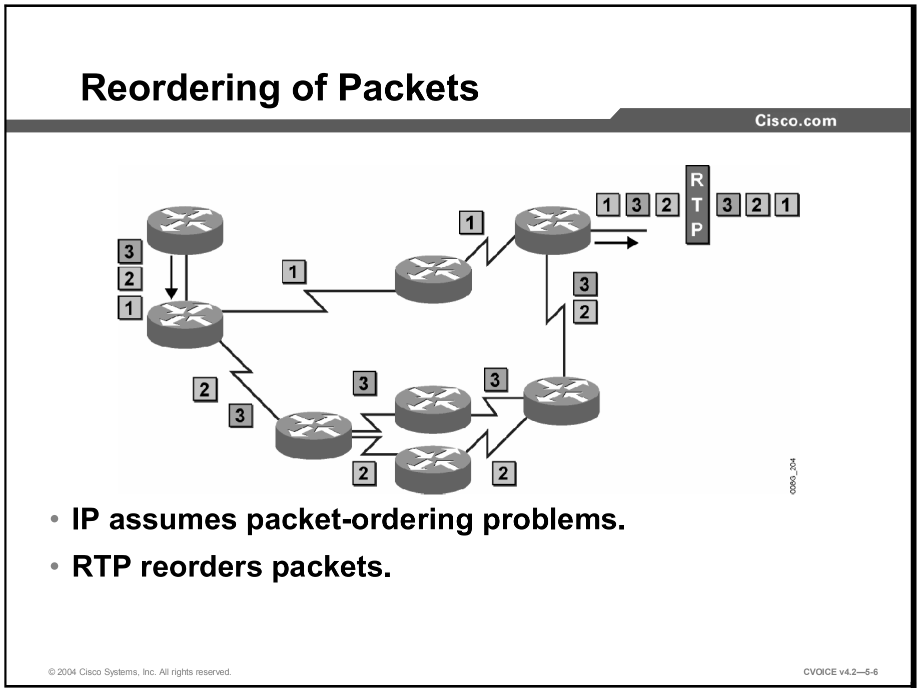
- **Queuing:** The act of holding packets so that they can be handled with a specific priority when leaving the router interface. Queuing enables routers and switches to handle bursts of traffic, measure network congestion, prioritize traffic, and allocate bandwidth. Cisco routers offer several different queuing mechanisms that can be implemented based on traffic requirements. Low Latency Queuing (LLQ) is one of the newest Cisco queuing mechanisms.
- **Congestion avoidance:** Congestion avoidance techniques monitor network traffic loads. The aim is to anticipate and avoid congestion at common network and internetwork bottlenecks before it becomes a problem. These techniques provide preferential treatment under congestion situations for premium (priority) class traffic, such as voice. At the same time, these techniques maximize network throughput and capacity use and minimize packet loss and delay. Weighted random early detection (WRED) is one of the QoS congestion avoidance mechanisms used in Cisco IOS software.

- **Header compression:** In the IP environment, voice is carried in Real-Time Transport Protocol (RTP), which is carried in User Datagram Protocol (UDP), which is then put inside an IP packet. This constitutes 40 bytes of RTP/UDP/IP header. This header size is large when compared to the typical voice payload of 20 bytes. Compressed RTP (CRTP) reduces the headers to 2 bytes in most cases, thus saving considerable bandwidth and providing for better throughput.
- **Resource Reservation Protocol:** Resource Reservation Protocol (RSVP) is a transport layer protocol that enables a network to provide differentiated levels of service to specific flows of data. Unlike routing protocols, RSVP is designed to manage flows of data rather than make decisions for each individual datagram. Data flows consist of discrete sessions between specific source and destination machines. Hosts use RSVP to request a QoS level from the network on behalf of an application data stream. Routers use RSVP to deliver QoS requests to other routers along the paths of the data stream. After an RSVP reservation is made, weighted fair queuing (WFQ) is the mechanism that actually delivers the queue space at each device. Voice calls in the IP environment can request RSVP service to provide guaranteed bandwidth for a voice call in a congested environment.
- **Fragmentation:** Fragmentation defines the maximum size for a data packet and is used in the voice environment to prevent excessive serialization delays. Serialization delay is the time that it takes to actually place the bits onto an interface; for example, a 1500-byte packet takes 187 ms to leave the router over a 64-kbps link. If a best-effort data packet of 1500 bytes is sent, real-time voice packets are queued until the large data packet is transmitted. This delay is unacceptable for voice traffic. However, if best-effort data packets are fragmented into smaller pieces, they can be interleaved with real-time (voice) packets. In this way, both voice and data packets can be carried together on low-speed links without causing excessive delay to the real-time voice traffic.

There are many QoS tools that can be used to ensure consistent throughput. When these mechanisms are employed, voice traffic on the network is assured priority and its delivery is more consistent.

Reordering of Voice Packets

This topic describes how RTP ensures consistent delivery order of voice packets in an IP internetwork.



In traditional telephony networks, voice samples are carried in an orderly manner through the use of time-division multiplexing (TDM). Because the path is circuit-switched, the path between the source and destination is reserved for the duration of the call. All of the voice samples stay in order as they are transmitted across the wire. Because IP provides connectionless transport with the possibility of multiple paths between sites, voice packets cannot arrive out of order at the destination. Because voice rides in UDP/IP packets, there is no automatic reordering of packets.

RTP provides end-to-end delivery services for data that require real-time support, such as interactive voice and video. According to RFC 1889, the services provided by RTP include payload-type identification, sequence numbering, time stamping, and delivery monitoring.

Example: Reordering Voice Packets

In the figure, RTP reorders the voice packets through the use of sequence numbers before playing them out to the user.

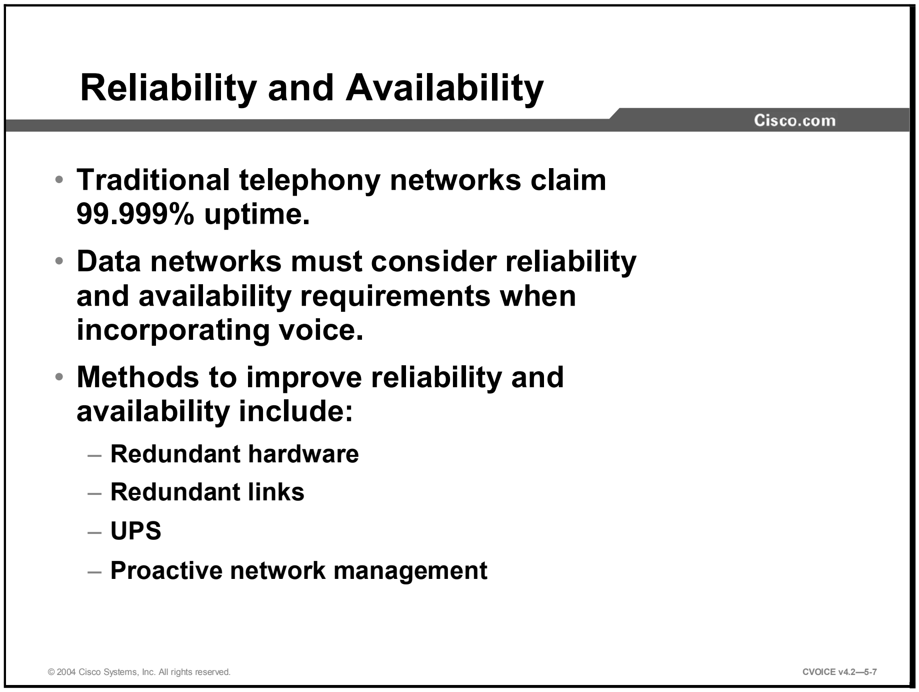
The table illustrates the various stages of packet reordering by RTP.

Sequencing of Packets by RTP

Stage	What Happens
Voice packets enter the network.	IP assumes packet-ordering problems.
RTP reorders the voice packets.	The voice packets are put in order through the use of sequence numbers.
RTP retimes the voice packets.	The voice packets are spaced according to the time stamp contained in each RTP header.
	The user hears the voice packets in order and with the same timing as when the voice stream left the source.
RTCP (Real-Time Transport Control Protocol) sends occasional report packet for delivery monitoring.	Both the sender and receiver send occasional report packets containing information, such as number of packets sent or received, the octet count, and the number of lost packets.

Reliability and Availability

The traditional telephony network strives to provide 99.999 percent uptime to the user. This corresponds to 5.25 minutes per year of downtime. Many data networks cannot make the same claim. This topic describes methods that you can use to improve reliability and availability in data networks.



To provide telephony users the same—or close to the same—level of service as they experience with traditional telephony, the reliability and availability of the data network takes on new importance.

Reliability is a measure of how resilient a network can be. Efforts to ensure reliability may include choosing hardware and software with a low mean time between failure, or installing redundant hardware and links. Availability is a measure of how accessible the network is to the users. When a user wants to make a call, for example, the network should be accessible to that user at any time a call is required. Efforts to ensure availability may include installing proactive network management to predict failures before they happen, and taking steps to correct problems in design of the network as it grows.

When the data network goes down, it may not come back up for minutes or even hours. This delay is unacceptable for telephony users. Local users with network equipment, such as voice-enabled routers, gateways, or switches for IP Phones, now find that their connectivity is terminated. Administrators must, therefore, provide an uninterruptible power supply (UPS) to these devices in addition to providing network availability. Previously, depending on the type of connection the user had, they received their power directly from the telephone company central office (CO) or through a UPS that was connected to their keyswitch or PBX in the event of a power outage. Now the network devices must have protected power to continue to function and provide power to the end devices.

Network reliability comes from incorporating redundancy into the network design. In traditional telephony, switches have multiple redundant connections to other switches. If either a link or a switch becomes unavailable, the telephone company can route the call in different ways. This is why telephone companies can claim a high availability rate.

High availability encompasses many areas of the network. In a fully redundant network, the following components need to be duplicated:

- Servers and call managers
- Access layer devices, such as LAN switches
- Distribution layer devices, such as routers or multilayer switches
- Core layer devices, such as multilayer switches
- Interconnections, such as WAN links and public switched telephone network (PSTN) gateways, even through different providers
- Power supplies and UPSs

Example: Cisco Reliability and Availability

In some data networks, a high level of availability and reliability is not critical enough to warrant financing the hardware and links required to provide complete redundancy. If voice is layered onto the network, these requirements need to be revisited.

With Cisco Architecture for Voice, Video and Integrated Data (AVVID) technology, the use of Cisco CallManager clusters provides a way to design redundant hardware in the event of Cisco CallManager failure. When using gatekeepers, you can configure backup devices as secondary gatekeepers in case the primary gatekeeper fails. You must also revisit the network infrastructure. Redundant devices and Cisco IOS services, like Hot Standby Router Protocol (HSRP), can provide high availability. For proactive network monitoring and trouble reporting, a network management platform such as CiscoWorks2000 provides a high degree of responsiveness to network issues.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- IP networks need to use QoS parameters and protocols to adequately support VoIP.
- The characteristics of IP contribute to problems with voice traffic, including packet loss, delay, and jitter.
- Consistent throughput is enhanced by queuing, congestion avoidance, header compression, RSVP, and fragmentation.
- RTP provides payload-type identification, sequence numbering, time stamping, and delivery monitoring on IP networks.
- Redundant hardware and links, UPS, and proactive network management improve reliability and availability of data networks.

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Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 5-1: Basic VoIP

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) What is one cause of jitter when voice packets are transmitted over an IP network?

A) congestion leading to instantaneous buffer utilization

B) the corruptive effect of noise

C) the loss of some voice packets on high-traffic lines

D) use of the PSTN

Q2) What can be used to ensure proper delivery of voice packets over an IP network?

A) QoS parameters

B) Frame Relay as the Layer 2 technology

C) circuit-switched paths only

D) high-speed links

Q3) According to ITU G.114, what is the maximum delay that VoIP can tolerate before voice quality starts to degrade?

A) 120 ms

B) 150 ms

C) 200 ms

D) 250 ms

Q4) Which component in an IP telephony network can compensate for small amounts of packet loss?

A) the playout buffer

B) the codec

C) the codebook

D) the look-ahead buffer

Q5) Match the Cisco IOS tools for enhanced voice throughput with their description.

A) queuing

B) congestion avoidance

C) header compression

D) RSVP

E) fragmentation

_____ 1. provides preferential treatment for priority traffic

_____ 2. holds packets so that they can be handled with a specific priority leaving the router interface

_____ 3. defines the maximum size of a data packet

_____ 4. enables the network to provide differentiated levels of service to specific flows of data

_____ 5. saves bandwidth by compressing the header

Q6) Which technique is used to prevent excessive serialization delay?

A) queuing

B) congestion avoidance

C) header compression

D) RSVP

E) fragmentation

Q7) Which protocol automatically reorders packets?

A) TDM

B) IP

C) UDP

D) RTP

Q8) What is the function of RTCP in ensuring consistent delivery of voice packets in an IP network?

A) reorders the voice packets through the use of sequence numbers

B) retimes the voice packets based on time stamps

C) sends occasional report packets for delivery monitoring

D) provides a suite of monitoring tools

Q9) Which Cisco technology can be used to provide proactive network management?

A) Cisco IOS services

B) Cisco AVVID

C) Cisco CallManager

D) CiscoWorks2000

Q10) What do traditional telephony networks typically use to provide redundancy?

A) multiple servers

B) multilayer switches

C) multiple connections between switches

D) auto-configuring switches

Quiz Answer Key

Q1) A

Relates to: Packet Loss, Delay, and Jitter

Q2) A

Relates to: Real-Time Voice in a Best-Effort IP Internetwork

Q3) B

Relates to: Packet Loss, Delay, and Jitter

Q4) B

Relates to: Packet Loss, Delay, and Jitter

Q5) 1-B, 2-A, 3-E, 4-D, 5-C

Relates to: Consistent Throughput

Q6) E

Relates to: Consistent Throughput

Q7) D

Relates to: Reordering of Voice Packets

Q8) C

Relates to: Reordering of Voice Packets

Q9) D

Relates to: Reliability and Availability

Q10) C

Relates to: Reliability and Availability

Understanding Gateways and Their Roles

Overview

This lesson describes the role of gateways in integrating VoIP with the traditional voice technologies that are found in enterprise and service provider networks.

Relevance

Support for protocols, signaling capabilities, voice features, and voice applications is changing and growing quickly. Gateways play an important role in providing access to the right mix of functionality. You must understand the main features and functions required in enterprise and service provider environments to choose the appropriate gateway.

Objectives

Upon completing this lesson, you will be able to select the correct gateway for an enterprise and service provider network. This includes being able to meet these objectives:

- Describe the role of gateways and their application when connecting VoIP to traditional PSTN and telephony equipment
- Select the correct gateway to connect VoIP to traditional PSTN and telephony equipment
- Determine gateway interconnection requirements in an enterprise environment
- Describe three gateway interconnection requirements in service provider environments

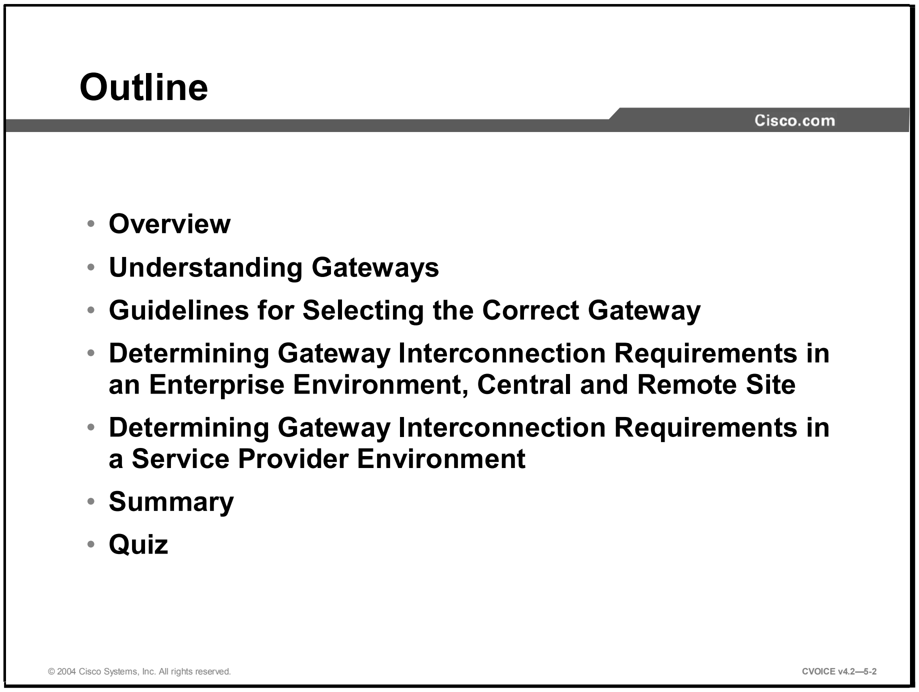
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Understanding of telephony signaling
- Understanding the basic functions of voice gateways

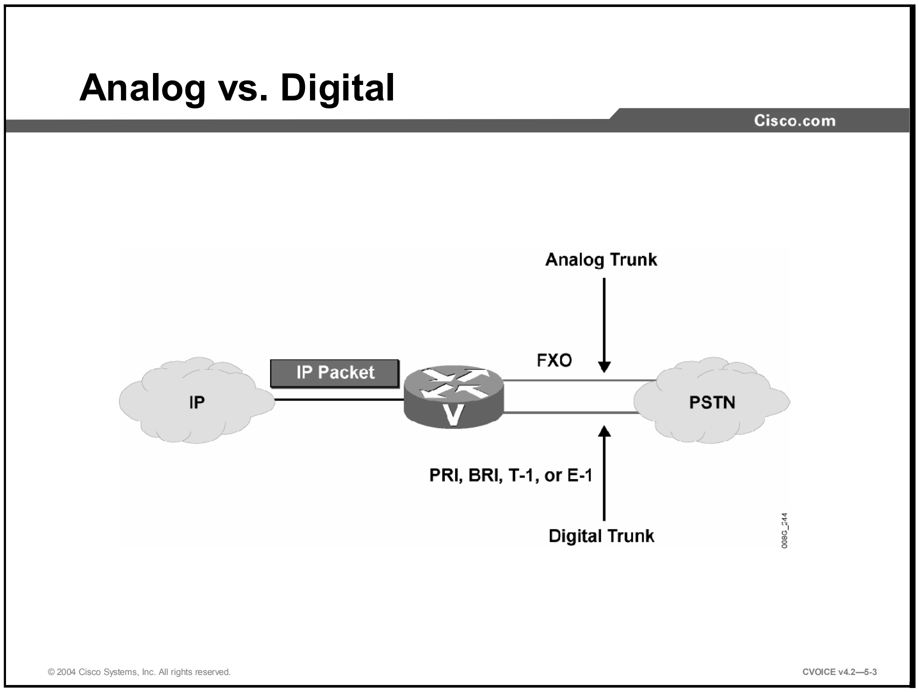
Outline

The outline lists the topics included in this lesson.



Understanding Gateways

This topic describes the role of voice gateways and their application when connecting VoIP to traditional PSTN and telephony equipment.



A gateway is a device that translates one type of signal to a different type of signal. There are different types of gateways, including the voice gateway.

A voice gateway is a router or switch that converts IP voice packets to analog or digital signals that are understood by TDM trunks or stations. Gateways are used in several situations; for example, to connect the PSTN, a PBX, or a key system to a VoIP network.

Example: Analog and Digital Gateways

In the figure, the voice-enabled router examines the incoming IP packet to determine if it is a voice packet and where it is heading. Based on information inside the voice packet, the router translates the digitized signal or voice into the appropriate analog or digital signal to be sent to the PSTN. For a call coming from the PSTN, the gateway interprets the dialed digits and determines the IP destination for this call.

Guidelines for Selecting the Correct Gateway

This topic describes the guidelines for selecting the correct gateway.

Gathering the Requirements

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- Is an analog or digital gateway required?
- What is the required capacity of the gateway?
- What type of connection is the gateway going to use? Is Foreign Exchange Office (FXO), FXS, E&M, T1, E1, PRI, or BRI signaling required?
- What signaling protocol is used? H.323, Media Gateway Control Protocol (MGCP), or session initiation protocol (SIP)?
- Is voice compression a part of the design? If so, which type?
- Are direct inward dialing (DID), calling line identification (CLID), modem relay, or fax relay required?
- Is the device acting only as gateway or as gateway and router/LAN switch? Is inline power for IP Phones required?
- Is remote site survivability required?
- To which country is the hardware shipped?

 CVOICE v4.2—5-4

Understanding gateways and being able to select the correct gateway out of numerous gateway options is challenging. Factors to consider include the protocols that are supported, the density and types of interfaces on the gateway, and the features that are required. Knowing the requirements will guide you to the correct solution.

One criterion involves defining the type of site that the gateway supports. Is it a small office/home office (SOHO), branch office, enterprise campus environment, or service provider? Each type of site has its own set of requirements.

The figure lists the questions that you should be asking before selecting a gateway. The answers will help to define the gateway functions and determine if the proposed design meets current requirements and encompasses future growth.

A key step is identifying the number and type of voice interfaces that are necessary and verifying the protocol support. Are supplementary services supported? Which codecs must be supported? Is fax relay necessary? Many of these functions are features of specific Cisco IOS software releases. Identification of the proper IOS software release that is necessary to support the features is critical.

Another key question is whether the gateway is acting as a gateway only or needs to combine the functions of gateway and router within one device. This, too, points to a specific set of hardware and software.

When planning gateways for location in other countries, verify that the device meets the government standards for PSTN connection in that country. Also, if the device supports encryption capabilities, verify the legality of export to the destination country.

Example: Selecting a Gateway

For example, if the requirements are to support Foreign Exchange Station (FXS) and recEive and transMit (E&M) connections, as well as T1 PRI from the PBX, then a suitable choice would be a Cisco 3745 Multiservice Access Router with a two-slot voice network module (VNM), 1 FXS voice interface card (VIC), 1 E&M VIC, and a High Density Voice (HDV) module.

Determining Gateway Interconnection Requirements in an Enterprise Environment, Central and Remote Site

This topic describes the guidelines for determining gateway interconnection requirements in an enterprise environment for both central and remote sites.

Enterprise Gateway Considerations—Remote Site

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	Role	Features	Platforms
General	• Branch office router	• WAN • QoS • Router • Security	• 17xx • 26/36/37
Voice Gateway (GW)	• Standalone GW • Branch office router – GW – SRST – LAN switch – Optional gatekeeper	• QoS • Voice interfaces • Voice features • LAN switching • Branch survivability	• 17XX • VG224 • VG248 • 26/36/37 • Cat 6k
Additional	• Analog fax/modem GW • Digital fax/modem	• Fax • Modem	• Same as voice GWs

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As IP telephony services become a standard in the corporate environment, a broad mix of requirements surface in the enterprise environment. The IP telephony deployment typically begins by connecting to the PSTN to manage off-net calls and using a Cisco CallManager infrastructure to manage on-net calls.

Example: Gateway Interconnect Considerations

The table shows examples of questions that you must ask to determine the requirements for gateway interconnections.

Determining Gateway Interconnection Requirements

Question	Reasoning
How do you control the gateways?	You must ensure support for proper call processing, such as Media Gateway Control Protocol (MGCP), session initiation protocol (SIP), or H.323.
Is cost an issue?	Distributed call processing is easier to implement, but costs are higher when deploying intelligent devices at each site.
Is remote site survivability an issue?	Remote site survivability is not an issue with a distributed model unless there is a need for redundancy. This is an issue for a centralized model that must be addressed by providing Survivable Remote Site Telephony (SRST). This means ensuring that the version of Cisco IOS software supports the feature.
Are gatekeepers in the design, and if so, how are the zones structured?	Gatekeepers are normally used in enterprise sites for scalability and manageability. The design must include proper planning for zone configurations.
Are the gateways switches or routers?	This question determines how other features, such as QoS, are implemented. Numerous switches and routers are available that have voice gateway functionality along with other core services. These services include Layer 2 and Layer 3 QoS implementations, inline power, and security features.
Is fax or modem support required?	This requirement means the gateway must be capable of fax and modem relay functions. Another option for the enterprise customer may be to purchase IP telephony services from a service provider. In that case, a decision must be made regarding who manages the gateway and what type of connection is required; for example, SIP, H.323, or MGCP.

Enterprise Gateway Considerations— Central Site

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- Dial plan integration
- Voice-mail integration
- Gateway for PBX interconnect
- Inline power requirements for IP Phones

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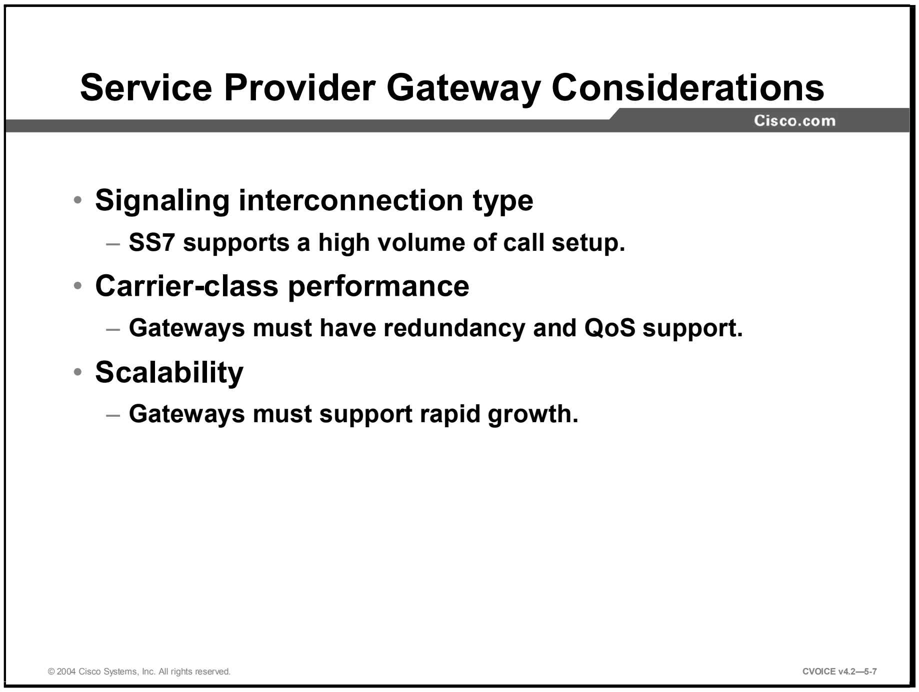
At the central site, the specific issues that need to be addressed include the following:

- **Dial plan integration:** For consistent reachability, the new dial plan for the IP voice network must integrate with the existing dial plan. It is essential that you have a thorough understanding of how the dial plans interact.
- **Voice-mail integration:** After a voice-mail application is selected, the designer must ensure that all users can seamlessly reach the voice-mail server and that all incoming calls are properly forwarded when the recipient does not answer the telephone. This may mean dedicating gateway connections for an existing voice mail server, or dedicating an entire gateway for the express purpose of voice mail server integration.
- **Gateway for PBX interconnect:** When the IP voice network interconnects PBXs, the designer must determine what type of connection is supported by the PBX and which gateway will support that connection.
- **Inline power requirements for IP Phones:** Beyond the gateway, when the design includes IP Phones, the power requirements must be considered. In many cases, it is desirable to provide inline power to the telephones. A number of devices provide inline power. The decision about inline power requirements is based on capacity and the current power options.

Note The network administrator should evaluate the need for inline power depending on the network design.

Determining Gateway Interconnection Requirements in a Service Provider Environment

This topic provides the gateway interconnection requirements in service provider environments.



Service providers must provide a level of service that meets or exceeds PSTN standards. The gateways that service providers implement must provide for reliable, high-volume voice traffic with acceptable levels of latency and jitter. The following functions address those requirements:

- **Signaling interconnection type:** Signaling System 7 (SS7) interconnect supports a high volume of call setup and benefits from redundant interconnect capabilities directly into the PSTN switch network.
- **Carrier-class performance:** Carrier-class performance can be provided through the proper redundant design for high availability in addition to the proper implementation of QoS features to ensure acceptable delay and jitter.
- **Scalability:** Scalability is a critical factor in the service provider arena. Customers who need access should be serviced promptly. Choosing a gateway with capacity for rapid growth is an important design decision. Gateways can scale upward to T3 capabilities for large-scale environments.

Example: Service Provider Requirements

An IP telephony service provider needs to upgrade their existing gateway platforms because of business growth. The service provider sells a managed IP telephony service to small and medium businesses and provides connections to many different low-cost, long-distance carriers for their customers. Their issues are call quality over the IP network, so delay and jitter need to be controlled. Service providers also must consider scalability and the ability to provide differentiated levels of service through QoS. They also need connectivity to SS7 networks of long-distance carriers to reduce costs, and, finally, they need to consider the overall cost of implementation. SS7 capabilities and a redundant design enable the service provider to deliver a reliable level of service.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- A voice gateway converts IP voice packets to analog or digital signals.
- When you are selecting a gateway, you must consider what protocols are supported, the density and types of interfaces on the gateway, and the features that are required.
- Enterprise interconnection design issues include distributed versus centralized call processing, SRST, QoS, and fax/modem relay requirements.
- Service provider interconnection requirements typically include use of SS7 to signal the PSTN, carrier-class performance, and scalability.

 CVOICE v4.2—5-8

References

For additional information, refer to these resources:

- *Product Bulletin, No. 1596*

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) In which of the following situations would a gateway be required?

A) a CallManager-to-IP network connection

B) a PBX-to-voice mail connection

C) a key system-to-IP network connection

D) a PBX-to-PBX connection

Q2) The gateway translates the digitized signal or voice into the appropriate analog or digital signal for the PSTN, based on which of the following?

A) configuration of the gateway

B) protocols being used on the network

C) default settings of the gateway

D) information inside the voice packet

Q3) Give two reasons why is it important to know which country the gateway will be located in when you are planning a network. (Choose two.)

A) network requirements

B) standards for PSTN connections

C) design requirements

D) functions required of the gateway

E) legal issues involving encryption services

Q4) Which three signaling protocols can be used with gateways? (Choose three.)

A) H.323

B) MGCP

C) SIP

D) SCCP

E) MEGACO

Q5) Which two issues need to be addressed when selecting a gateway for an enterprise central site? (Choose two.)

A) signaling interconnection type

B) carrier-class performance

C) scalability

D) inline power requirements for IP Phones

E) dial plan integration

F) voice-mail integration

Q6) Why is it important to determine if the gateways are switches or routers?

A) It dictates how many gateways to deploy.

B) It determines how QoS, inline power, and security features are to be implemented.

C) It affects zone configurations.

D) It determines the Cisco IOS version that should be used.

Q7) Which three features need to be considered when selecting a gateway for a service provider network? (Choose three.)

A) high-volume call setup

B) high availability

C) scalability

D) SIP support

E) interoperability with customer applications

Q8) If implemented properly, which two features of a service provider gateway can provide carrier-class performance? (Choose two.)

A) scalability

B) redundant design planning

C) prompt service for customers

D) implementation of QoS features

E) support for a high volume of call setup

Q9) When a call arrives from the PSTN, how does the gateway know where to send the call?

A) from the IP address included in the call

B) from the dialed digits in the call

C) from the calling digits included in the call

D) from the CLID information in the call

Q10) Which three features will influence the type of gateway that will be required for an enterprise remote site? (Choose three.)

A) the routing protocol the LAN and WAN need to support

B) the type of QoS required

C) the number of analog phones or modems that will need support

D) the level of security that will need to be supported

E) the type of network management that will need to be supported

Quiz Answer Key

Q1) C

Relates to: Understanding Gateways

Q2) D

Relates to: Understanding Gateways

Q3) B, E

Relates to: Guidelines for Selecting the Correct Gateway

Q4) A, B, C

Relates to: Guidelines for Selecting The Correct Gateway

Q5) E, F

Relates to: Determining Gateway Interconnection Requirements in an Enterprise Environment, Central and Remote Site

Q6) B

Relates to: Determining Gateway Interconnection Requirements in an Enterprise Environment, Central and Remote Site

Q7) A, B, C

Relates to: Determining Gateway Interconnection Requirements in a Service Provider Environment

Q8) B, D

Relates to: Determining Gateway Interconnection Requirements in a Service Provider Environment

Q9) B

Relates to: Understanding Gateways

Q10) B, C, D

Relates to: Determining Gateway Interconnection Requirements in an Enterprise Environment, Central and Remote Site

Encapsulating Voice in IP Packets

Overview

This lesson discusses the VoIP protocol stack, its applied headers, and the use of CRTP to reduce header size.

Relevance

A number of protocols and tools are used to successfully carry voice in a data network. In defining the VoIP protocol stack, you must understand at which layer these tools and protocols reside and how they interact with other layers. When packaging voice into IP packets, additional headers are created to carry voice-specific information. These headers can create significant additional overhead in the IP network. Understanding which protocols to use and knowing how to limit overhead is crucial in carrying voice efficiently across the network.

Objectives

Upon completing this lesson, you will be able to reduce header size to efficiently carry voice across the network, using VoIP protocols and CRTP. This includes being able to meet these objectives:

- Define the major VoIP protocols and how they map to the seven layers of the OSI model
- Describe the functions of RTP and RTCP as they relate to a VoIP network
- Describe how IP voice headers are compressed using CRTP
- Identify three conditions that necessitate the compression of the RTP header

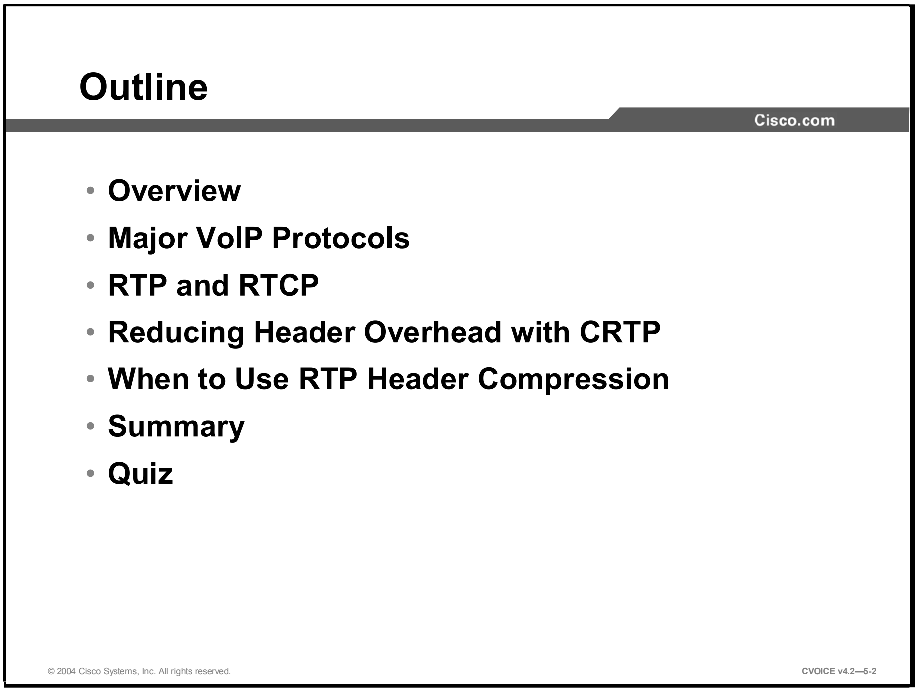
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- General knowledge of the seven-layer Open System Interconnection (OSI) model
- General knowledge of header compression

Outline

The outline lists the topics included in this lesson.



Major VoIP Protocols

This topic defines the major VoIP protocols and matches them with the seven layers of the OSI model.

Major VoIP Protocols

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VoIP Protocol	Description
H.323	ITU standard protocol for interactive conferencing. Evolved from H.320 ISDN standard. Flexible, complex.
MGCP	Emerging Internet Engineering Task Force (IETF) standard for PSTN gateway control, thin device control.
SIP	IETF protocol for interactive and noninteractive conferencing. Simpler, but less mature, than H.323.
RTP	IETF standard media streaming protocol.
RTCP	IETF protocol that provides out-of-band control information for an RTP flow.

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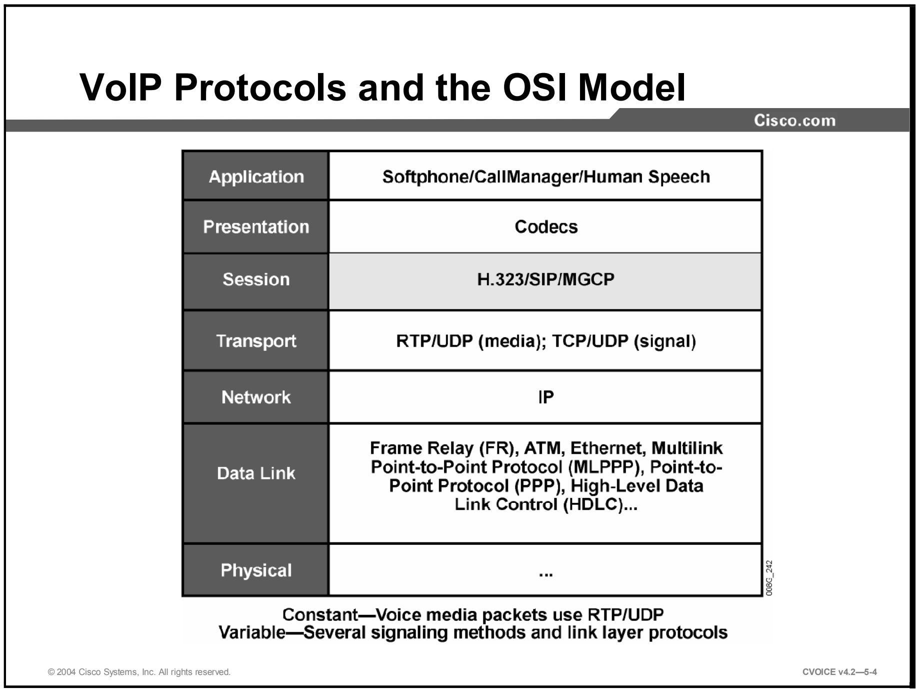
The major VoIP protocols include the following:

- **H.323:** An ITU standard protocol for interactive conferencing. The ITU standard protocol was originally designed for multimedia in a connectionless environment, such as a LAN. The H.323 is an umbrella of standards that defines all aspects of synchronized voice, video, and data transmission. H.323 defines end-to-end call signaling.
- **MGCP:** An emerging standard for PSTN gateway control or thin device control. Specified in RFC 2705, MGCP defines a protocol to control VoIP gateways connected to external call-control devices, referred to as call agents. MGCP provides the signaling capability for less expensive edge devices, such as gateways, that may not contain a full voice-signaling stack, such as H.323. In essence, any time an event such as off hook occurs at the voice port of a gateway, the voice port reports that event to the call agent. The call agent then signals that device to provide a service, such as dial-tone signaling.
- **SIP:** A detailed protocol that specifies the commands and responses to set up and tear down calls. It also details features such as security, proxy, and transport (TCP or UDP) services. SIP and its partner protocols, Session Announcement Protocol (SAP) and Session Description Protocol (SDP), provide announcements and information about multicast sessions to users on a network. SIP defines end-to-end call signaling between devices. SIP is a text-based protocol that borrows many elements of HTTP, using the same transaction request and response model, and similar header and response codes. It also adopts a modified form of the URL-addressing scheme used within e-mail that is based on Simple Mail Transfer Protocol (SMTP).
- **RTP:** An Internet Engineering Task Force (IETF) standard media-streaming protocol. RTP carries the voice payload across the network. RTP provides sequence numbers and time stamps for the orderly processing of voice packets.

- **RTCP:** Provides out-of-band control information for an RTP flow. Every RTP flow has a corresponding RTCP flow that reports statistics on the call. RTCP is used for QoS reporting.



Example: VoIP and the OSI Model

Successfully integrating connection-oriented voice traffic in a connectionless-oriented IP network requires enhancements to the signaling stack. In some ways, the user must make the connectionless network appear more connection-oriented.

Applications such as Cisco IP Softphone and Cisco CallManager provide the interface for users to originate voice at their PCs or laptops and convert and compress it before passing it to the network. If a gateway is used, a standard telephone becomes the interface to users, so human speech is the application.

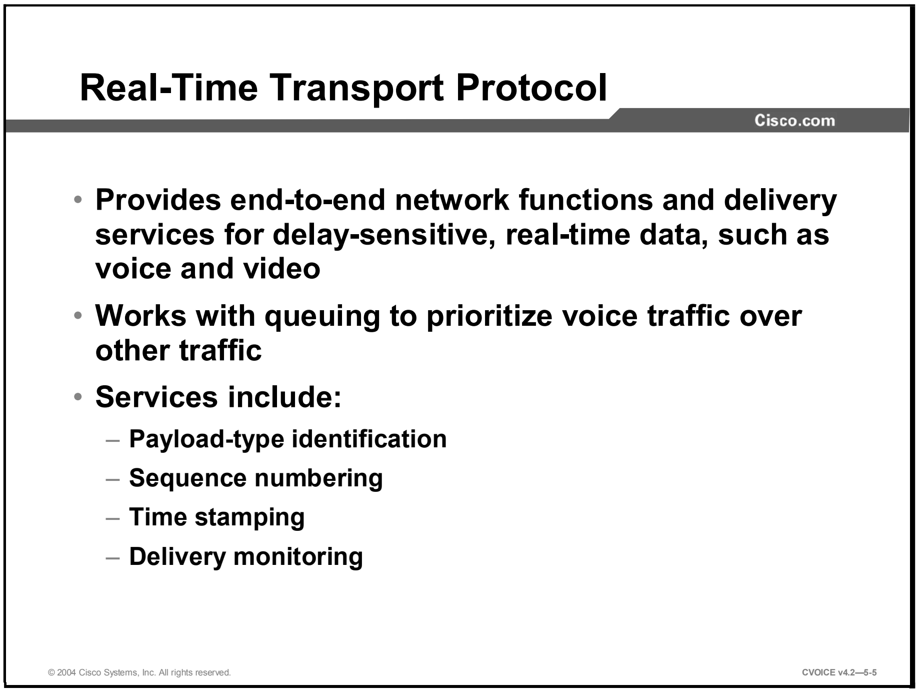
Codecs define how the voice is compressed. The user can configure which codec to use or a codec is negotiated according to what is available.

One of the constants in VoIP implementation is that voice uses RTP inside of UDP to carry the payload across the network. Because IP voice packets can reach the destination out of order and unsynchronized, the packets must be reordered and resynchronized before playing them out to the user. Since UDP does not provide services such as sequence numbers or time stamps, RTP provides sequencing functionality.

The variables in VoIP are the signaling methods used. H.323 and SIP define end-to-end call-signaling methods. MGCP defines a method to separate the signaling function from the voice call function. MGCP uses a call agent to control signaling on behalf of the endpoint devices, such as gateways. The central control device participates in the call setup only. Voice traffic still flows directly from endpoint to endpoint.

RTP and RTCP

This topic describes the functions of RTP and RTCP as they relate to the VoIP network.



RTP provides end-to-end network transport functions intended for applications transmitting real-time requirements, such as audio and video. Those functions include payload-type identification, sequence numbering, time stamping, and delivery monitoring.

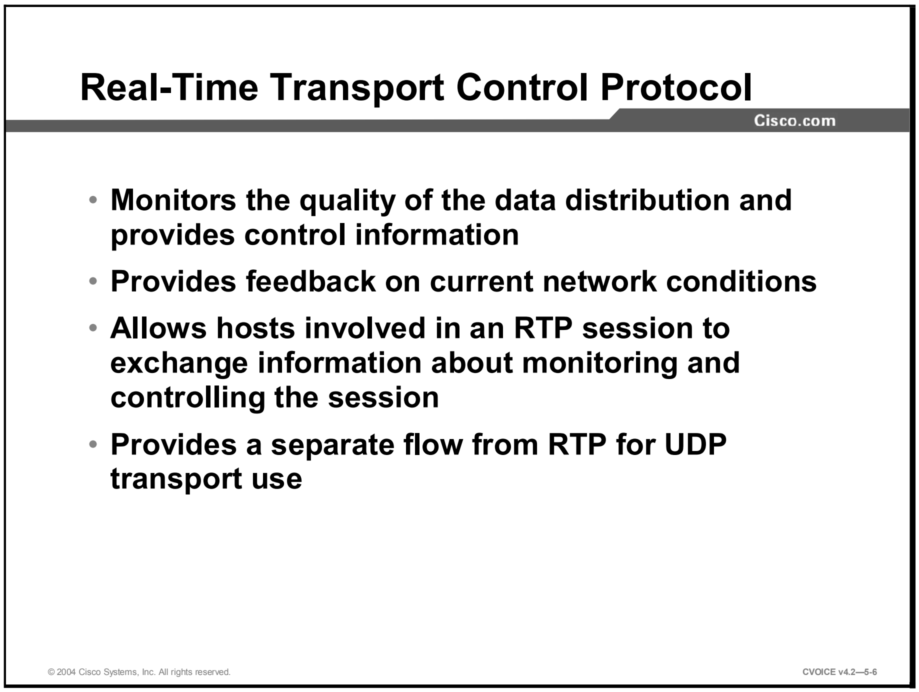
RTP typically runs on top of UDP to use the multiplexing and checksum services of that protocol. Although RTP is often used for unicast sessions, it is primarily designed for multicast sessions. In addition to the roles of sender and receiver, RTP also defines the roles of translator and mixer to support the multicast requirements.

RTP is a critical component of VoIP because it enables the destination device to reorder and retime the voice packets before they are played out to the user. An RTP header contains a time stamp and sequence number, which allows the receiving device to buffer and remove jitter and latency by synchronizing the packets to play back a continuous stream of sound. RTP uses sequence numbers to order the packets only. RTP does not request retransmission if a packet is lost.

Example: RTP Application

As voice packets are placed on the network to reach a destination, they may take one or more paths to reach their destination. Each path may have a different length and transmission speed, resulting in the packets being out of order when they arrive at their destination. As the packets were placed on the wire at the source of the call, RTP tagged the packets with a time stamp and sequence number. At the destination, RTP can reorder the packets and send them to the digital signal processor (DSP) at the same pace as they were placed on the wire at the source.

Note	For more information on RTP, refer to RFC 1889.



RTCP monitors the quality of the data distribution and provides control information. RTCP provides the following feedback on current network conditions:

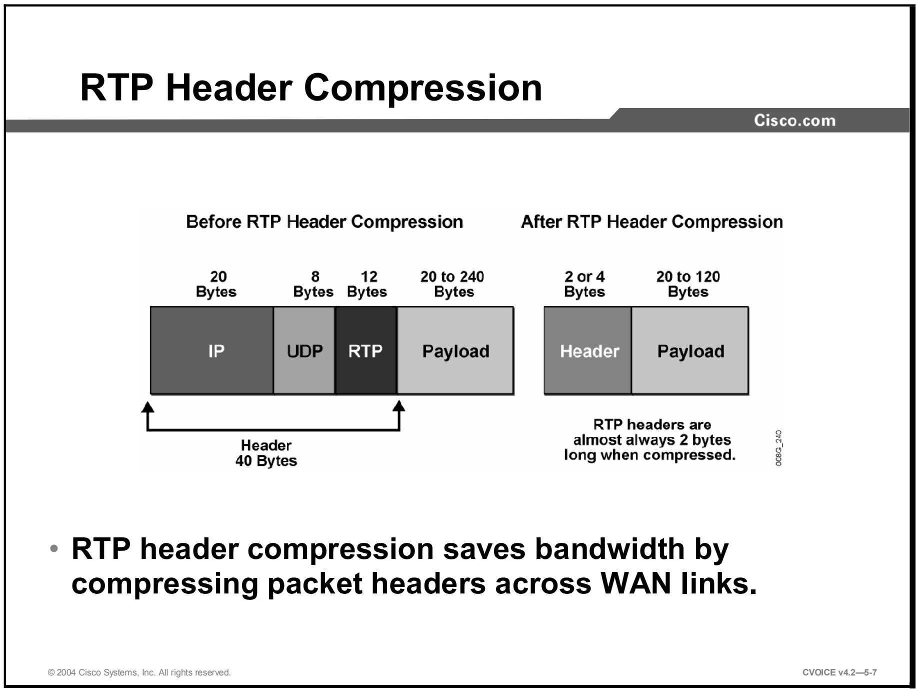
- RTCP provides a mechanism for hosts involved in an RTP session to exchange information about monitoring and controlling the session. RTCP monitors the quality of elements such as packet count, packet loss, delay, and interarrival jitter. RTCP transmits packets as a percentage of session bandwidth, but at a specific rate of at least every 5 seconds.
- The RTP standard states that the Network Time Protocol (NTP) time stamp is based on synchronized clocks. The corresponding RTP time stamp is randomly generated and based on data-packet sampling. Both NTP and RTP are included in RTCP packets by the sender of the data.
- RTCP provides a separate flow from RTP for transport use by UDP. When a voice stream is assigned UDP port numbers, RTP is typically assigned an even-numbered port and RTCP is assigned the next odd-numbered port. Each voice call has four ports assigned: RTP plus RTCP in the transmit direction and RTP plus RTCP in the receive direction.

Example: RTCP Application

Throughout the duration of each RTP call, the RTCP report packets are generated at least every 5 seconds. In the event of poor network conditions, a call may be disconnected due to high packet loss. When viewing packets using a packet analyzer, a network administrator could check information in the RTCP header that includes packet count, octet count, number of packets lost, and jitter. The RTCP header information would shed light on why the calls were disconnected.

Reducing Header Overhead with CRTP

This topic describes how IP voice headers are compressed using CRTP.



Given the number of protocols that are necessary to transport voice over an IP network, the packet header can be large. You can use CRTP headers on a link-by-link basis to save bandwidth.

Using CRTP compresses the IP/UDP/RTP header from 40 bytes to 2 bytes without UDP checksums and from 40 bytes to 4 bytes with UDP checksums. RTP header compression is especially beneficial when the RTP payload size is small; for example, with compressed audio payloads between 20 and 50 bytes.

In addition, CRTP works on the premise that most of the fields in the IP/UDP/RTP header do not change, or that the change is predictable. Static fields include source and destination IP address, source and destination UDP port numbers, as well as many other fields in all three headers. For those fields where the change is predictable, the CRTP process is illustrated in the following table:

CRTP

Stage	What Happens
The change is predictable.	The sending side tracks the predicted change.
The predicted change is tracked.	The sending side sends a hash of the header.
The receiving side predicts what the constant change is.	The receiving side substitutes the original stored header and calculates the changed fields.
There is an unexpected change.	The sending side sends the entire header without compression.

CRTP Packet Components

In a packet voice environment when speech samples are framed every 20 ms, a payload of 20 bytes is generated. Without CRTP, the total packet size includes the following components:

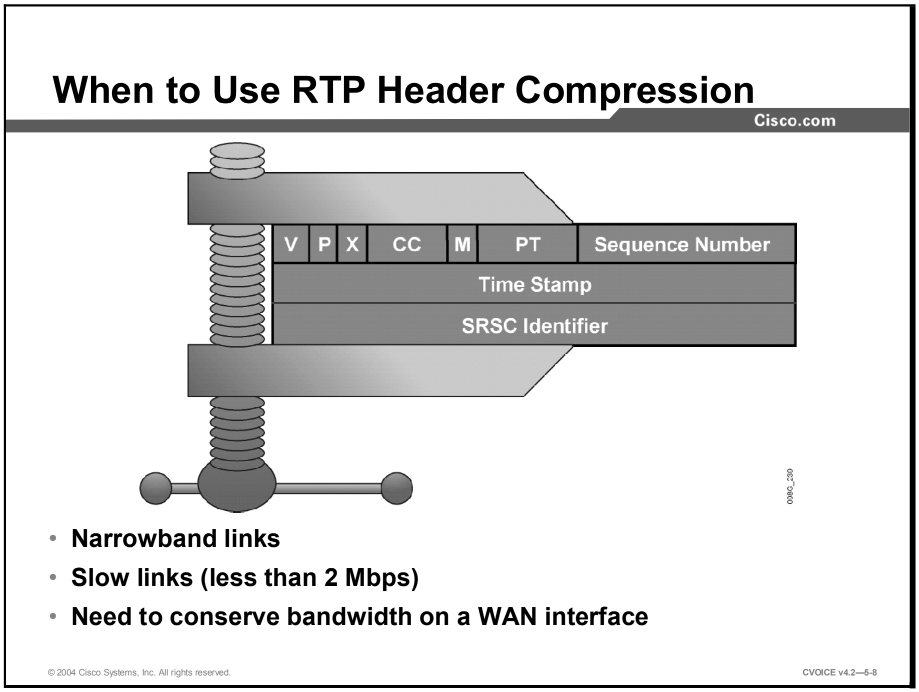
- IP header (20 bytes)
- UDP header (8 bytes)
- RTP header (12 bytes)
- Payload (20 bytes)

The header is twice the size of the payload: IP/UDP/RTP (20 + 8 + 12 = 40 bytes) versus payload (20 bytes). When generating packets every 20 ms on a slow link, the header consumes a large portion of bandwidth.

In the figure, RTP header compression reduces the header to 2 bytes. The compressed header is one tenth of the payload size.

When to Use RTP Header Compression

This topic describes when to use CRTP.



You must configure CRTP on a specific serial interface or subinterface if you have any of these conditions:

- Narrowband links
- Slow links (less than 2 Mbps)
- Need to conserve bandwidth on a WAN interface

Compression works on a link-by-link basis and must be enabled for each link that fits these requirements. You must enable compression on both sides of the link for proper results. Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if there is a significant volume of RTP traffic on that slow link.

Note Compression adds to processing overhead. You must check resource availability on each device prior to turning on RTP header compression.

Example: Applying CRTP

If you want the router to compress RTP packets, use the **ip rtp header-compression** command. The **ip rtp header-compression** command defaults to active mode when it is configured. However, this command provides a passive mode setting in instances where you want the router to compress RTP packets *only* if it has received compressed RTP on that interface. When applying to a Frame Relay interface, use the **frame-relay ip rtp header-compression** command.

By default, the software supports a total of 16 RTP header compression connections on an interface. Depending on the traffic on the interface, you can change the number of header compression connections with the **ip rtp compression-connections** *number* command.

Note	Do not use CRTP if you have high-speed interfaces or links faster than 2 Mbps.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Major VoIP protocols include H.323 and SIP for end-to-end call control, MGCP for PSTN gateway control, RTP for media streaming, and RTCP for out-of-band control information for RTP flow.
- RTP carries packetized audio traffic over an IP network.
- RTCP provides feedback on the quality of the call, including statistics on packet loss, delay, and jitter. RTP header compression compresses the IP/UDP/RTP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes most of the time.
- RTP header compression is useful if you are running VoIP over narrowband or slow links, or if you need to conserve bandwidth on a WAN interface.

 CVOICE v4.2—5-9

References

For additional information, refer to these resources:

- *Request for Comments*
  ftp://www.ietf.org/internet-drafts/
- *Configuring Compressed Real-Time Protocol*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/12cgcr/qos_c/qcpart6/qc rtphc.htm

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Match the VoIP protocol with its function.

A) H.323

B) MGCP

C) SIP

D) RTP

E) RTCP

_____ 1. adopts a modified form of the URL addressing scheme used within e-mail based on SMTP

_____ 2. allows control of VoIP gateways connected to external call control devices

_____ 3. reports statistics on the call

_____ 4. defines all aspects of synchronized voice, video, and data transmission

_____ 5. provides sequence numbers and time stamps for orderly processing of voice packets

Q2) Which VoIP protocol operates at the transport layer of the OSI model?

A) H.323

B) MGCP

C) SIP

D) RTP

Q3) Which of the three are components of RTP? (Choose three.)

A) monitors jitter

B) payload-type identification

C) sequence numbering

D) time stamping

E) monitors packet count

F) monitors packet loss

Q4) Which Layer 4 protocol is used for transporting RTCP packets?

A) IP

B) TCP

C) UDP

D) RTP

Q5) What is the size of the IP/UDP/RTP header?

A) 20 bytes

B) 32 bytes

C) 40 bytes

D) 48 bytes

Q6) To what size does CRTP compress the IP/UDP/RTP header when UDP checksums are used?

A) 2 bytes

B) 4 bytes

C) 8 bytes

D) 12 bytes

Q7) Which two conditions necessitate the use of CRTP? (Choose two.)

A) need for bandwidth conservation on WAN interfaces

B) high-speed interfaces

C) links that are as slow as 4 Mbps

D) narrowband links

E) broadband links

Q8) On which two areas should compression be enabled? (Choose two.)

A) each link that requires it

B) the entire network

C) both sides of a slow link

D) links with high traffic

E) all links using RTP

Quiz Answer Key

Q1) 1-C, 2-B, 3-E, 4-A 5-D

Relates to: Major VoIP Protocols

Q2) D

Relates to: Major VoIP Protocols

Q3) B, C, D

Relates to: RTP and RTCP

Q4) C

Relates to: RTP and RTCP

Q5) C

Relates to: Reducing Header Overhead with CRTP

Q6) B

Relates to: Reducing Header Overhead with CRTP

Q7) A, D

Relates to: When to Use RTP Header Compression

Q8) A, C

Relates to: When to Use RTP Header Compression

Calculating Bandwidth Requirements

Overview

This lesson describes, in detail, the bandwidth requirements for VoIP. Several variables affecting total bandwidth are explained, as well as the method of calculating and reducing total bandwidth.

Relevance

Since WAN bandwidth is probably the most expensive component of an enterprise network, network administrators must know how to calculate the total bandwidth required for voice traffic and how to reduce overall consumption.

Objectives

Upon completing this lesson, you will be able to list the bandwidth requirements for various codecs and data links, and describe the methods to reduce bandwidth consumption. This includes being able to meet these objectives:

- List five types of codecs and their associated bandwidth requirements
- Describe how the number of voice samples that are encapsulated impacts bandwidth requirements
- List the overhead for various Layer 2 protocols
- Describe how IPSec and GRE/L2TP affect bandwidth overhead
- Describe how MPLS, MLP, and other technologies affect bandwidth overhead
- Use a formula to calculate the total bandwidth that is required for a VoIP call
- Describe the operation of, and bandwidth savings associated with, the use of VAD

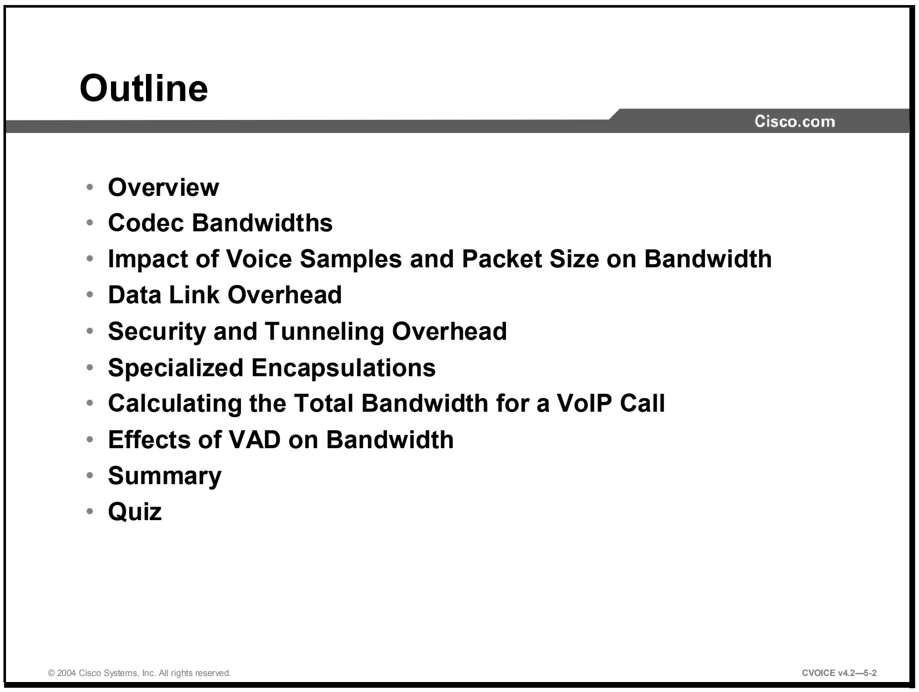
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Basic knowledge of VoIP
- Knowledge of TCP/IP networks
- Knowledge of Layer 2 technologies such as Frame Relay and ATM
- Knowledge of voice compression standards

Outline

The outline lists the topics included in this lesson.



Codec Bandwidths

This topic describes the bandwidth that each codec uses and illustrates its impact on total bandwidth.

Bandwidth Implications of Codec

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Codec	G.711	G.726 r32	G.726 r24	G.726 r16	G.728	G.729	G.723 r63	G.723 r53
Bandwidth	64 kbps	32 kbps	24 kbps	16 kbps	16 kbps	8 kbps	6.3 kbps	5.3 kbps

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CVOICE v4.2—5-3

One of the most important factors for the network administrator to consider while building voice networks is proper capacity planning. Network administrators must understand how much bandwidth is used for each VoIP call. With a thorough understanding of VoIP bandwidth, the network administrator can apply capacity-planning tools.

Following is a list of codecs and their associated bandwidth:

- **G.711:** The G.711 pulse code modulation (PCM) coding scheme uses the most bandwidth. It takes samples 8000 times per second, each of which is 8 bits in length, for a total of 64,000 bps.
- **G.726:** The G.726 adaptive differential pulse code modulation (ADPCM) coding schemes use somewhat less bandwidth. While each coding scheme takes samples 8000 times per second like PCM, it uses 4, 3, or 2 bits for each sample, thereby resulting in total bandwidths of 32,000, 24,000, or 16,000 bps.
- **G.728:** The G.728 low-delay code excited linear prediction (LDCELP) coding scheme compresses PCM samples using codebook technology. It uses a total bandwidth of 16,000 bps.
- **G.729:** The G.729 and G.729A Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP) coding scheme also compresses PCM using advanced codebook technology. It uses 8000 bps total bandwidth.
- **G.723:** The G.723 and G.723A multipulse maximum likelihood quantization (MPMLQ) coding schemes use a look-ahead algorithm. These compression schemes result in 6300 or 5300 bps.

The network administrator should balance the need for voice quality against the cost of bandwidth in the network when choosing codecs. The higher the codec bandwidth, the higher the cost of each call across the network.

Impact of Voice Samples and Packet Size on Bandwidth

This topic illustrates the effect of voice sample size on bandwidth.

Impact of Voice Samples

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Codec	Bandwidth	Sample Size	Packets
G.711	64000	240	33
G.711	64000	160	50
G.726r32	32000	120	33
G.726r32	32000	80	50
G.726r24	24000	80	25
G.726r24	24000	60	33
G.726r16	16000	80	25
G.726r16	16000	40	50
G.728	16000	80	13
G.728	16000	40	25
G.729	8000	40	25
G.729	8000	20	50
G.723r63	6300	48	16
G.723r63	6300	24	33
G.723r53	5300	40	17
G.723r53	5300	20	33

 CVOICE v4.2—5-4

Voice sample size is a variable that can affect total bandwidth used. A voice sample is defined as the digital output from a codec DSP that is encapsulated into a protocol data unit (PDU). Cisco uses DSPs that output samples based on digitization of 10 ms-worth of audio. Cisco voice equipment encapsulates 20 ms of audio in each PDU by default, regardless of the codec used. You can apply an optional configuration command to the dial peer to vary the number of samples encapsulated. When you encapsulate more samples per PDU, total bandwidth is reduced. However, encapsulating more samples per PDU comes at the risk of larger PDUs, which can cause variable delay and severe gaps if PDUs are dropped.

Example: Encapsulated Bytes Calculation

Using a simple formula, it is possible for you to determine the number of bytes encapsulated in a PDU based on the codec bandwidth and the sample size (20 ms is default):

Bytes_per_Sample = (*Sample_Size* ∗ *Codec_Bandwidth*) / 8

If you apply G.711 numbers, the formula reveals the following:

Bytes_per_Sample = (.020 ∗ 64000) / 8

Bytes_per_Sample = 160

The figure illustrates various codecs and sample sizes and the number of packets that are required for VoIP to transmit one second of audio. The larger the sample size, the larger the packet, and the fewer the encapsulated samples that have to be sent (which reduces bandwidth).

Data Link Overhead

This topic lists overhead sizes for various Layer 2 protocols.

Data Link Overhead

Cisco.com

- Ethernet
  - 18 bytes overhead
- MLP
  - 6 bytes overhead
- Frame Relay
  - 6 bytes overhead

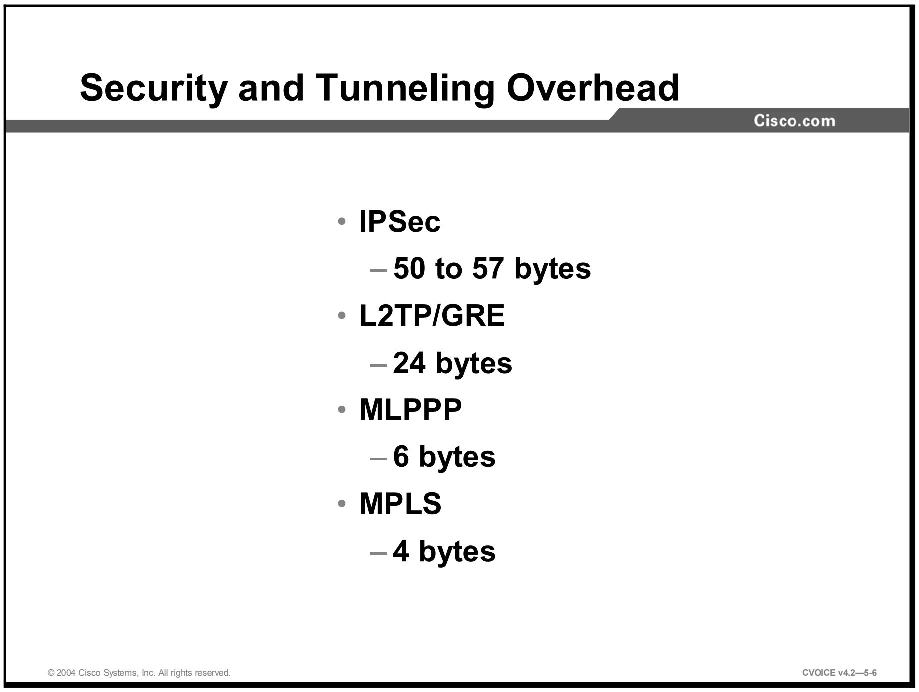
CVOICE v4.2—5-5

Another contributing factor to bandwidth is the Layer 2 protocol used to transport VoIP. VoIP alone carries a 40-byte IP/UDP/RTP header, assuming uncompressed RTP. Depending on the Layer 2 protocol used, the overhead could grow substantially. The larger the Layer 2 overhead, the more bandwidth required to transport VoIP. The following points illustrate the Layer 2 overhead for various protocols:

- **Ethernet II:** Carries 18 bytes of overhead; 6 bytes for source MAC, 6 bytes for destination MAC, 2 bytes for type, and 4 bytes for cyclic redundancy check (CRC)
- **Multilink Point-to-Point Protocol (MLP):** Carries 6 bytes of overhead; 1 byte for flag, 1 byte for address, 2 bytes for control (or type), and 2 bytes for CRC
- **FRF.12:** Carries 6 bytes of overhead; 2 bytes for data-link connection identifier (DLCI) header, 2 bytes for FRF.12, and 2 bytes for CRC

Security and Tunneling Overhead

This topic describes overhead associated with various security and tunneling protocols.



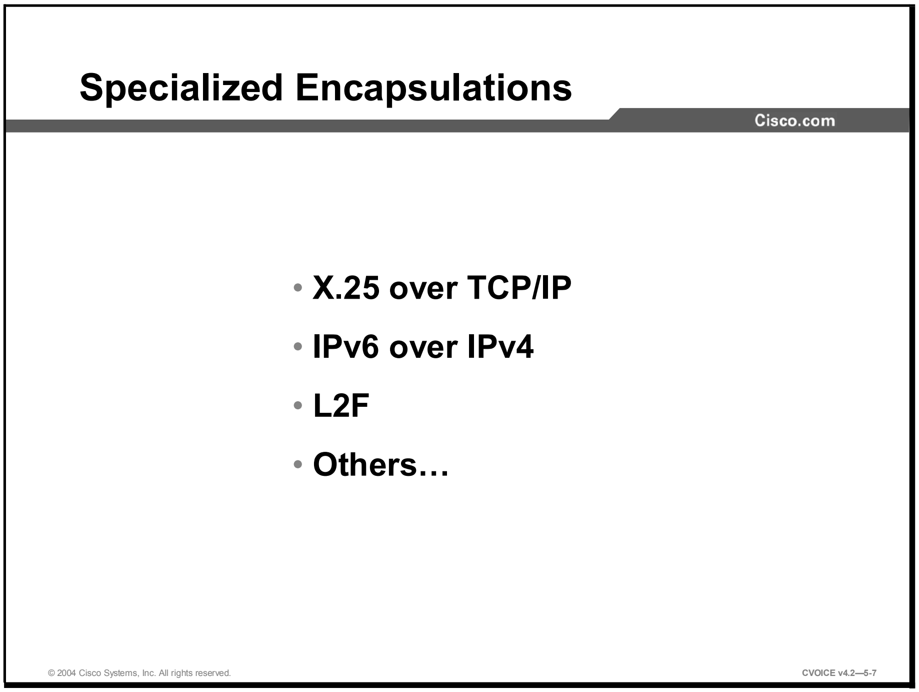
Certain security and tunneling encapsulations will also add overhead to voice packets and should be considered when calculating bandwidth requirements. When using a Virtual Private Network (VPN), IP Security (IPSec) will add 50 to 57 bytes of overhead, a significant amount when considering small voice packets. Layer 2 Tunneling Protocol/generic routing encapsulation (L2TP/GRE) adds 24 bytes. When using MLP, 6 bytes will be added to each packet. Multiprotocol Label Switching (MPLS) adds a 4-byte label to every packet. All of these specialized tunneling and security protocols must be considered when planning for bandwidth demands.

Example: VPN Overhead

Many companies have their employees telecommute from home. These employees initiate a VPN connection into their enterprise for secure Internet transmission. When deploying a remote telephone at the employee's home using a router and a PBX Off-Premises eXtension (OPX), the voice packets will experience additional overhead associated with the VPN.

Specialized Encapsulations

This topic describes considerations for specialized encapsulations for VoIP.



There exist many other encapsulations to consider when transporting VoIP. Specialized encapsulations include protocol-specific encapsulation such as X.25, experimental encapsulations such as IPv6 over IPv4, Layer 2 Forwarding (L2F) Protocol, and other vendor-specific encapsulations. Each must be considered when calculating total bandwidth.

Calculating the Total Bandwidth for a VoIP Call

This topic calculates the total bandwidth required for a VoIP call using codec, data link, and sample size.

Total Bandwidth Required

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Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with CRTP	Ethernet	Ethernet with CRTP
G.711	64000	240	76267	66133	78933	68800
G.711	64000	160	82400	67200	86400	71200
G.726r32	32000	120	44267	34133	46933	36800
G.726r32	32000	80	50400	35200	54400	39200
G.726r24	24000	80	37800	26400	40800	29400
G.726r24	24000	60	42400	27200	46400	31200
G.726r16	16000	80	25200	17600	27200	19600
G.726r16	16000	40	34400	19200	38400	23200
G.728	16000	80	25200	17600	27200	19600
G.728	16000	40	34400	19200	38400	23200
G.729	8000	40	17200	9600	19200	11600
G.729	8000	20	26400	11200	30400	15200
G.723r63	6300	48	12338	7350	13650	8663
G.723r63	6300	24	18375	8400	21000	11025
G.723r53	5300	40	11395	6360	12720	7685
G.723r53	5300	20	17490	7420	20140	10070

 CVOICE v4.2—5-8

Codec choice, data-link overhead, sample size, and compressed RTP have positive and negative impacts on total bandwidth. To perform the calculations, you must consider these contributing factors as part of the equation:

- More bandwidth required for the codec = more total bandwidth required
- More overhead associated with the data link = more total bandwidth required
- Larger sample size = less total bandwidth required
- Compressed RTP = significantly reduced total bandwidth required

Example: Total Bandwidth Calculation

The following calculation was used to produce the figure:

Total_Bandwidth = ([Layer_2_Overhead + IP_UDP_RTP Overhead + Sample_Size] / Sample_Size) * Codec_Speed

For example, assume a G.729 codec, 20-byte sample size, using Frame Relay without CRTP:

Total_Bandwidth = ([6 + 40 +20]/20) * 8000

Total_Bandwidth = 26,400 bps

Effects of VAD on Bandwidth

This topic describes the effect of voice activity detection (VAD) on total bandwidth.

Effect of VAD

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Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with VAD
G.711	64000	240	76267	49573
G.711	64000	160	82400	53560
G.726r32	32000	120	44267	28773
G.726r32	32000	80	50400	32760
G.726r24	24000	80	37800	24570
G.726r24	24000	60	42400	27560
G.726r16	16000	80	25200	16380
G.726r16	16000	40	34400	22360
G.728	16000	80	25200	16380
G.728	16000	40	34400	22360
G.729	8000	40	17200	11180
G.729	8000	20	26400	17160
G.723r63	6300	48	12338	8019
G.723r63	6300	24	18375	11944
G.723r53	5300	40	11395	7407
G.723r53	5300	20	17490	11369

 CVOICE v4.2—5-9

On average, an aggregate of 24 calls or more may contain 35 percent silence. With traditional telephony voice networks, all voice calls use 64-kbps fixed-bandwidth links regardless of how much of the conversation is speech and how much is silence. In Cisco VoIP networks, all conversations and silences are packetized. VAD suppresses packets of silence. Instead of sending VoIP packets of silence, VoIP gateways interleave data traffic with VoIP conversations to more effectively use network bandwidth.

VAD provides a maximum of 35 percent bandwidth savings based on an average volume of more than 24 calls.

Note Bandwidth savings of 35 percent is an average figure and does not take into account loud background sounds, differences in languages, and other factors.

The savings are not realized on every individual voice call, or on any specific point measurement.

Note For the purposes of network design and bandwidth engineering, VAD should *not* be taken into account, especially on links that will carry fewer than 24 voice calls simultaneously.

Various features, such as music on hold (MOH) and fax, render VAD ineffective. When the network is engineered for the full voice call bandwidth, all savings provided by VAD are available to data applications.

VAD is enabled by default for all VoIP calls. VAD reduces the silence in VoIP conversations but it also provides comfort noise generation (CNG). Because you can mistake silence for a disconnected call, CNG provides locally generated *white noise* to make the call appear normally connected to both parties.

Example: VAD Bandwidth Savings

The figure shows examples of the VAD effect in a Frame Relay VoIP environment. In the example using G.711 with a 160-byte payload, the bandwidth required is 82,400 bps. By turning VAD on, you can reduce the bandwidth utilization to 53,560 bps. This is a bandwidth savings of 35 percent.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Different codecs have different bandwidth requirements.
- Voice sample size affects the bandwidth that is required.
- Overhead in Layer 2 protocols affects the bandwidth used.
- Security and tunneling adds significant overhead.
- Specialized encapsulations add overhead.
- Codec, Layer 2 protocol, sample size, and VAD must all be used when calculating VoIP bandwidth.
- VAD may lower bandwidth use up to 35 percent.

CVOICE v4.2—5-10

References

For additional information, refer to these resources:

- *Voice over IP – Per Call Bandwidth Consumption*
  http://www.cisco.com/warp/public/788/pkt-voice-general/bwidth_consume.html
- *Voice over IP (VoIP) Frame Relay (VoFR)*
  http://www.cisco.com/pcgi-bin/Support/browse/psp_view.pl?p=Technologies:VoX:VoFR
- *Voice over ATM (VoATM)*
  http://www.cisco.com/pcgi-bin/Support/browse/psp_view.pl?p=Technologies:VoX:VoATM

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which type of codec has the lowest bandwidth requirement?

A) G.711

B) G.723

C) G.726

D) G.728

E) G.729

Q2) Match the codec with the coding scheme it uses.

A) G.711

B) G.726

C) G.728

D) G.729

E) G.723

_____ 1. ADPCM

_____ 2. CS-ACELP

_____ 3. LDCELP

_____ 4. MPMLQ

_____ 5. PCM

Q3) What is the disadvantage of encapsulating more samples per PDU?

A) Total bandwidth is reduced.

B) The delay becomes more variable.

C) More bandwidth is required.

D) The speed of the interface is reduced.

Q4) What would be the size of voice samples from Cisco voice equipment if a G.728 codec were used?

A) 10 ms

B) 20 ms

C) 24 ms

D) 40 ms

Q5) What is the overhead for Frame Relay?

A) 3 bytes

B) 5 bytes

C) 6 bytes

D) 18 bytes

Q6) How many bytes in the Ethernet II overhead are used for CRC?

A) 1 byte

B) 2 bytes

C) 4 bytes

D) 6 bytes

Q7) What is the overhead associated with IPSec?

A) 50 to 57 bytes

B) 24 bytes

C) 4 to 6 bytes

D) 6 bytes

Q8) What is the overhead associated with MPLS?

A) 50 to 57 bytes

B) 24 bytes

C) 4 bytes

D) 6 bytes

Q9) Which two are examples of protocol-specific encapsulation? (Choose two.)

A) X.25 over TCP/IP

B) IPv6 over IPv4

C) L2F

D) L2TP

Q10) Which two are examples of experimental encapsulation? (Choose two.)

A) X.25 over TCP/IP

B) IPv6 over IPv4

C) L2F

D) L2TP

Q11) Which three factors must be considered when calculating the total bandwidth of a VoIP call? (Choose three.)

A) codec size

B) CRC usage

C) network-link overhead

D) sample size

E) capacity of network links

Q12) Using a G.729 codec and Frame Relay without CRTP, what is the total bandwidth required for a 40-byte voice sample size?

A) 9600 bps

B) 11600 bps

C) 17200 bps

D) 19200 bps

Q13) What is the function of comfort noise generation?

A) provides features such as MOH

B) provides white noise to make the call sound connected

C) provides full voice call bandwidth

D) reduces the delay in VoIP connections

Q14) If the bandwidth requirement for a voice call over Frame Relay is 11395 bps, what would be the bandwidth requirement if VAD was used?

A) 3989 bps

B) 7407 bps

C) 9116 bps

D) 11180 bps

Q15) Which of the following formulas is used to calculate the total bandwidth?

A) Total_Bandwidth = [Layer_2_Overhead + IP_UDP_RTP Overhead + Sample_Size] * Codec_Speed

B) Total_Bandwidth = ([Layer_2_Overhead + IP_UDP_RTP Overhead + Sample_Size] / Sample_Size) * Codec_Speed

C) Total_Bandwidth = ([Layer_3_Overhead + IP_UDP_RTP Overhead] / Sample_Size) * Codec_Speed

D) Total_Bandwidth = ([Layer_2_Overhead + IP_UDP_RTP Overhead + Sample_Size] / Sample_Size)

E) Total_Bandwidth = ([Layer_3_Overhead + IP_UDP_RTP Overhead + Sample_Size] / Sample_Size) * Codec Speed

Quiz Answer Key

Q1) B

Relates to: Codec Bandwidths

Q2) 1-B, 2-D, 3-C, 4-E, 5-A

Relates to: Codec Bandwidths

Q3) B

Relates to: Impact of Voice Samples and Packet Size on Bandwidth

Q4) B

Relates to: Impact of Voice Samples and Packet Size on Bandwidth

Q5) C

Relates to: Data Link Overhead

Q6) C

Relates to: Data Link Overhead

Q7) A

Relates to: Security and Tunneling Overhead

Q8) C

Relates to: Security and Tunneling Overhead

Q9) A, B

Relates to: Specialized Encapsulations

Q10) B, C

Relates to: Specialized Encapsulations

Q11) A, C, D

Relates to: Calculating the Total Bandwidth for a VoIP Call

Q12) C

Relates to: Calculating the Total Bandwidth for a VoIP Call

Q13) B

Relates to: Effects of VAD on Bandwidth

Q14) B

Relates to: Effects of VAD on Bandwidth

Q15) B

Relates to: Calculating the Total Bandwidth for a VoIP Call

Understanding Security Implications

Overview

This lesson describes the implications of implementing security measures in IP networks that transport voice.

Relevance

Security is a top priority in most networks. The many types of security solutions include router access lists, stateful firewalls, and VPNs. These solutions may be standalone or layered. Implementing voice in a secure network environment requires an understanding of potential issues and in-depth knowledge of existing security measures and how they affect the transit of voice through the network.

Objectives

Upon completing this lesson, you will be able to describe the implications of implementing security measures in IP networks that will transport voice. This includes being able to meet these objectives:

- Describe three elements of a security policy that are necessary for a VoIP network
- Describe how the Cisco SAFE Blueprint details best practices for secure VoIP
- Describe the dynamic access control process used by firewalls to allow voice packets to pass
- Outline the steps needed to reduce overhead and delay for VoIP in a VPN
- Describe how to calculate the bandwidth required for a VPN packet

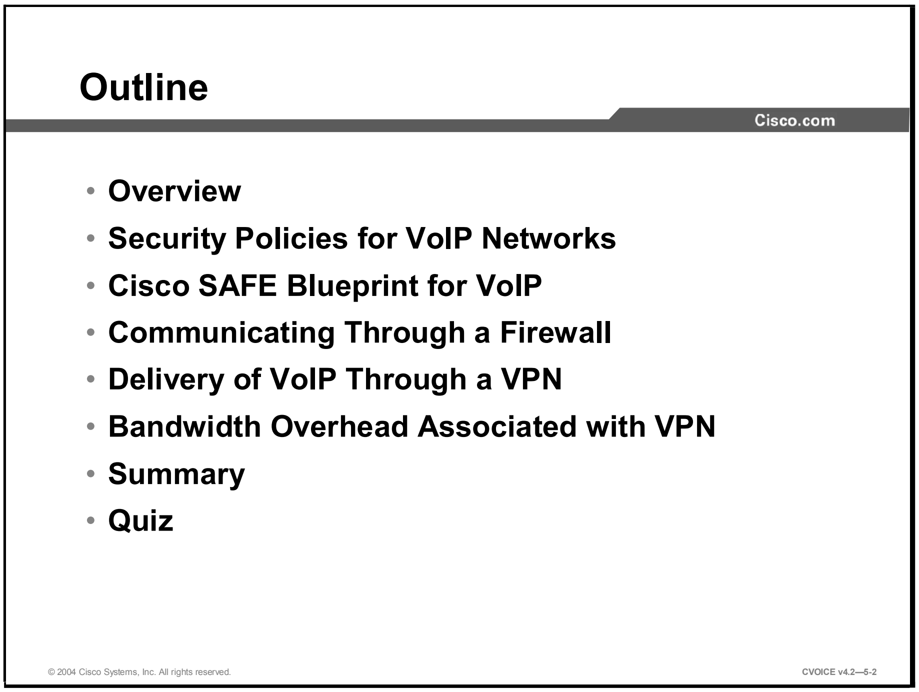
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- General knowledge of VPNs
- General knowledge of firewalls
- Understanding of voice requirements, including delay and port numbering

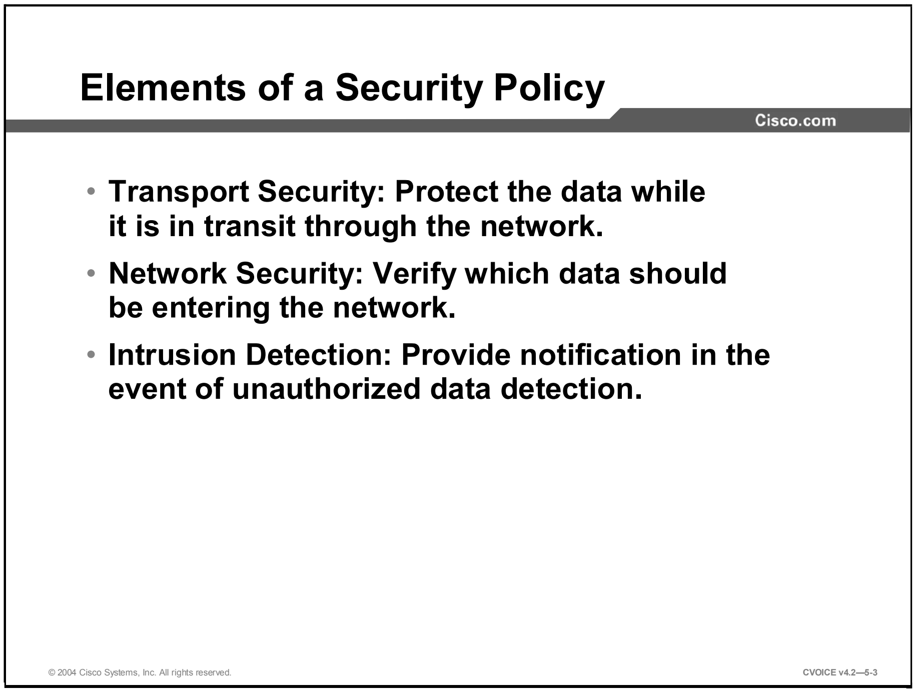
Outline

The outline lists the topics included in this lesson.



Security Policies for VoIP Networks

This topic describes the elements of a security policy for a VoIP network.



Numerous problems, from device failures to malicious attacks, affect the uptime of networks. With the reliance on the IP network for telephony, IP-based threats must be mitigated. Varying levels of security are available to suit individual corporate requirements. The requirements for secured IP telephony include the following:

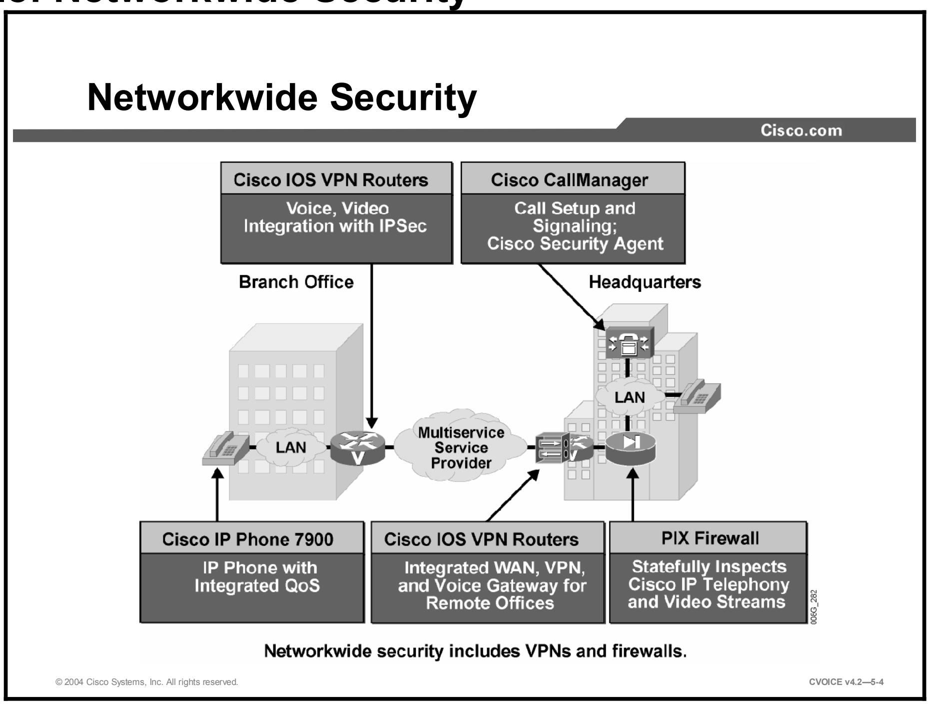
- It must provide ubiquitous IP telephony services to the locations and to the users that require them.
- It must maintain as many of the characteristics of traditional telephony as possible, while doing so in a secure manner.
- It must integrate with existing security architecture and not interfere with existing functions.

The starting point for any security implementation is the development of a security policy. In a converged network, the security policy must account for the impact of security measures on voice traffic. The security policy should address the following points that affect voice:

- **Transport security:** Traffic traversing public access and backbone networks must be properly secured. IPSec and VPNs provide transport security by ensuring data confidentiality using encryption, data integrity, and data authentication between participating peers. Encryption adds to the overhead and delays the voice packet. You must factor in encryption when testing the delay budget and bandwidth calculations.

- **Network security:** Cisco Systems firewalls provide stateful perimeter security that is critical to any public-facing network, such as a VPN. When deploying voice and video across VPNs, it is critical to statefully inspect all multiservice traffic traversing the firewall. Firewalls must be configured to allow known signal and payload ports to pass into the network. It is important to understand where the VPN terminates. If the VPN terminates inside the firewall, then the traffic passing through the firewall is encrypted and is subject to stateful inspection. If the VPN terminates outside the firewall, then the firewall has access to RTP/UDP/TCP/IP headers and is able to inspect the packet for call setup.
- **Intrusion detection:** The Cisco Security Agent provides threat protection for server and desktop computing systems, also known as endpoints. It identifies and prevents malicious behavior, thereby eliminating known and unknown security risks and helping to reduce operational costs. The Cisco Security Agent aggregates and extends multiple endpoint security functions by providing host intrusion prevention, distributed firewall capabilities, malicious mobile code protection, operating system integrity assurance, and audit log consolidation, all within a single product. In addition, because Cisco Security Agent analyzes behavior rather than relying on signature matching, it provides robust protection with reduced operational costs.

Example: Networkwide Security

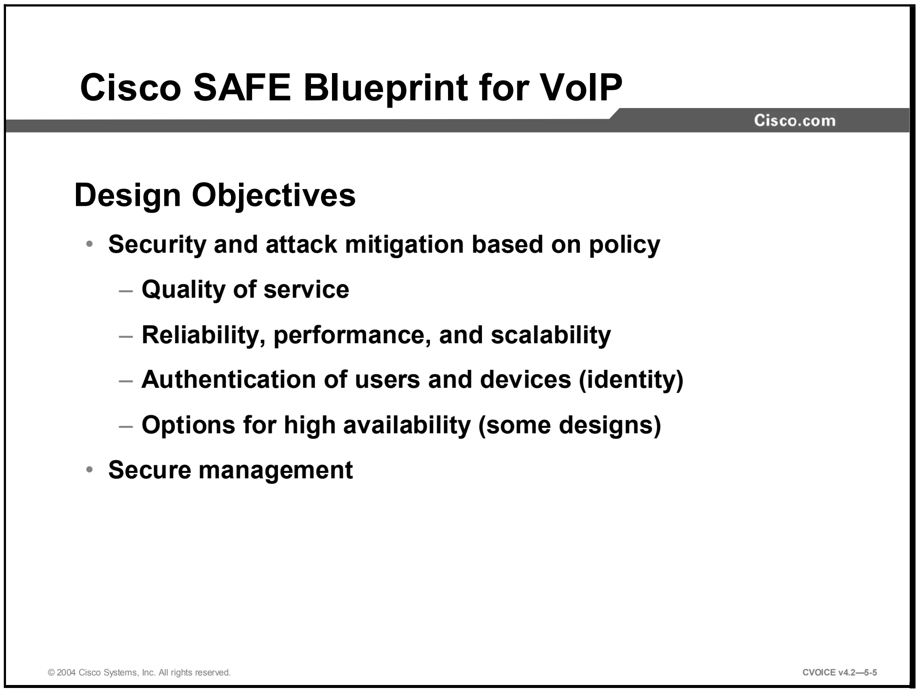


In the figure, the network between the branch office and the headquarters has a firewall. The firewall allows the users from the branch office only to access the headquarters network. A user without proper network identification will not be allowed to pass through the firewall. Network identification protects voice networks from hackers with ulterior motives.

The model chosen for IP voice networks today parallels that chosen for legacy voice systems, which are generally wide open and require little or no authentication to gain access.

Cisco SAFE Blueprint for VoIP

This topic describes the Cisco SAFE Blueprint for VoIP.



The Cisco SAFE Blueprint is a flexible, dynamic blueprint for security and VPN networks, based on the Cisco AVVID architecture, that enables businesses to securely and successfully take advantage of e-business economies and compete in the Internet economy.

Cisco has significantly enhanced the SAFE Blueprint, and extended network security and VPN options to small branch offices, teleworkers, and small-to-medium networks.

SAFE IP telephony emulates as closely as possible the functional requirements of modern networks. Implementation decisions vary, depending on the network functionality required. However, the following design objectives, listed in order of priority, guide the decision-making process.

- Security and attack mitigation based on policy
- Quality of service
- Reliability, performance, and scalability
- Authentication of users and devices (identity)
- Options for high availability (some designs)
- Secure management

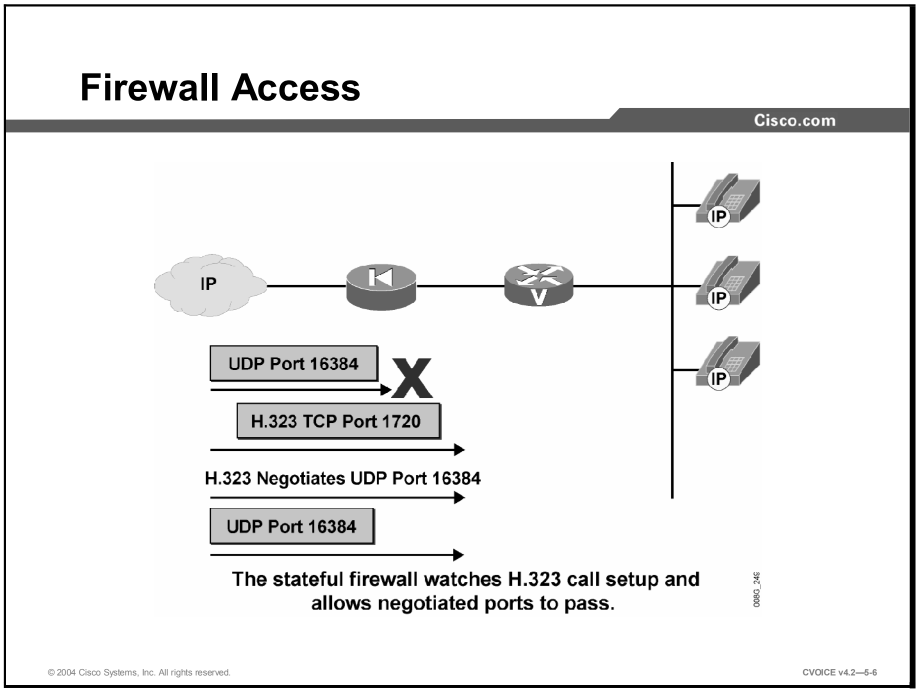
SAFE IP telephony must provide ubiquitous IP telephony services to the locations and users that require it. It must maintain as many of the characteristics of traditional telephony as possible while doing so in a secure manner. Finally, it must integrate with existing network designs based on the SAFE security architecture and not interfere with existing functions.

Example: SAFE Applied

At many points in the network design process, you need to choose between using integrated functionality in a network device or using a specialized functional appliance. The integrated functionality is often attractive because you can implement it on existing equipment, or because the features can interoperate with the rest of the device to provide a better functional solution. Appliances are often used when the depth of functionality required is very advanced or when performance needs require using specialized hardware. Make your decisions based on the capacity and functionality of the appliance versus the integration advantages of the device. For example, sometimes you can choose an integrated higher-capacity Cisco IOS router with IOS firewall software as opposed to a smaller IOS router with a separate firewall. Throughout this architecture, both types of systems can be used. When the design requirements do not dictate a specific choice, the design decision is to go with integrated functionality in order to reduce the overall cost of the solution.

Communicating Through a Firewall

This topic describes how voice is transmitted through a firewall.



Firewalls inspect packets and match them against configured rules. It is difficult to specify ahead of time which ports will be used in a voice call because they are dynamically negotiated during call setup.

H.323 is a complex, dynamic protocol that consists of several interrelated subprotocols. The ports and addresses used with H.323 require detailed inspection as call setup progresses. As the dynamic ports are negotiated, the firewall must maintain a table of current ports associated with the H.323 protocol. As calls are torn down, the firewall must remove those ports from the table. The process of removing ports from the table is called *stateful inspection of packets*. In addition to checking static ports and recognizing protocols that negotiate dynamic ports (for example, H.323), the firewall looks into the packets of that protocol to track the flows.

Example: Stateful Firewall

Any application may use a port in the range of 1024 to 65536. In the figure, the firewall initially blocks all packets destined for UDP port 16384. The firewall becomes H.323-aware when it is configured to look for TCP port 1720 for call setup and UDP port assignments.

The table illustrates the dynamic access control process used by firewalls.

Dynamic Access Control

Stage	What Happens
The firewall detects a new call setup destined for UDP port 16384.	The firewall places the port, the associated source, and the destination IP address into the table.
The firewall opens port 16384.	The firewall allows all packets with UDP port 16384 and the proper source/destination IP address through the firewall.
The firewall detects call teardown.	The firewall removes the ports from the list and the packets destined for the UDP port are blocked.

If the firewall does not support this dynamic access control based on the inspection, an H.323 proxy can be used. The H.323 proxy passes all H.323 flows to the firewall with the appearance of a single static source IP address plus TCP/UDP port number. The firewall can then be configured to allow that static address to pass through.

Firewalls can introduce variable delay into the path of the voice packet. It is extremely important that you ensure that the firewall has the proper resources to handle the load.

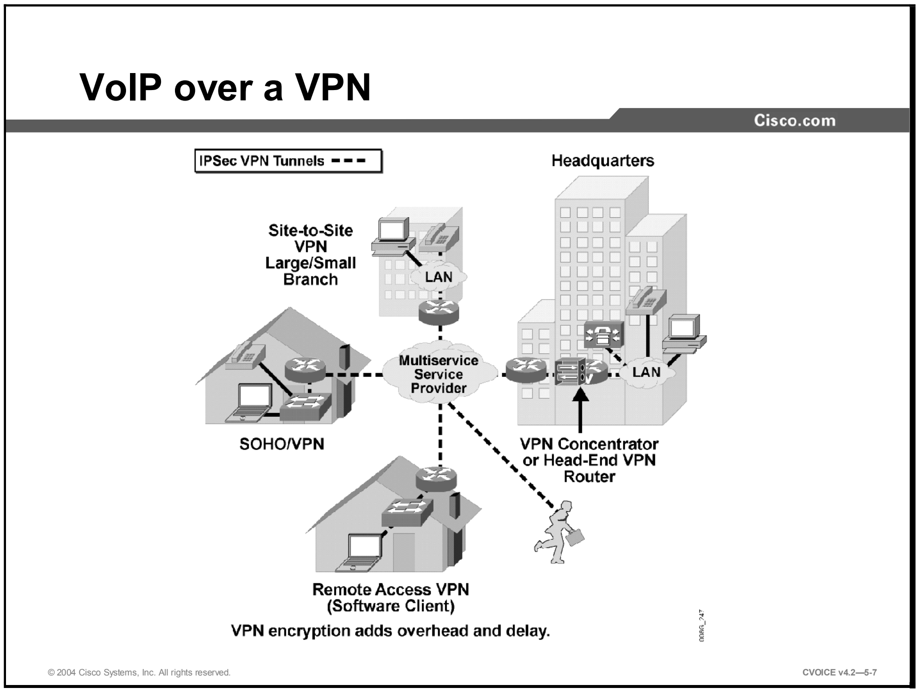
The table lists the ports used for various voice protocols.

Voice Protocol Ports

Protocol	Ports	TCP/UDP	Description
H.323	1718	UDP	Gatekeeper discovery
H.323	1719	UDP	Gatekeeper registration
H.323 (H.225)	1720	TCP	Call setup
MGCP	2427, 2727	UDP	MGCP gateway, MGCP call agent
SIP	5060	TCP or UDP	SIP call
Skinny	2000	TCP	Client
Skinny	2001	TCP	Digital gateway
Skinny	2002	TCP	Analog gateway
Skinny	2003	TCP	Conference bridge

Delivery of VoIP Through a VPN

This topic explains how to optimize the delivery of VoIP through a VPN.



VPNs are widely used to provide secure connections to the corporate network. The connections can originate from a branch office, a SOHO, a telecommuter, or a roaming user.

Frequently asked questions about voice over VPN generally deal with overhead and delay, which impact the QoS for the call.

Note	One important consideration to remember is the absence of QoS when deploying VPNs across the Internet or a public network. Where possible, QoS should be addressed with the provider through a service level agreement (SLA). An SLA is a document that details the expected QoS parameters for packets transiting the provider network.

Voice communications do not work with latency, not even a modest amount of it. Because secure VPNs encrypt data, they may create a throughput bottleneck when they process packets through their encryption algorithm. The problem usually gets worse as security increases. For example, Triple Data Encryption Standard (3DES) uses a long, 168-bit key. 3DES requires that each packet be encrypted three times, effectively tripling the encryption overhead.

The table describes how to reduce overhead and delay in VPNs.

Reducing Overhead and Delay

Step	Action
1.	Optimize the encryption algorithm and data path.
2.	Handle all processing in a dedicated encryption processor.
3.	Ensure that the device uses hardware encryption instead of software encryption.
4.	Use proper QoS techniques.
5.	Use proper VPN technologies.

VoIP can be secure and free of perceptible latency on a VPN. The solution is to optimize the encryption algorithm and the data path and to handle all processing in a dedicated encryption processor. You must ensure that the device is utilizing hardware encryption instead of software encryption. Software encryption relies on CPU resources and could severely impact voice quality.

Delay can be further minimized through use of proper QoS techniques. QoS and bandwidth management features allow a VPN to deliver high transmission quality for time-sensitive applications such as voice and video. Each packet is tagged to identify the priority and time sensitivity of its payload, and traffic is sorted and routed based on its delivery priority. Cisco VPN solutions support a wide range of QoS features.

It is important to understand that QoS cannot be completely controlled, independent of the underlying network. QoS is only as good as the network through which the voice travels. Users must confirm with a potential service provider that the network can support priority services over a VPN. SLAs with carriers can guarantee expectations of network stability and QoS.

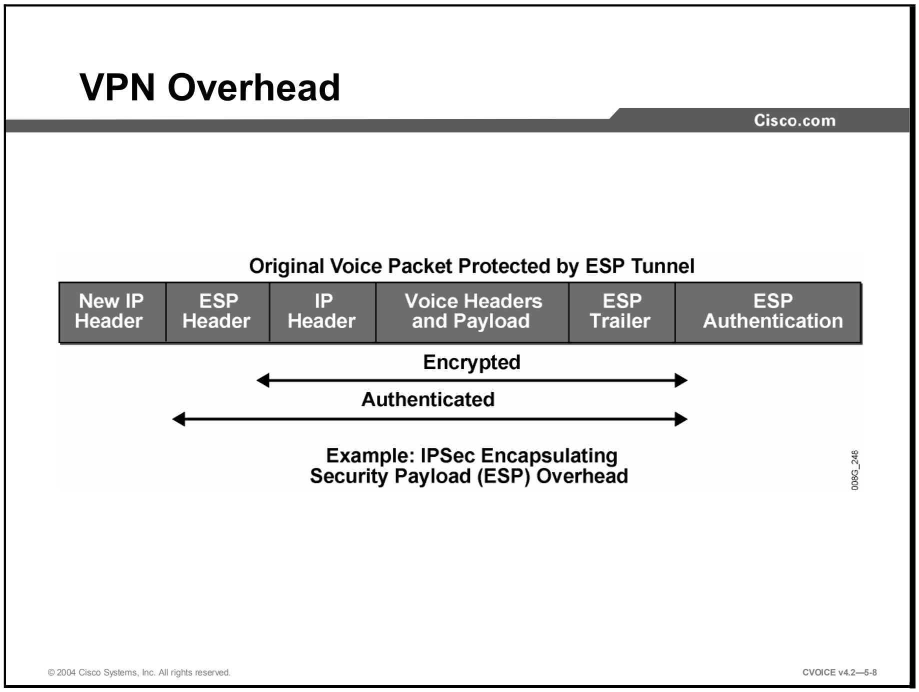
Overhead can be minimized if you understand the proper use of VPN technologies. VPNs can be implemented at Layer 2 through Point-to-Point Tunneling Protocol (PPTP) or Layer 2 Tunneling Protocol (L2TP), as well as at Layer 3 with IPSec. Often Layer 2 and Layer 3 technologies are combined to provide additional security. It is crucial that you understand the reasoning and requirements behind combining Layer 2 with Layer 3 security, because the combination adds overhead to the VoIP packet.

International Issues

VoIP and either Data Encryption Standard (DES) or 3DES encryptions are fully compatible with each other, assuming that the VPN delivers the necessary throughput. Internationally, however, corporations can run into other factors. The U.S. Department of Commerce places restrictions on the export of certain encryption technology. Usually, DES is exportable while 3DES is not, but that generality takes numerous forms, from total export exclusions applied to certain countries to allowing 3DES export to specific industries and users. Most corporations whose VPNs extend outside the United States should find out if their VPN provider has exportable products and how export regulations impact networks built with those products.

Bandwidth Overhead Associated with VPN

This topic discusses the bandwidth overhead associated with a VPN.



VPN implementations vary, and there are many options to explore.

IPSec is the predominant VPN in use today. Generally speaking, IPSec encrypts and/or authenticates the IP packet and adds additional headers to carry the VPN information. The VPN places the original IP packet into another IP packet so that the original information, including headers, is not easily seen or read.

To properly calculate bandwidth overhead, the user must have a thorough understanding of the VPN technology by asking the following questions:

- Should the VPN be a Layer 2 tunnel running PPTP or L2TP?
- Should the VPN be a Layer 3 tunnel running IPSec?
- If the VPN is IPSec, is it using Authentication Header (AH) or Encapsulating Security Payload (ESP)?
- If the VPN is running AH or ESP, is it in transport mode or tunnel mode?

Example: VPN Bandwidth

The VPN adds a new IP header that is 20 bytes, plus the VPN header, which can add as much as 20 to 60 bytes more, depending on which variation of VPN is installed. The table shows what a complete VPN packet can look like.

VPN Packet

Field	Subhead
Voice payload (G.729)	20 bytes
RTP header	12 bytes
UDP header	8 bytes
IP header	20 bytes
VPN header	20 to 60 bytes
New IP header	20 bytes

The total size of the packet will be 100 to 160 bytes.

To calculate the total bandwidth for a 160-byte G.729 packet, use the following calculations:

- Total bandwidth = 160 bytes * 8 = 1280 bits
- Total bandwidth = 1280 bits / 20 ms
- Total bandwidth = 64000 bps

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Since voice relies on the IP network, the network itself must be secured.
- Cisco SAFE Blueprint provides for security based on AVVID.
- Stateful firewalls inspect voice signaling packets to determine which UDP ports to allow through. Firewalls that are not capable of stateful inspection require the presence of an H.323 proxy server.
- VPN encryption headers introduce additional overhead that negatively impacts voice traffic.
- To calculate bandwidth overhead, you must understand the VPN technology and protocols.

CVOICE v4.2—5-9

References

For additional information, refer to these resources:

- *SAFE VPN: IPSec Virtual Private Networks in Depth*
  http://www.cisco.com/warp/public/cc/so/cuso/epso/sqfr/safev_wp.htm
- *RFC 2401 Security Architecture for the Internet Protocol*
  http://www.ietf.org/rfc/rfc2401.txt?number=2401
- *Cisco SAFE Blueprint*
  http://www.cisco.com/en/US/partner/netsol/ns340/ns394/ns171/ns128/networking_solutions_package.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which feature provides transport security for a VoIP network?

A) encryption and data authentication

B) inspection of traffic at the firewall

C) proactive notification of an attempted attack

D) intrusion detection

Q2) In which situation can the firewall inspect the packet for call setup?

A) the firewall is configured for call setup

B) the VPN terminates outside the firewall

C) the VPN terminates inside the firewall

D) the firewall does not recognize the signal or payload ports

Q3) SAFE IP telephony must provide ubiquitous IP telephony services to _____.

A) secured users, all of the time

B) executives only

C) telcos

D) locations and users that require it

Q4) The SAFE Blueprint is a flexible, dynamic blueprint for security and _____.

A) VPN networks

B) AVVID

C) MGCP

D) firewalls

Q5) Which ports does the firewall include in the list of ports that are allowed access?

A) all TCP and UDP ports

B) ports that are configured on the firewall

C) ports that were used during past call setups

D) ports that are currently being used for call setup

Q6) What can you do if the firewall does NOT support dynamic access control?

A) use a different protocol

B) use an H.323 proxy

C) configure the firewall for all possible ports

D) use static port numbers for call setup and teardown

Q7) When compared to software encryption, how does hardware encryption reduce overhead and delay?

A) It optimizes the encryption algorithm and data path.

B) It handles all processing in a dedicated compression engine.

C) It does not rely on CPU resources.

D) It uses proper QoS technologies.

Q8) Which three of the following encryption technologies can be used on international networks? (Choose three.)

A) DES

B) 3DES

C) ESP

D) IPSec

E) IDEA

Q9) How many bytes of overhead does VPN add to the voice packet?

A) 20 bytes

B) 40 bytes

C) 20 to 60 bytes

D) 40 to 80 bytes

Q10) Which type of VPN is most commonly used today?

A) IPSec

B) ESP

C) PPTP

D) L2TP

Q11) A converged network security policy should address which three of the following points? (Choose three.)

A) intrusion detection

B) application policy

C) transport security

D) network security

E) WAN access security

Q12) Which of the following design objectives has the highest priority in a SAFE IP implementation?

A) secure management

B) security and attack mitigation based on policy

C) quality of service

D) reliability, performance, and scalability

Quiz Answer Key

Q1) A

Relates to: Security Policies for VoIP Networks

Q2) B

Relates to: Security Policies for VoIP Networks

Q3) D

Relates to: Cisco SAFE Blueprint for VoIP

Q4) A

Relates to: Cisco SAFE Blueprint for VoIP

Q5) D

Relates to: Communicating Through a Firewall

Q6) B

Relates to: Communicating Through a Firewall

Q7) C

Relates to: Delivery of VoIP Through a VPN

Q8) A, C, D

Relates to: Delivery of VoIP Through a VPN

Q9) C

Relates to: Bandwidth Overhead Associated with VPN

Q10) A

Relates to: Bandwidth Overhead Associated with VPN

Q11) A, C, D

Relates to: Security Policies for VoIP Networks

Q12) B

Relates to: Cisco SAFE Blueprint for VoIP

Module 6

VoIP Signaling and Call Control

Overview

To provide voice communication over an IP network, Real-Time Transport Protocol (RTP) sessions are created. These sessions are dynamically created and facilitated by one of several call control procedures. Typically, these procedures also embody mechanisms for signaling events during voice calls and for managing and collecting statistics about the voice calls. This module focuses on three protocols that offer call control support for Voice over IP (VoIP): H.323, the session initiation protocol (SIP), and the Media Gateway Control Protocol (MGCP).

Module Objectives

Upon completing this module, you will be able to compare centralized and decentralized call control and signaling protocols.

Module Objectives

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- Identify the appropriate call control model for your network
- Configure, monitor, and troubleshoot H.323 gateways and gatekeepers
- Configure, monitor, and troubleshoot SIP on a Cisco router
- Configure, monitor, and troubleshoot MGCP on a Cisco router
- Determine the best call control model for your network

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Module Outline

The outline lists the components of this module.

Module Outline

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- Understanding the Need for Signaling and Call Control
- Configuring H.323
- Configuring SIP
- Configuring MGCP
- Comparing Call Control Models

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Understanding the Need for Signaling and Call Control

Overview

Signaling and call control are fundamental to the call establishment, management, and administration of voice communication in an IP network. This lesson discusses the benefits of using signaling and call control and offers an overview of what signaling and call control services provide.

Relevance

Because signaling and call control are fundamental to VoIP, you must understand the roles that signaling and call control play in establishing, managing, and administering connections.

Objectives

Upon completing this lesson, you will be able to identify the appropriate call control model for your network. This includes being able to meet these objectives:

- Describe the endpoints and the common control components that are used in VoIP signaling
- List five protocols used for call control in VoIP
- Explain why the call control gateway must translate signals from different call control models to support end-to-end calls
- List five call parameters that must be negotiated before placing a call
- Name the functions that are performed by call accounting and administration as common call control components
- Name the functions that are performed by call status and call detail records as common call control components
- Describe the address registration and address resolution functions that are performed by the address management common control component
- Explain how admission control can protect network resources when it is used as a common control component

- Illustrate the setup of a centralized call control model
- Illustrate the setup of a distributed call control model
- Identify the advantages and disadvantages of centralized and distributed call control

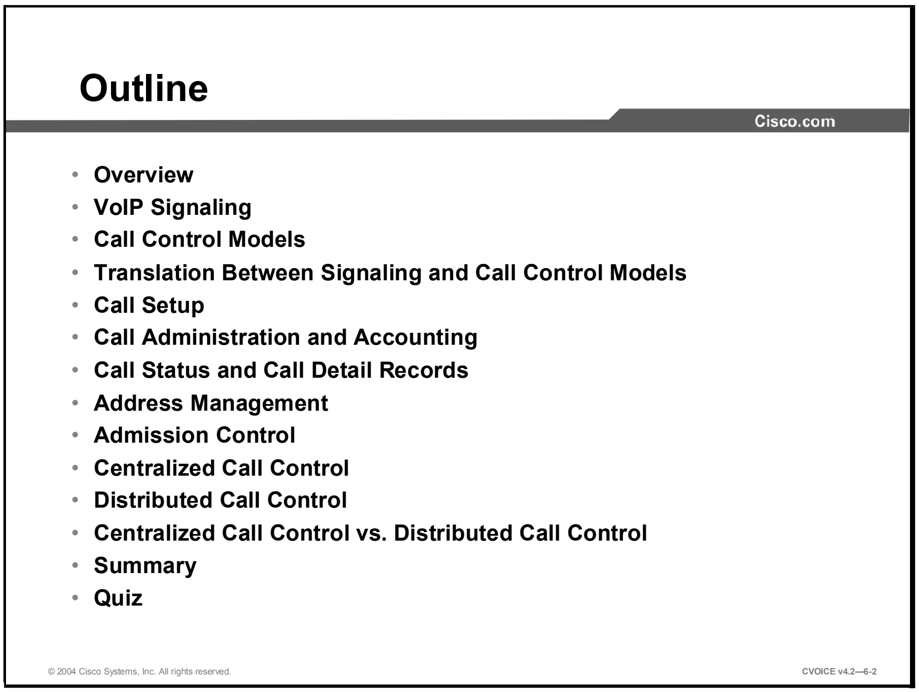
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- An understanding of the protocol environment in which VoIP operates

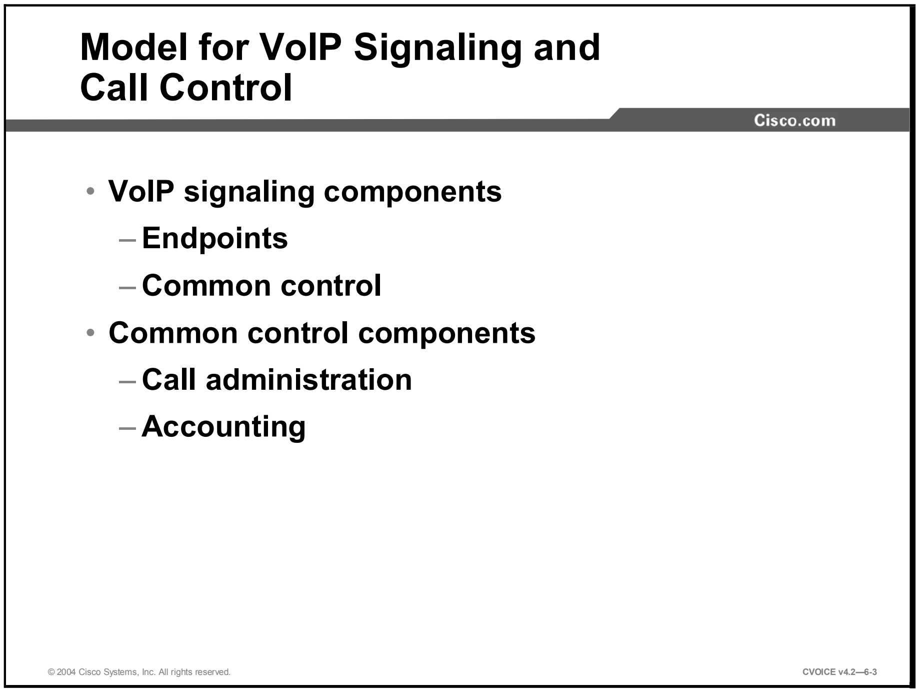
Outline

The outline lists the topics included in this lesson.



VoIP Signaling

In a traditional voice network, call establishment, progress, and termination are managed by interpreting and propagating signals. Transporting voice over an IP internetwork creates the need for mechanisms to support signaling over the IP component of the end-to-end voice path. This topic introduces the components and services provided by VoIP signaling.



In the traditional telephone network, a voice call consists of two paths: an audio path carrying the voice and a signaling path carrying administrative information such as call setup, teardown messages, call status, and call-progress signals. ISDN D channel signaling and Common Channel Signaling System 7 (CCSS7) or Signaling System 7 (SS7) are two examples of signaling systems that are used in traditional telephony.

By introducing VoIP into the call path, the end-to-end path involves at least one call leg that uses an IP internetwork. As in a traditional voice call, support for this VoIP call leg requires two paths: a protocol stack that includes RTP, which provides the audio call leg, and one or more call control models that provide the signaling path.

A VoIP signaling and call control environment model includes endpoints and optional common control components, as follows:

- **Endpoints:** Endpoints are typically simple, single-user devices, such as terminals, that support either a voice process (for example, the Cisco IP Phone application) or a gateway. In either case, the endpoint must be able to participate in signaling with other VoIP endpoints—directly or indirectly—through common control components. The endpoints must also be able to manipulate the audio that is in the audio path. This may involve performing analog-to-digital conversion or converting the format to digital voice so that it takes advantage of compression technology.

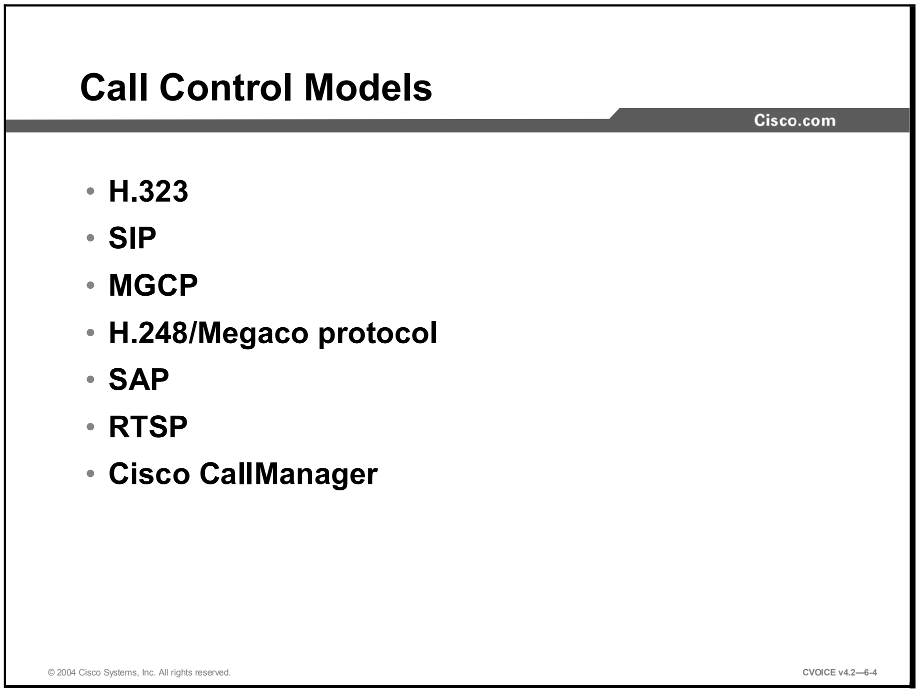
  Gateways provide physical or logical interfaces to the traditional telephone network. A gateway that is connected digitally to a service provider central office (CO) switch is an example of a gateway providing a physical interface. A gateway that provides access to an interactive response dialog application is an example of a gateway providing a logical interface.

- **Common control:** In some call control models, the common control component is not defined; in others, it is employed optionally. Common control components provide call administration and accounting. These components provide a variety of services to support call establishment, including the following:
  - Call status
  - Address registration and resolution
  - Admission control

Typically, the services of the common control components are implemented as applications. These services are colocated in a single physical device, or distributed over several physical devices with standalone endpoints and gateways.

Call Control Models

This topic describes several call control models and their corresponding protocols.

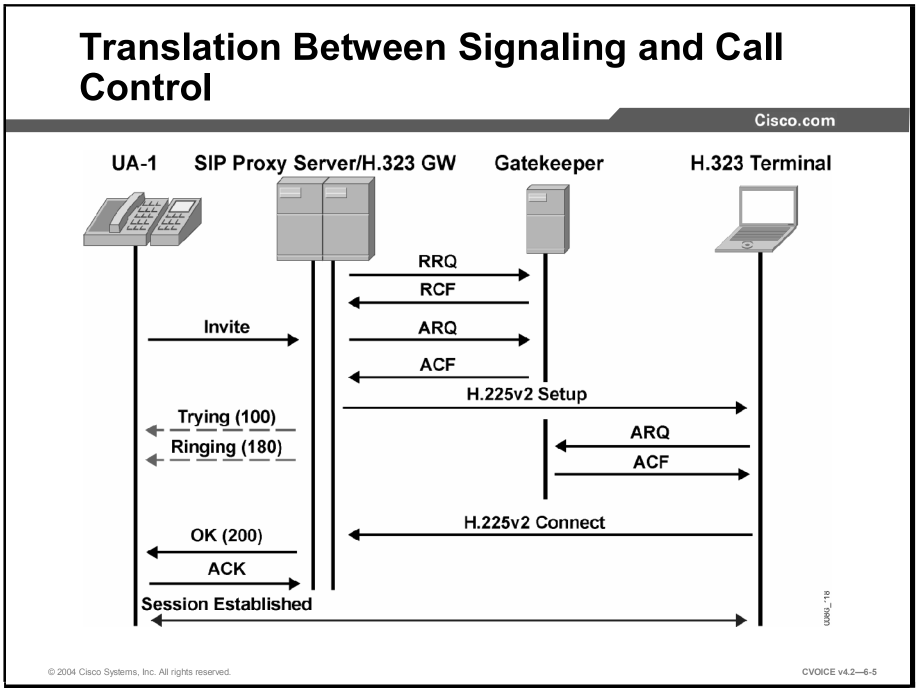


The following call control models and their corresponding protocols exist or are in development:

- **H.323:** International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation H.323 describes the architecture to support multimedia communications over networks without quality of service (QoS) guarantees. Originally intended for LANs, H.323 has been adapted for IP.
- **SIP:** SIP is an Internet Engineering Task Force (IETF) RFC 3261 call control model for creating, modifying, and terminating multimedia sessions or calls.
- **MGCP:** MGCP (IETF RFC 2705) defines a call control model that controls VoIP gateways from an external call control element or call agent.
- **H.248/Megaco protocol:** The Megaco protocol is used in environments in which a media gateway consists of distributed subcomponents, and communication is required between the gateway subcomponents. The Megaco protocol is a joint effort of IETF (RFC 3015) and ITU-T (Recommendation H.248).
- **Session Announcement Protocol (SAP):** SAP (IETF RFC 2974) describes a multicast mechanism for advertising the session characteristics of a multimedia session, including audio and video.
- **Real Time Streaming Protocol (RTSP):** RTSP (IETF RFC 2326) describes a model for controlled, on-demand delivery of real-time audio and video.
- **Cisco CallManager (“Skinny”):** Cisco CallManager is a proprietary Cisco Systems implementation of a call control environment that provides basic call processing, signaling, and connection services to configured devices, such as IP telephones, VoIP gateways, and software applications.

Translation Between Signaling and Call Control Models

When VoIP endpoints support different call control procedures, the calls between the endpoints require cooperation between the originating and terminating procedures. This topic identifies the need for interworking or translation between call control models.



In the traditional telephone network, the individual call legs contributing to an end-to-end call often involve different signaling systems and procedures. In the graphic, an IP Phone is communicating with its SIP proxy server using the SIP protocol. However, it is also attempting to reach an H.323 endpoint. Because the two VoIP protocols are different, a translation is necessary at the SIP proxy server (namely, an H.323 gateway) to allow the two telephony endpoints to establish a connection.

Example: Call Control Translation

A call between a residential user and an office worker likely involves a signaling system that is unique to the various call legs that exist between the originator and the destination. In this scenario, the sequence of signaling systems includes the following:

- Analog signaling (Foreign Exchange Station [FXS] or Foreign Exchange Office [FXO] loop start) to the CO
- CCSS7 between the COs
- ISDN PRI signaling to the PBX
- Proprietary signaling to the desktop telephone

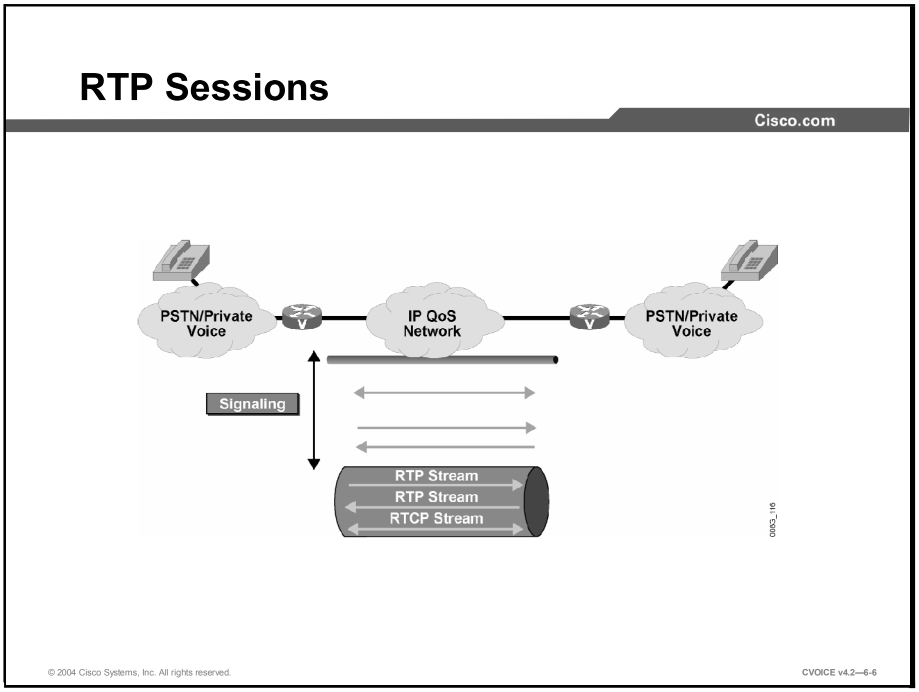
When part of the path is replaced with an IP internetwork, the audio path between the IP endpoints is provided by RTP, and the call control mechanism is based on a call control protocol, such as SIP, H.323, or MGCP.

But what if different call control models represent the endpoints? What if, for example, the originating endpoint uses H.323 and the destination is managed as an SIP endpoint?

To complete calls across the IP internetwork, a call control gateway that recognizes the procedures of *both* call control models is required. In particular, the translating gateway interprets the call setup procedure on the originating side and translates the request to the setup procedure on the destination side. Ideally, this translation is transparent to the endpoints that are involved and results in a single endpoint-to-endpoint audio relationship.

Call Setup

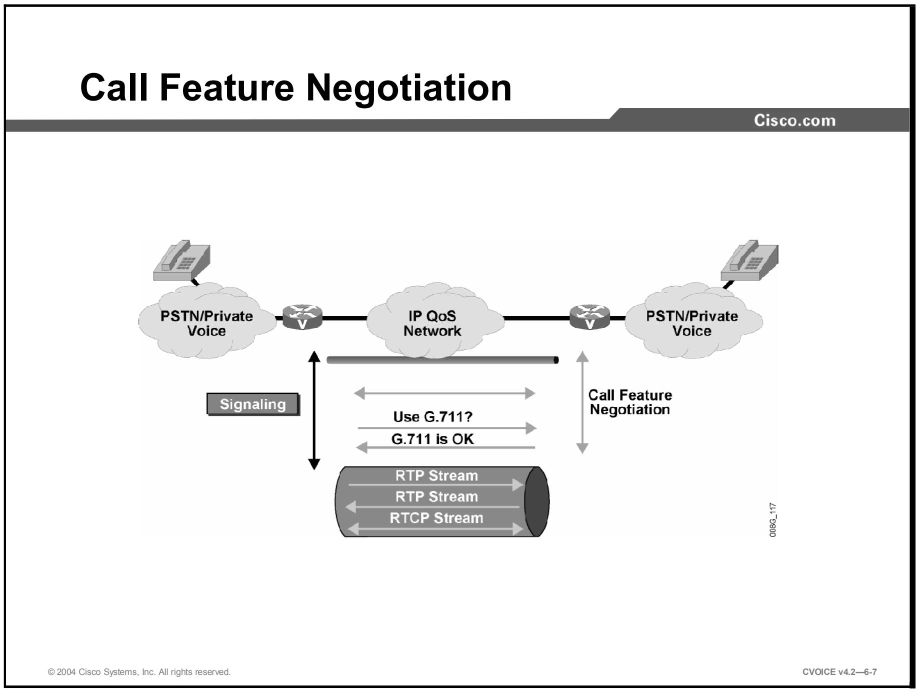
A fundamental objective of VoIP call control is to initiate communication between VoIP endpoints. This topic discusses the role of call control in establishing RTP sessions and negotiating features during the call setup procedure.



An audio path of a VoIP call leg is dependent on the creation of RTP sessions. These RTP sessions transport voice unidirectionally, so that bidirectional voice uses two RTP sessions. (In principle, if voice is needed in one direction only, as in the case of a recorded announcement or voice mail, only one RTP session is required.) The figure shows RTP sessions being created during call setup.

To create RTP sessions, each endpoint must recognize the IP address and User Datagram Protocol (UDP) port number of its peer. In a limited implementation of VoIP, these values are preprogrammed. However, to be truly scalable, the addresses and port numbers must be recognized dynamically and on demand.

During call setup, call control procedures exchange the IP address and UDP port numbers for the RTP sessions.



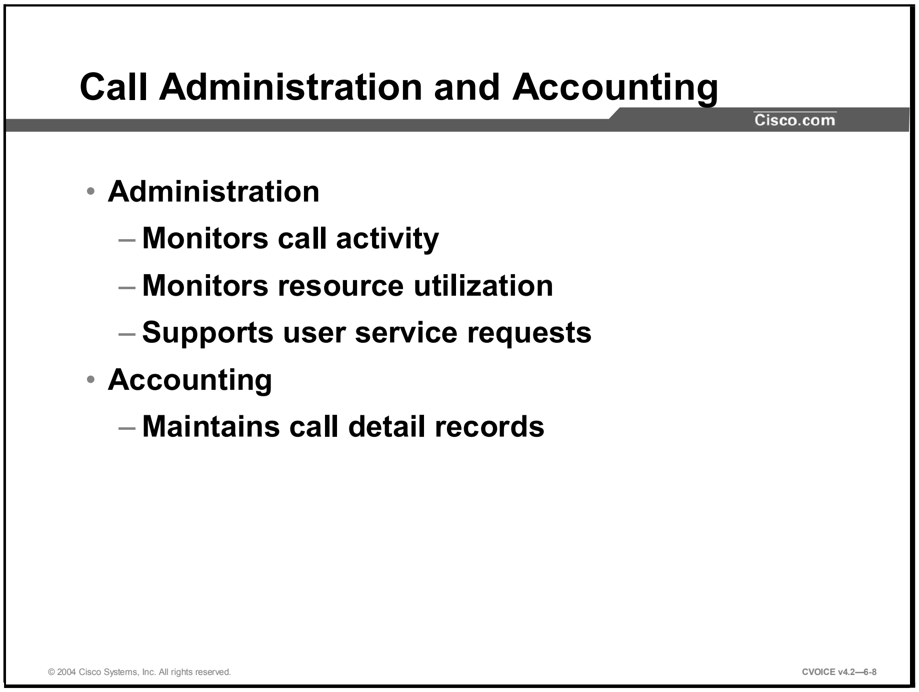
Creating the RTP sessions is not the only task of call control during call setup. The endpoints need to establish a bilateral agreement in which the communicating parties discover acceptable call parameters and then agree on the operating parameters of the call. When agreement is not possible, the call is not completed and is dropped.

Following are some examples of call parameters:

- **Coder-decoder (codec):** Each endpoint must share a common format for the voice, or at least must recognize the opposite endpoint choice for voice encoding. This is an example of a mandatory agreement. Not finding a common format is analogous to calling a foreign land and discovering that you are unable to carry on a conversation because the other party speaks a different language.
- **Receive/transmit:** Based on the application, the voice is one-way or two-way. Some endpoints do not meet the requirement for the session because they are designed to handle receive-only or transmit-only traffic when the call requests two-way communication.
- **Multipoint conferences:** The types of conferences and parameters to join.
- **Media type:** Audio, video, or data.
- **Bit rate:** Throughput requirements.

Call Administration and Accounting

Call control procedures typically provide support for call administration and accounting. This topic discusses administration and accounting capabilities of call control.



Call administration and accounting functions provide optional services for the improved operation, administration, and maintenance of a VoIP environment.

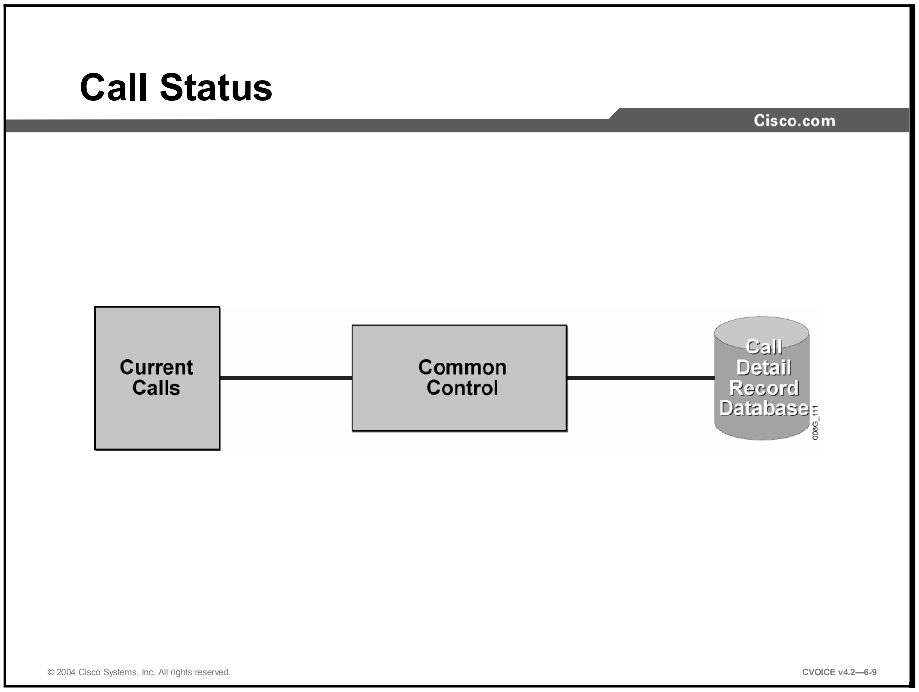
Accounting makes use of the historical information that is usually formatted as call detail records. Call detail records are useful for cost allocation, and for determining call distribution and service grade for capacity-planning purposes.

Call administration includes the following capabilities:

- **Call status:** Monitoring calls in real time
- **Address management:** Supporting users with services such as address resolution
- **Admission control:** Ensuring that resources are being used effectively

Call Status and Call Detail Records

Several call control protocols offer dynamic access to the status of calls within the VoIP network. This topic discusses the benefits of maintaining call status information and describes where and how call status is used.



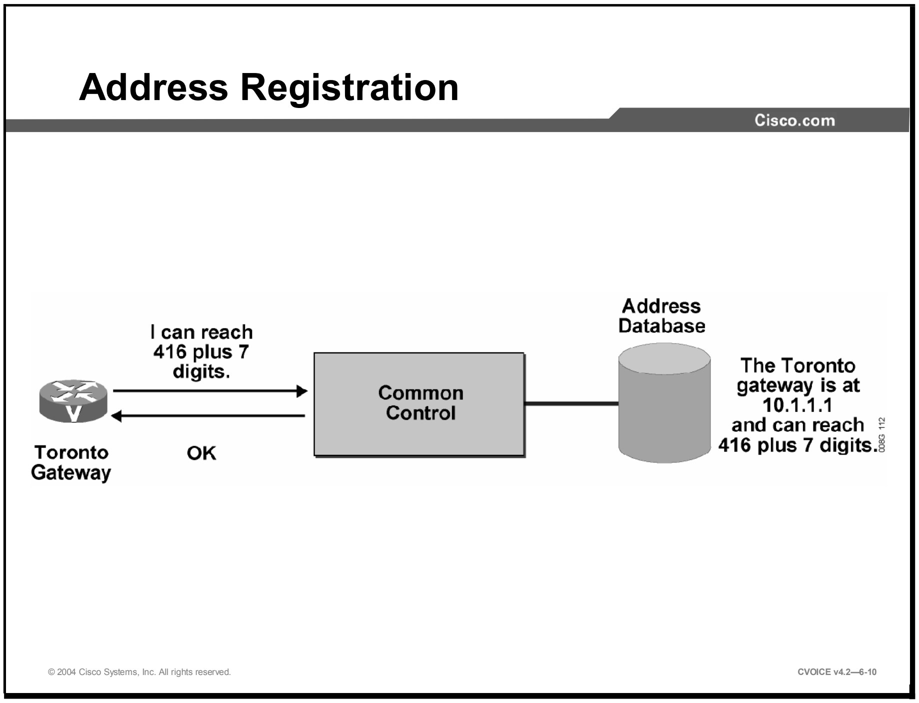
Several of the responsibilities that are assigned to call administration and accounting are dependent on access to current call status information or records of changes in the call status. Call status has both historical and instantaneous (real-time) benefits. Call detail records have consequential benefits in terms of distributing costs and planning capacity.

Call status provides an instantaneous view of the calls that are in progress. This view assists other processes (for example, bandwidth management) or assists an administrator with troubleshooting or user support.

Call detail records include information about a call start time, duration, origin, destination, and other statistics that may be useful for a variety of purposes. This data is collected as a function of call status.

Address Management

As an aspect of call administration, call control maintains a database of endpoints and their identifiers. This topic discusses how endpoints register their addresses and how these addresses are resolved to IP addresses.

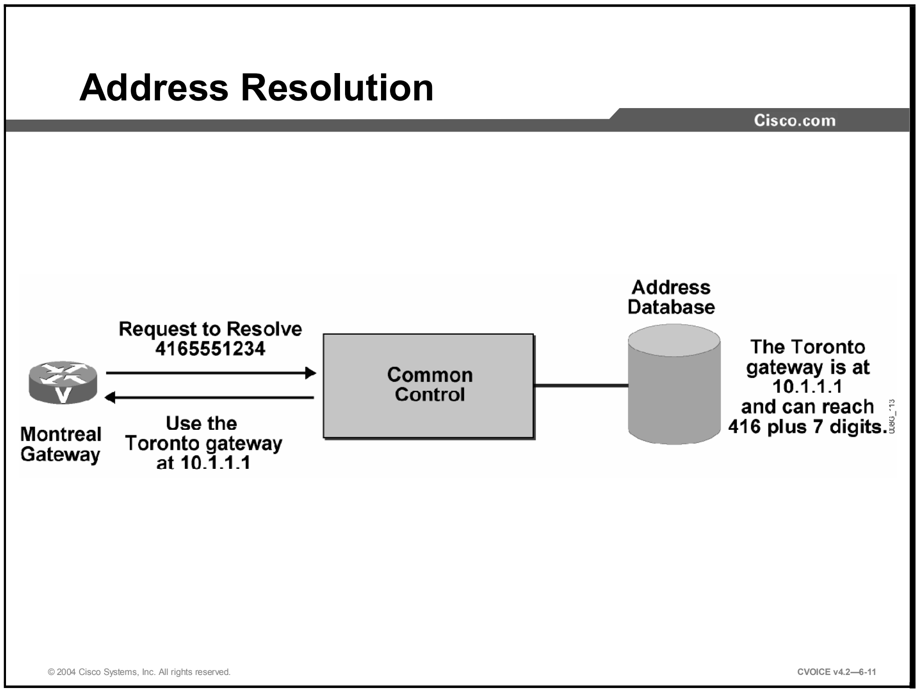


When an endpoint registers with a call control component, it supplies its telephone address or addresses and, if it is a gateway, the addresses of the destinations it can reach. The endpoint provides other information relating to its capabilities. Multiple destinations that are reachable through a gateway are usually represented by a prefix. The use of a prefix allows call control to create a database that associates a telephony-type address, for example, with its corresponding IP address.

In the traditional telephone network, the address of a station is limited to the keys available on a dual-tone multifrequency (DTMF) keypad. In VoIP, an address takes on one of several other formats as well; for example, the address can be a host name or a URL.

Example: Address Registration

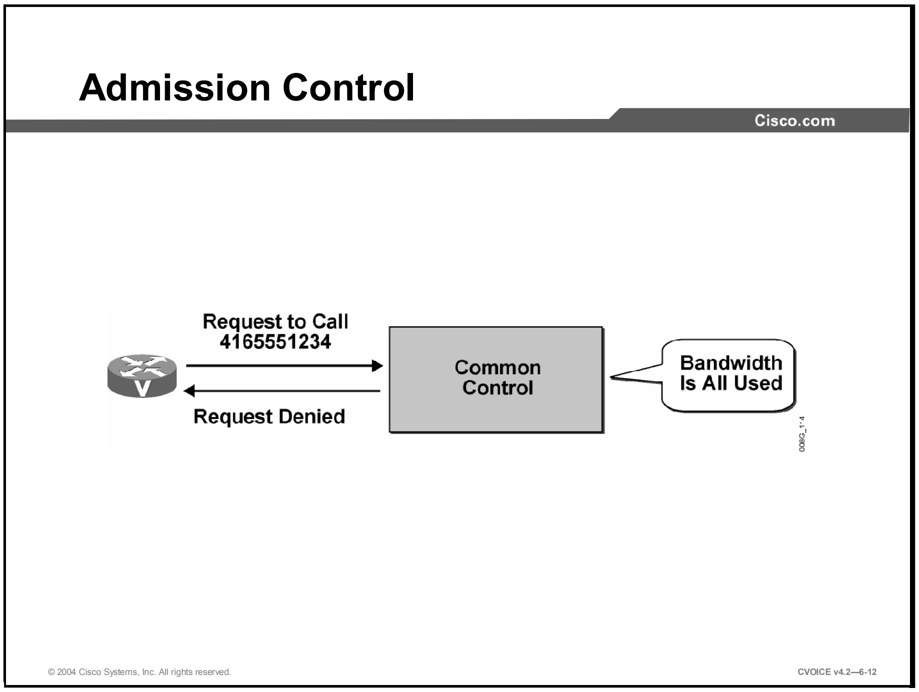
The figure illustrates the Toronto gateway registering its accessibility information. In this example, the gateway informs the common control component that it can reach all telephone numbers in the 416 area. This information is deposited into a database for future reference.



After an address is registered, it can be discovered through address resolution. Address resolution translates a multimedia user address to an IP address so that the endpoints can communicate with each other to establish a control relationship and create an audio path.

Admission Control

This topic discusses how common control and bandwidth management restrict access to the network.



Admission control has at least two aspects: authorization and bandwidth management.

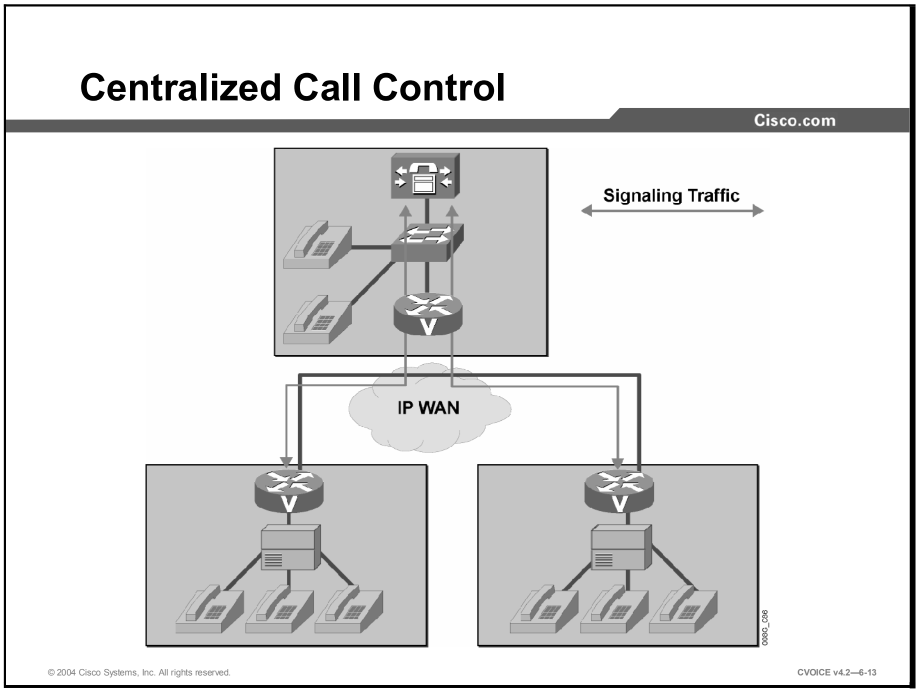
Access to a network should not imply permission to use the resources of the network. Common control limits access to the resources by checking the intentions and credentials of users before authorizing them to proceed.

Bandwidth is a finite resource. Appropriate bandwidth management is essential to maintaining voice quality. Allowing too many voice calls over an IP internetwork results in loss of quality for both new and existing voice calls.

To avoid degrading voice quality, a call control model establishes a bandwidth budget. By using data available from call status, the bandwidth management and call admission control functions monitor current bandwidth consumption. Calls may proceed up to the budgeted level, but are refused when the budget has reached its limit. This process is illustrated in the figure.

Centralized Call Control

This topic describes the function of endpoints in a centralized call control model.



The level of "intelligence" associated with an endpoint classifies call control models. In centralized call control, the intelligence in the network is associated with one or several controllers. Endpoints provide physical interconnection to the telephone network. However, endpoints still require the central controller to dictate when and how to use their interconnect capability. For example, the central controller may require the endpoint to provide dial tone to a telephone and alert the controller when a telephone goes off hook. The endpoint does not make decisions autonomously.

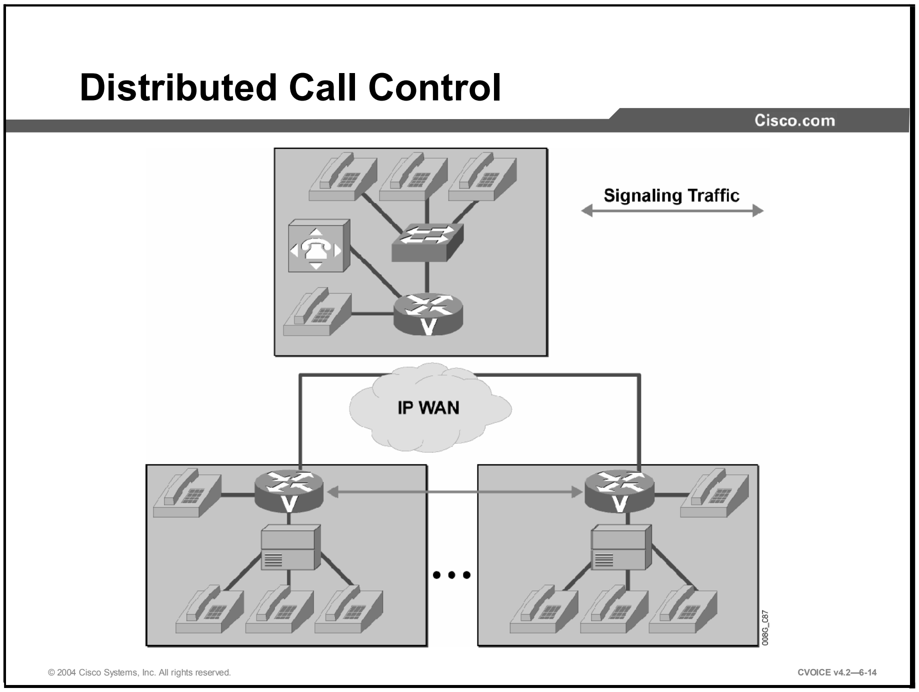
Example: Centralized Call Control

The figure illustrates two gateways interacting with a centralized call control component for call setup and management. Notice that the media path is between the two gateways.

A centralized call control model is similar to the mainframe model of computing and uses centralized common control concepts that are found in circuit-switching solutions.

Distributed Call Control

This topic describes the functions of the endpoints in a distributed call control model.



Endpoints in a distributed call control model have sufficient intelligence to autonomously manage the telephone interface and initiate VoIP call setup. Although the endpoint in a distributed model often depends on shared common control components for scalability, it possesses enough intelligence to operate independently from the common control components. If a common control component fails, an endpoint can use its own capabilities and resources to make routing or rerouting decisions.

Example: Distributed Call Control

In the figure, the two gateways have established a call control path and a media path between themselves, without the assistance of a centralized call control component.

A distributed call model is analogous to a desktop computing model.

Centralized Call Control vs. Distributed Call Control

This topic compares the advantages and disadvantages of the centralized and distributed call control models.

Centralized Call Control vs. Distributed Call Control

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Centralized Call Control	Distributed Call Control
Centralized administration	Distributed administration
Ease of dial plan consistency and updating	Dial plan consistency and updating is more difficult
Supplementary services (PBX features)	Supplementary services harder to implement
Difficult to scale; all new features and applications must be implemented on the central controller, central breakpoint, or bottleneck	Scalable: need more applications functions or performance. Add more servers and they can be located anywhere
Difficult to provide resiliency over network failures	Resilient over network failures
Difficult to add new endpoints and applications; elements are tightly associated	Conceptually easy to add new endpoints and applications
WAN inefficient	WAN efficient
Dial delay bears no relation to distance between endpoints	Dial delay roughly proportional to the distance between the endpoints, as in the public switched telephone network (PSTN)
Static endpoint capabilities	Flexible: negotiation of endpoint capabilities per session

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The figure shows a comparison of the centralized and distributed call control models. Both models have advantages and disadvantages. Features that are considered advantages of one type of call control model are disadvantages of the other type. The main differences between the two models are in the following areas:

- **Configuration:** The centralized call control model provides superior control of the configuration and maintenance of the dial plan and endpoint database. It simplifies the introduction of new features and supplementary services and provides a convenient location for the collection and dissemination of call detail records. The distributed model requires distributed administration of the configuration and management of endpoints, thus complicating the administration of a dial plan. Although distributed call control simplifies the deployment of additional endpoints, new features and supplementary services are difficult to implement.
- **Security:** Centralized call control requires that endpoints be known to a central authority, which avoids (or at least reduces) security concerns. The autonomy of endpoints in the distributed model elevates security concerns.
- **Reliability:** The centralized model is vulnerable because of its single point-of-failure and contention. It places high demands on the availability of the underlying data network, necessitating a fault-tolerant WAN design. The distributed call control model minimizes the dependence on shared common control components and network resources, thus reducing vulnerability.

- **Efficiency:** Centralized call control fails to take full advantage of computer-based technology that resides in the endpoints. It also consumes bandwidth through the interaction of the call agent and its endpoints. Distributed call control takes advantage of the inherent computer-based technology in endpoints.

Example: Comparing Centralized and Distributed

Because the centralized model increases vulnerability, it mandates the implementation of survivability and load management strategies that involve the replication of the central components. Few implementations of call control are totally distributed. For example, although H.323 and SIP operate in a purely distributed mode, for scalability reasons, both are most often deployed with common control components that give endpoints many of the advantages of a centralized call control environment. Unfortunately, these implementations also inherit many of the disadvantages of centralized call control.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- VoIP signaling and call control require endpoints and common control components, such as call administration and accounting.
- Protocols used for signaling and call control in VoIP include H.323, SIP, MGCP, Megaco, SAP, RTSP, and Skinny.
- Translation between signaling and call control can be achieved at a gateway that implements two or more of the call control capabilities. Translation does not require manipulation of the audio path, only the control path.
- One of the main objectives of signaling and call control is to exchange parameters for RTP session establishment and to allow the negotiation of special call features.

 CVOICE v4.2—6-16

Summary (Cont.)

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- Call administration includes call status, address management, and admission control services. Accounting provides call detail records.
- Call status provides access to information about calls in progress and facilitates the creation of historical records for cost distribution and network planning.
- Address management facilitates registering and locating endpoints.
- Access control limits unauthorized access and oversubscription to network resources.
- The centralized call control model implements switching intelligence in a central controller. The endpoints do not possess the intelligence to operate independently.

 CVOICE v4.2—6-17

Summary (Cont.)

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- The decentralized call control model distributes the intelligence for initiating calls to the endpoints. Common control equipment offers support to the endpoints.
- Centralized call control improves operations and maintenance but places higher demands on the common control equipment and elevates the risk of outages through higher exposure to the WAN.
- Decentralized call control offloads risk by distributing responsibility but increases the role of operations and maintenance.

CVOICE v4.2—6-18

References

For additional information, refer to these resources:

- ITU-T Recommendation H.323
- IETF RFC 3261, *SIP: Session Initiation Protocol*
- IETF RFC 2705: *Media Gateway Control Protocol (MGCP), version 1.0*

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which two are examples of signaling systems used in traditional telephony? (Choose two.)

A) cross-channel signaling

B) YBTM signaling

C) CCSS7

D) interswitch signaling

E) ISDN D channel

Q2) Why do endpoints in a VoIP signaling and call control model convert the packet format to digital?

A) to allow compression technology

B) to allow the use of DSPs

C) to allow faster transmission

D) to reduce errors in transmission

Q3) Which three are examples of VoIP call control protocols? (Choose three.)

A) SIP

B) RSVP

C) Megaco

D) SAP

E) SGBP

Q4) Match the call control model with its description.

A) H.323

B) MGCP

C) H.248

D) RTSP

E) Skinny

_____ 1. a model for controlled, on-demand delivery of real-time audio and video

_____ 2. allows control of VoIP gateways from a call agent

_____ 3. controls communication between the gateway subcomponents

_____ 4. provides basic call processing, signaling, and connection services to configured devices

_____ 5. describes the architecture to support multimedia communications over networks without QoS guarantees

Q5) On a VoIP network, which protocol carries the audio path?

A) RTP

B) SIP

C) H.323

D) MGCP

Q6) When two endpoints using different call control models communicate, which call control function is required at the gateway?

A) authentication

B) compression

C) registration

D) translation

Q7) Which two of the following call parameters may be negotiated during call setup? (Choose two.)

A) codec

B) port number

C) media type

D) E&M requirements

E) IP address

Q8) Which information does each endpoint need to recognize to create RTP sessions? (Choose two.)

A) call agent address

B) IP address of its peer

C) SMTP protocol used by its peer

D) type of firewall traffic being transmitted by its peer

E) UDP port number of its peer

Q9) Which capabilities are NOT included in call administration?

A) call status

B) call detail records

C) address management

D) admission control

Q10) Match the call administration and accounting service with its description.

A) call status

B) address management

C) admission control

D) call detail records

_____ 1. ensures resources are being used effectively

_____ 2. allows determination of call distribution and grade of service

_____ 3. supports users with services such as address resolution

_____ 4. monitors call activity in real time

Q11) Which two benefits are associated with call detail records? (Choose two.)

A) bandwidth management

B) troubleshooting

C) cost distribution

D) user support

E) capacity planning

Q12) What is the function of call status?

A) estimate the current bandwidth

B) provide call detail records

C) provide authentication information

D) report registered addresses

Q13) What is the purpose of address resolution?

- A) obtain the capabilities of an endpoint
- B) translate a multimedia user address to an IP address
- C) discover routing table information
- D) register the telephone numbers that an endpoint can reach

Q14) How does an endpoint provide information about all the destinations it can reach?

- A) with a list of all the telephone numbers it can reach
- B) with a list of all the IP addresses it can reach
- C) with a routing table
- D) with a prefix

Q15) What are two aspects of admission control? (Choose two.)

- A) authentication
- B) authorization
- C) call detail record creation
- D) bandwidth management
- E) address resolution

Q16) What information do the bandwidth management and call admission control functions use to monitor bandwidth consumption?

- A) router configuration
- B) network speed
- C) call detail records
- D) call status data

Q17) Which device has the most intelligence in a centralized call control model?

- A) originating endpoint
- B) central controller
- C) gateway
- D) gatekeeper
- E) intermediate router

Q18) Which statement is true regarding endpoints in a centralized call control model?

- A) Endpoints automatically provide dial tone when a telephone goes off hook.
- B) Endpoints automatically start call setup when a telephone goes off hook.
- C) Endpoints manage calls that are in progress.
- D) Endpoints provide physical connections for the telephone network.
- E) Endpoints decide when and how to use their interconnect abilities.

Q19) Which capability of a distributed call control model endpoint is the same as a centralized call control endpoint?

A) Endpoints manage the telephone interface.

B) Endpoints autonomously initiate call setup.

C) Endpoints route calls in case a common control component fails.

D) Endpoints have a media path between them.

Q20) What may be one reason for endpoints to share common control components in a distributed call control model?

A) bandwidth conservation

B) scalability

C) ease of call setup

D) faster routing

Q21) What are two advantages of a distributed call control model? (Choose two.)

A) superior configuration control

B) fault tolerance

C) simplified introduction of new features

D) simplified deployment of additional endpoints

E) superior maintenance of the endpoint database

Q22) What are two advantages of a centralized call control model? (Choose two.)

A) minimized dependence on network resources

B) superior maintenance of the dial plan

C) simplified introduction of supplementary services

D) simplified deployment of additional endpoints

E) reduced bandwidth consumption

Quiz Answer Key

Q1) C, E

Relates to: VoIP Signaling

Q2) A

Relates to: VoIP Signaling

Q3) A, C, D

Relates to: Call Control Models

Q4) 1-D, 2-B, 3-C, 4-E, 5-A

Relates to: Call Control Models

Q5) A

Relates to: Translation Between Signaling and Call Control Models

Q6) D

Relates to: Translation Between Signaling and Call Control Models

Q7) A, C

Relates to: Call Setup

Q8) B, E

Relates to: Call Setup

Q9) B

Relates to: Call Administration and Accounting

Q10) 1-C, 2-D, 3-B, 4-A

Relates to: Call Administration and Accounting

Q11) C, E

Relates to: Call Status and Call Detail Records

Q12) A

Relates to: Call Status and Call Detail Records

Q13) B

Relates to: Address Management

Q14) D

Relates to: Address Management

Q15) B, D

Relates to: Admission Control

Q16) D

Relates to: Admission Control

Q17) B

Relates to: Centralized Call Control

Q18) D

Relates to: Centralized Call Control

Q19) D

Relates to: Distributed Call Control

Q20) B

Relates to: Distributed Call Control

Q21) B, D

Relates to: Centralized Call Control vs. Distributed Call Control

Q22) B, C

Relates to: Centralized Call Control vs. Distributed Call Control

Configuring H.323

Overview

H.323 and its associated ITU-T recommendations represent a distributed environment for establishing voice, video, and data communication in a nonguaranteed QoS network that is typical of an IP internetwork. In addition to describing how to configure H.323, this lesson discusses the features and functions of the H.323 environment, including its components and how they interact. Scalability and survivability issues are also discussed.

Relevance

An understanding of the features and functions of H.323, its components, and the manner in which the components interact is important to implement a scalable, resilient, and secure H.323 environment.

Objectives

Upon completing this lesson, you will be able to configure, monitor, and troubleshoot H.323 gateways and gatekeepers. This includes being able to meet these objectives:

- Describe the recommendations that are associated with H.323 and the control and signaling functions they perform
- List the functions that are performed by the components of an H.323 environment
- Identify three types of end-to-end connections that are established with H.323 and list eight types of registration, admission, and status protocol messages that are used to establish these connections
- Provide two scenarios of call flow without a gatekeeper
- Provide a scenario of call flow with a gatekeeper and explain the function of gatekeeper-routed call signaling in the call setup
- Describe three types of multipoint conferences supported by H.323
- Provide a scenario of call flow with multiple gatekeepers and explain how this allows scalability
- Describe four strategies that are used by H.323 to provide fault-tolerant networks
- List the steps involved in using an H.323 proxy server to set up end-to-end connections

- List the components that are supported by the Cisco Systems implementation of H.323
- Use the commands that are required to configure gateways in a two-zone, two-gatekeeper scenario
- Use the commands that are required to configure gatekeepers in a two-zone, two-gatekeeper scenario
- List the **show** and **debug** commands that are used to monitor and troubleshoot Cisco H.323 gateways and gatekeepers

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- An understanding of the objectives and principles of signaling and call control in the context of VoIP

Outline

The outline lists the topics included in this lesson.

Outline

- Overview
- H.323 and Associated Recommendations
- Functional Components of H.323
- H.323 Call Establishment and Maintenance
- Call Flows Without a Gatekeeper
- Call Flows with a Gatekeeper
- Multipoint Conferences
- Call Flows with Multiple Gatekeepers
- Survivability Strategies

 CVOICE v4.2—6-2

Outline (Cont.)

- H.323 Proxy Server
- Cisco Implementation of H.323
- Configuring H.323 Gateways
- Configuring H.323 Gatekeepers
- Monitoring and Troubleshooting
- Summary
- Quiz
- Lab

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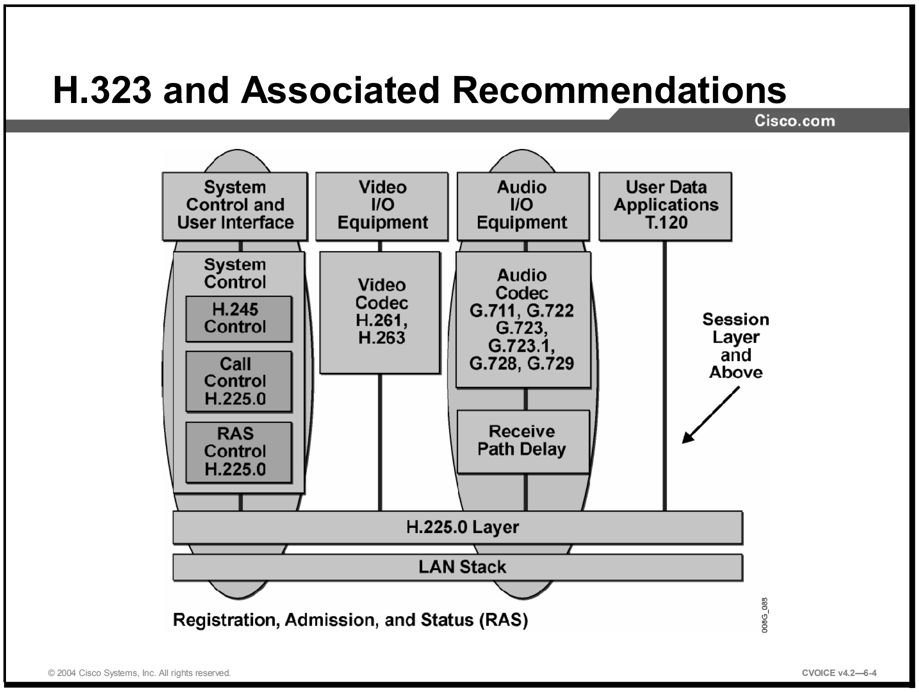
H.323 and Associated Recommendations

This topic describes H.323 and its protocols and explains how H.323 is used in the IP internetwork environment.



Recommendation H.323 describes an infrastructure of terminals, common control components, services, and protocols that are used for multimedia (voice, video, and data) communications. The figure illustrates the elements of an H.323 terminal and highlights the protocol infrastructure of an H.323 endpoint.

H.323 was originally created to provide a mechanism for transporting multimedia applications over LANs. Although numerous vendors still use H.323 for videoconferencing applications, it has rapidly evolved to address the growing needs of VoIP networks. H.323 is currently the most widely used VoIP signaling and call control protocol, with international and domestic carriers relying on it to handle billions of minutes of use each year.

H.323 is considered an "umbrella protocol" because it defines all aspects of call transmission, from call establishment to capabilities exchange to network resource availability. H.323 defines the following protocols:

- H.245 for capabilities exchange
- H.225.0 for call setup
- H.225.0 for registration, admission, and status (RAS) control for call routing

H.323 is based on the ISDN Q.931 protocol, which allows H.323 to easily interoperate with legacy voice networks, such as the public switched telephone network (PSTN) or SS7. In addition to providing support for call setup, H.225.0 provides a message transport mechanism for the H.245 control function and the RAS signaling function. Following is a description of these functions:

- **Call-signaling function:** The call-signaling function uses a call-signaling channel that allows an endpoint to create connections with other endpoints. The call-signaling function defines call setup procedures, based on the call setup procedures for ISDN (Recommendation Q.931). The call-signaling function uses messages formatted according to H.225.0.
- **H.245 control function:** The H.245 control function uses a control channel to transport control messages between endpoints or between an endpoint and a common control component, such as a gatekeeper or multipoint controller (MC). The control channel used by the H.245 control function is separate from the call-signaling channel.

  The H.245 control function is responsible for the following:

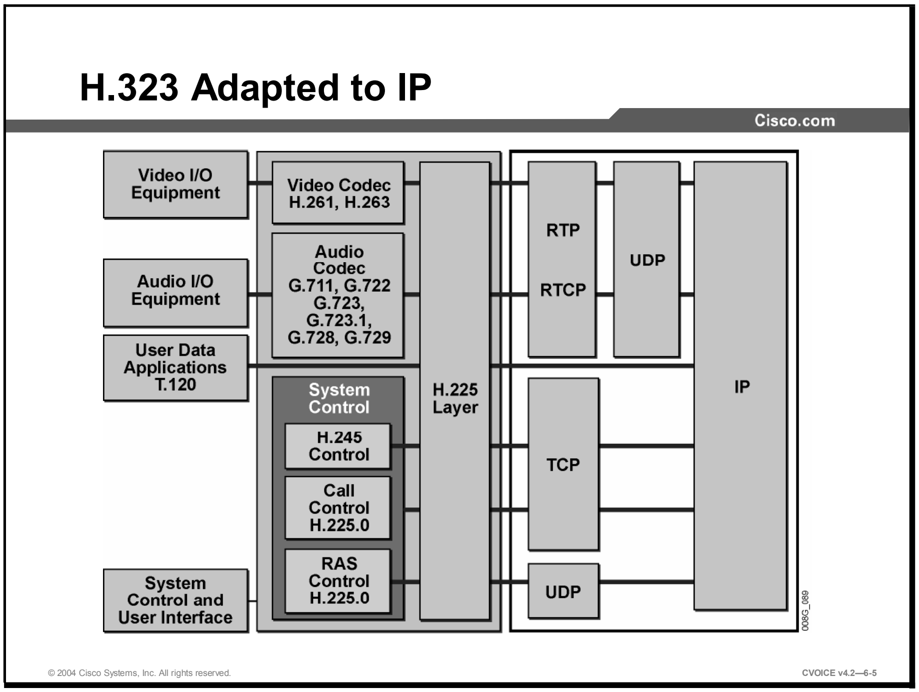
  — **Logical channel signaling:** Opens and closes the channel that carries the media stream

  — **Capabilities exchange:** Negotiates audio, video, and codec capability between the endpoints

  — **Master or responder determination:** Determines which endpoint is master and which is responder; used to resolve conflicts during the call

  — **Mode request:** Requests a change in mode, or capability, of the media stream

  — **Timer and counter values:** Establishes values for timers and counters and agreement of those values by the endpoints
- **RAS signaling function:** The RAS signaling function uses a separate signaling channel (RAS channel) to perform registration, admissions, bandwidth changes, status, and disengage procedures between endpoints and a gatekeeper. The RAS signaling function uses messages formatted according to H.225.0.

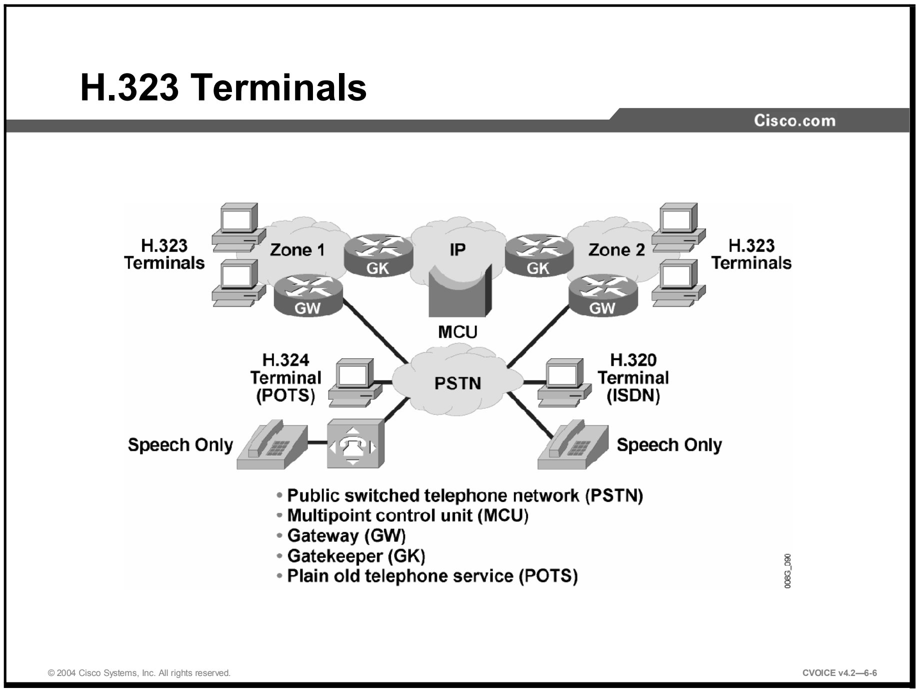


Example: H.323 Adapted to IP

A typical implementation of H.323 goes beyond the original LAN context of H.323. The figure illustrates a specific application of H.323 on an IP internetwork. Notice that real-time aspects of H.323 rely on UDP. Both the session-oriented control procedures and the data media type of H.323 use TCP.

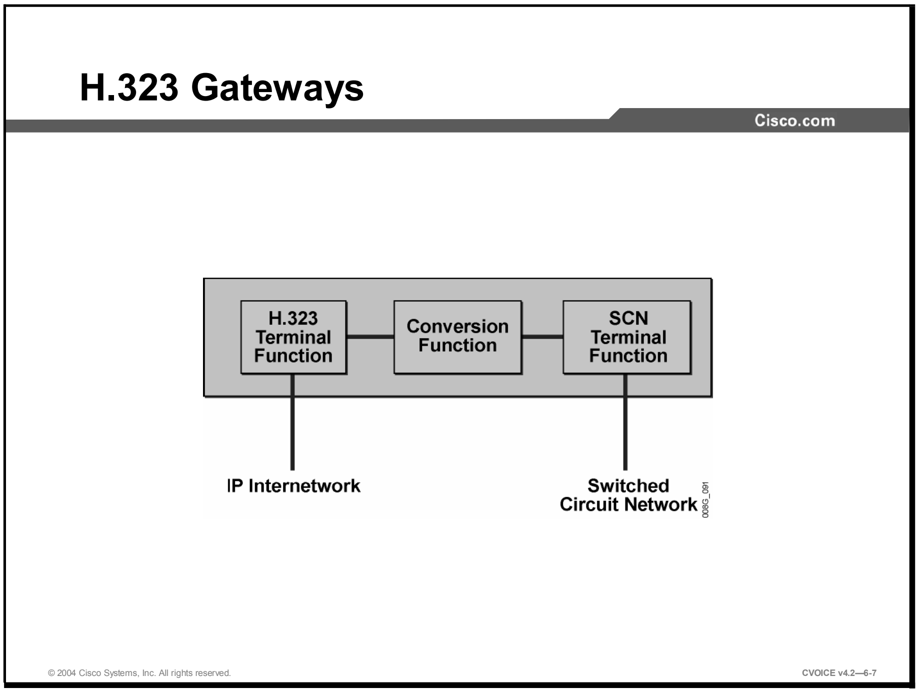
Functional Components of H.323

This topic describes the functional components that make up an H.323 environment.



An H.323 terminal is an endpoint that provides real-time voice (and optionally, video and data) communications with another endpoint, such as an H.323 terminal, gateway, or multipoint control unit (MCU).

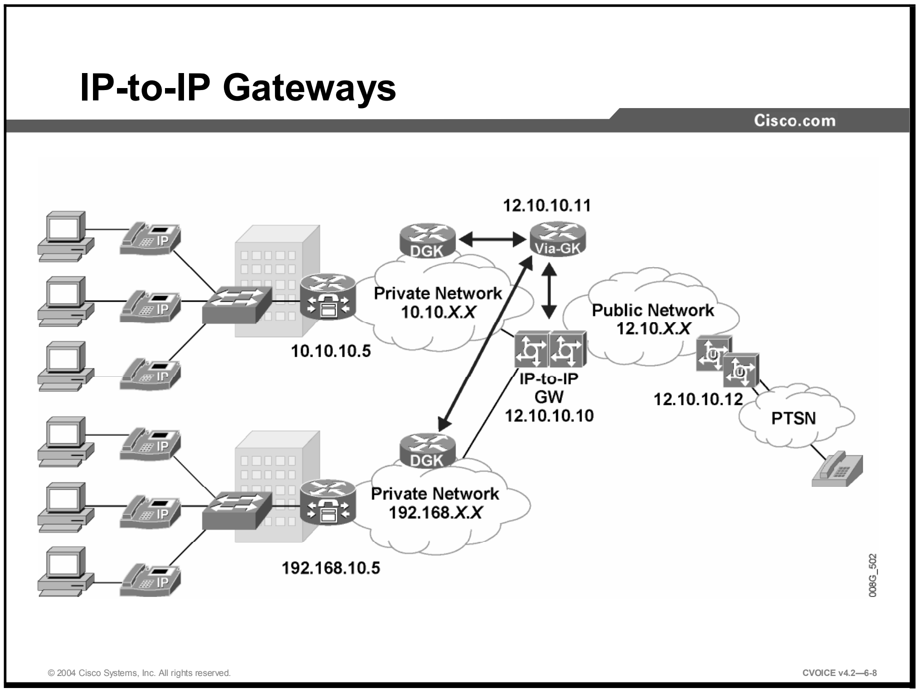
An H.323 terminal must be capable of transmitting and receiving G.711 (a-law and µ-law) 64-kbps pulse code modulation (PCM)-encoded voice, and may support other encoded voice formats, such as G.729 and G.723.1.



An H.323 gateway is an optional type of endpoint that provides interoperability between H.323 endpoints and endpoints located on a switched-circuit network (SCN), such as the PSTN or an enterprise voice network. Ideally, the gateway is transparent to both the H.323 endpoint and the SCN-based endpoint.

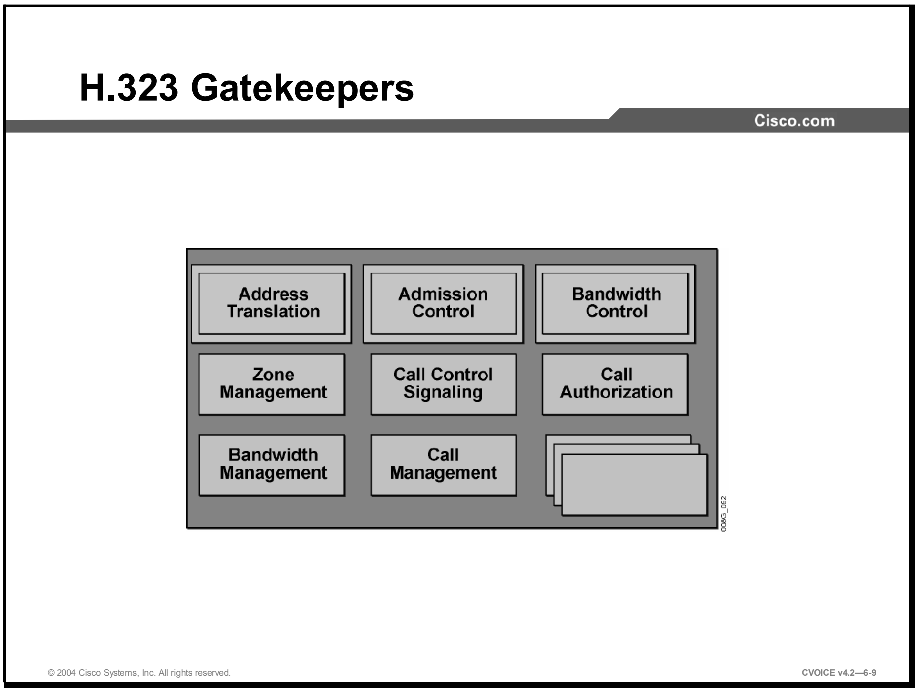
An H.323 gateway performs the following services:

- Translation between audio, video, and data formats
- Conversion between call setup signals and procedures
- Conversion between communication control signals and procedures



The IP-to-IP gateway facilitates easy and cost-effective connectivity between independent VoIP service provider networks. Some in the industry call IP-to-IP gateways "border elements" or "session border controllers." The IP-to-IP gateway provides a network-to-network interface point for billing, security, Cisco CallManager interconnectivity, call admission control, and signaling interworking. It will perform most of the same functions of a PSTN-to-IP gateway, but will join two VoIP call legs. Media packets can either flow through the gateway and hide the networks from each other, or flow around the IP-to-IP gateway if network security is not of primary importance.

The figure illustrates a basic IP-to-IP gateway network. From the perspective of the private, or customer, networks, the IP-to-IP gateway will appear as a single public address that must be routable on their private networks (in this case a 12.*x.x.x* address routable on the 10.10.*x.x* and 192.168.*x.x* networks). Care must be taken at the IP-to-IP gateway to ensure that proper routing restrictions are in place to prevent communication directly between the private networks attached to it. Also note that this model works only if no overlapping address schemes are used on the customers' networks. Finally, to the hop-off gateways on the public network, all calls will appear to originate from the 12.*x.x.x* address of the IP-to-IP gateway and not the private addresses on the customer networks. Also note that the gatekeepers shown in the diagram control each zone independently, with the 12.10.10.11 gatekeeper acting as the control point for the public network, and therefore the IP-to-IP gateway.



An H.323 gatekeeper is an optional component that provides call control support and services to H.323 endpoints. Although a gatekeeper is considered a distinct and optional component, it can be colocated with any other H.323 component.

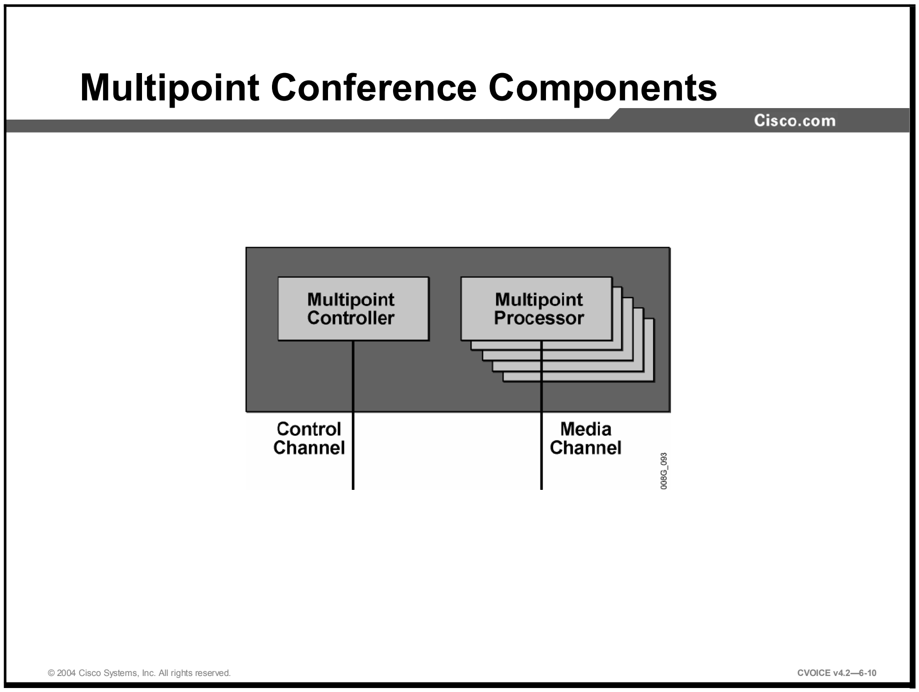
The scope of endpoints over which a gatekeeper exercises its authority is called a “zone.” H.323 defines a one-to-one relationship between a zone and a gatekeeper.

When a gatekeeper is included, it must perform the following functions:

- **Address translation:** Converts an alias address to an IP address
- **Admission control:** Limits access to network resources based on call bandwidth restrictions
- **Bandwidth control:** Responds to bandwidth requests and modifications
- **Zone management:** Provides services to registered endpoints

The gatekeeper may also perform:

- **Call control signaling:** Performs call signaling on behalf of the endpoint (gatekeeper-routed call signaling)
- **Call authorization:** Rejects calls based on authorization failure
- **Bandwidth management:** Limits the number of concurrent accesses to IP internetwork resources (Call Admission Control [CAC])
- **Call management:** Maintains a record of ongoing calls



Support for multipoint conferences is provided by the following three functional components:

- **Multipoint controller:** An MC provides the functions that are necessary to support conferences involving three or more endpoints. The MC establishes an H.245 control channel with each of the conference participants. Through the control channel, the MC completes a capability exchange during which the MC indicates the mode of the conference (decentralized or centralized).

  An MC is not modeled as a standalone component; it may be located with an endpoint (terminal or gateway), a gatekeeper, or a multipoint control unit.

- **Multipoint processor:** A multipoint processor (MP) adds functionality to multipoint conferences. An MP can receive multiple streams of multimedia input, process the streams by switching and mixing the streams, and then retransmit the result to all or some of the conference members.

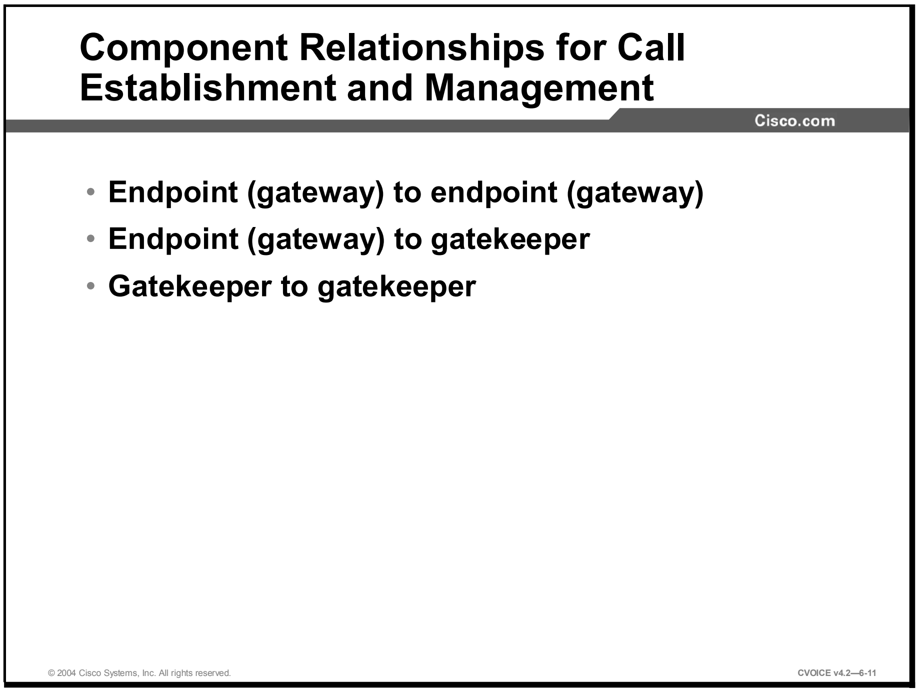
  Similar to an MC, an MP is not modeled as a standalone component; it resides in an MCU.

- **Multipoint control unit:** An MCU is modeled as an endpoint that provides support for multipoint conferences by incorporating one MC and zero or more MPs.

  An MCU is modeled as a standalone component.

H.323 Call Establishment and Maintenance

This topic describes possible component scenarios required to establish end-to-end connections and the commands used by the components to establish VoIP calls.



Although H.323 is based on the concepts of a distributed call control model, it often embodies centralized call control model concepts. Calls can be established between any of the following components:

- **Endpoint to endpoint:** The intelligence of H.323 endpoints allows them to operate autonomously. In this mode of operation, endpoints locate other endpoints through nonstandard mechanisms and initiate direct communication between the endpoints.
- **Endpoint to gatekeeper:** When a gatekeeper is added to the network, endpoints interoperate with the gatekeeper using the RAS channel.
- **Gatekeeper to gatekeeper:** In the presence of multiple gatekeepers, gatekeepers communicate with each other on the RAS channel.

RAS Messages

Cisco.com

Gatekeeper Discovery	Location Request
GatekeeperRequest (GRQ)	LocationRequest (LRQ)
GatekeeperConfirm (GCF)	LocationConfirm (LCF)
GatekeeperReject (GRJ)	LocationReject (LRJ)
Terminal/Gateway Registration	**Call Admission**
RegistrationRequest (RRQ)	AdmissionRequest (ARQ)
RegistrationConfirm (RCF)	AdmissionConfirm (ACF)
RegistrationReject (RRJ)	AdmissionReject (ARJ)
Terminal/Gateway Unregistration	**Disengage**
UnregistrationRequest (URQ)	DisengageRequest (DRQ)
UnregistrationConfirm (UCF)	DisengageConfirm (DCF)
UnregistrationReject (URJ)	DisengageReject (DRJ)

Bandwidth Change	Status Queries
Bandwidth Change Request (BRQ)	InfoRequest (IRQ)
Bandwidth Change Confirm (BCF)	InfoRequestResponse (IRR)
Bandwidth Change Reject (BRJ)	InfoRequestAck (IACK)
	InfoRequestNak (INAK)

 CVOICE v4.2—6-12

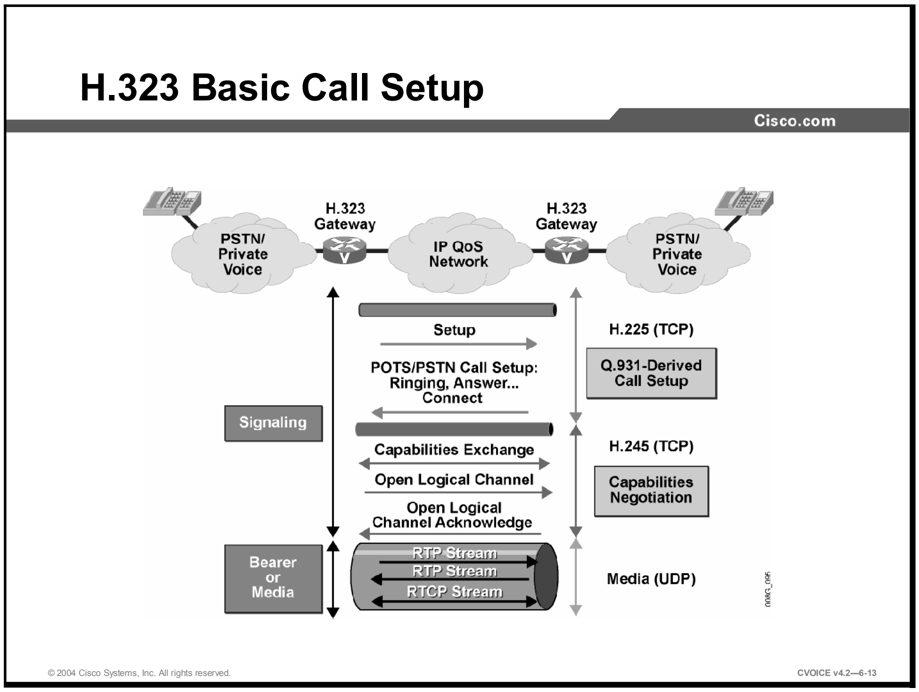
Gatekeepers communicate through the RAS channel using different types of RAS messages. These message types include the following:

- **Gatekeeper discovery:** An endpoint multicasts a gatekeeper discovery request (GRQ). A gatekeeper may confirm (gatekeeper confirmation [GCF]) or reject (gatekeeper rejection [GRJ]) an endpoint.
- **Terminal/gateway registration:** An endpoint sends a registration request (RRQ) to its gatekeeper to register and provide reachable prefixes. A gatekeeper confirms (registration confirmation [RCF]) or rejects (registration rejection [RRJ]) the registration.
- **Terminal/gateway unregistration:** An endpoint or gatekeeper sends an unregistration request (URQ) to cancel a registration. The responding device confirms (unregistration request [UCF]) or rejects (unregistration rejection [URJ]) the request.
- **Location request:** An endpoint or gatekeeper sends a location request (LRQ) to a gatekeeper. An LRQ is sent directly to a gatekeeper if one is known, or it is multicast to the gatekeeper discovery multicast address. An LRQ requests address translation of an E.164 address and solicits information about the responsible endpoint. The responding gatekeeper confirms (location confirmation [LCF]) with the IP address of the endpoint or rejects the request (location rejection [LRJ]) if the address is unknown.
- **Call admission:** An endpoint sends an admission request (ARQ) to its gatekeeper. The request identifies the terminating endpoint and the bandwidth required. The gatekeeper confirms (admission confirmation [ACF]) with the IP address of the terminating endpoint or rejects (admission rejection [ARJ]) if the endpoint is unknown or inadequate bandwidth is available.
- **Bandwidth change:** An endpoint sends a bandwidth change request (BRQ) to its gatekeeper to request an adjustment in call bandwidth. A gatekeeper confirms (bandwidth confirmation [BCF]) or rejects (bandwidth rejection [BRJ]) the request.
- **Disengage:** When a call is disconnected, the endpoint sends a disengage request (DRQ) to the gatekeeper. The gatekeeper confirms (disengage confirmation [DCF]) or rejects (disengage rejection [DRJ]) the request.

- **Status queries:** A gatekeeper uses status request (IRQ) to determine the status of an endpoint. In its response (IRR), the endpoint indicates whether it is online or offline. The endpoint may also reply that it understands the information request (information request acknowledged [IACK]) or that it does not understand the request (information request not acknowledged [INAK]).

Call Flows Without a Gatekeeper

This topic describes call setup scenarios without a gatekeeper and provides examples of actual call-flow procedures.



The figure shows an H.323 basic call setup exchange between two gateways. The optional gatekeeper is not present in this example. Although gateways are shown, the same procedure is used when one or both endpoints are H.323 terminals.

The flow procedure without a gatekeeper includes these steps:

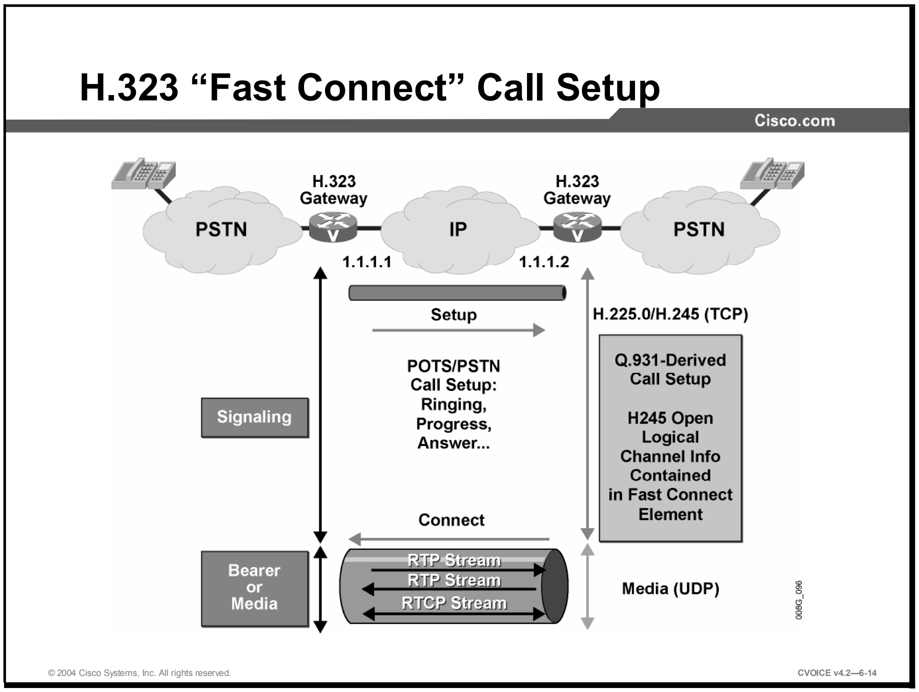
1. The originating gateway initiates an H.225.0 session with the destination gateway on registered TCP port 1720. The gateway determines the IP address of the destination gateway internally. The gateway has the IP address of the destination endpoint in its configuration or it knows a Domain Name System (DNS) resolvable domain name for the destination.

2. Call setup procedures based on Q.931 create a call-signaling channel between the endpoints.

3. The endpoints open another channel for the H.245 control function. The H.245 control function negotiates capabilities and exchanges logical channel descriptions.

4. The logical channel descriptions open RTP sessions.

5. The endpoints exchange multimedia over the RTP sessions.

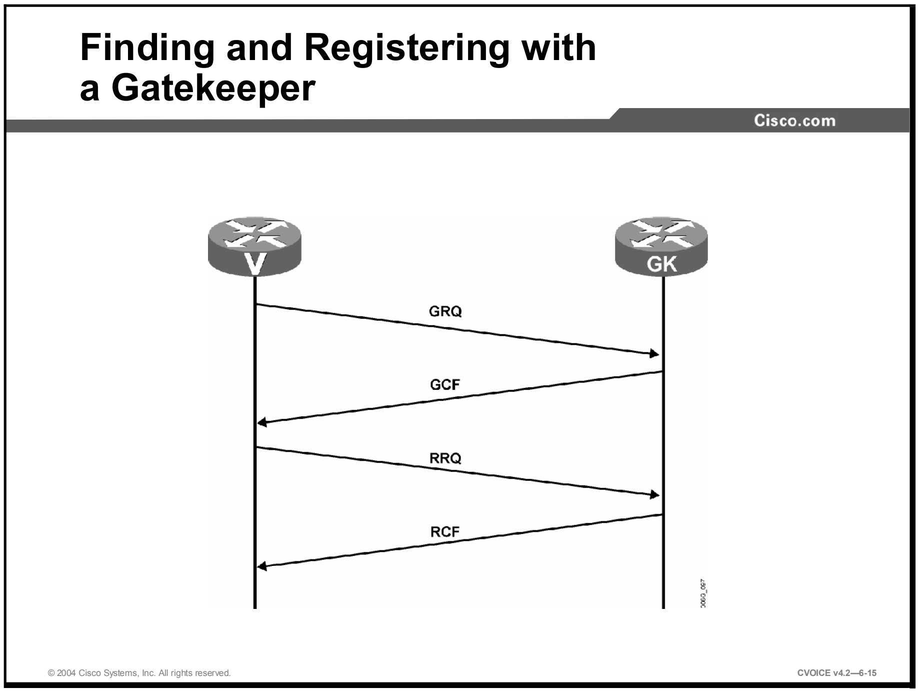


The figure shows an H.323 setup exchange that uses the Fast Connect abbreviated procedure available in version 2 of Recommendation H.323. The Fast Connect procedure reduces the number of round-trip exchanges and achieves the capability exchange and logical channel assignments in one round trip.

The Fast Connect procedure includes these steps:

1. The originating gateway initiates an H.225.0 session with the destination gateway on registered TCP port 1720.
2. Call setup procedures based on Q.931 create a combined call-signaling channel and control channel for H.245. Capabilities and logical channel descriptions are exchanged within the Q.931 call setup procedure.
3. Logical channel descriptions open RTP sessions.
4. The endpoints exchange multimedia over the RTP sessions.

Note Cisco H.323 voice equipment supports up to version 4 of H.323 and is backward compatible to earlier versions.



The figure illustrates how an endpoint locates and registers with a gatekeeper. A gatekeeper adds scalability to H.323. Without a gatekeeper, an endpoint must recognize or have the ability to resolve the IP address of the destination endpoint.

Before an endpoint can use a gatekeeper, it must register with the gatekeeper. To register, an endpoint must recognize the IP address of the gatekeeper.

One of these two methods are used to determine the address of the gatekeeper:

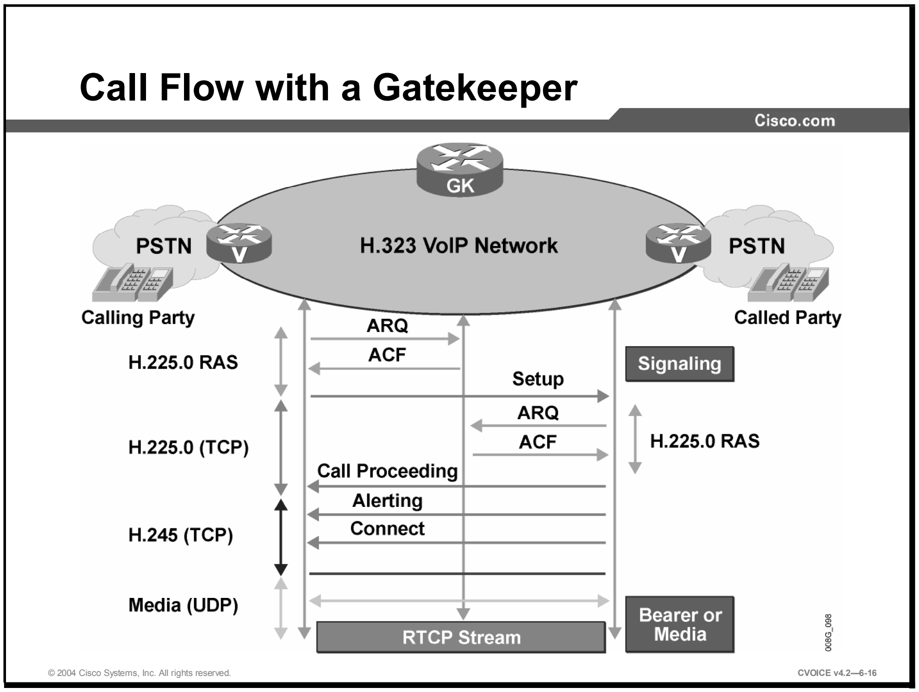
- An endpoint can be preconfigured to recognize the domain name or IP address of its gatekeeper. If configured to recognize the name, an endpoint must have a means to resolve the name to an IP address. A common address resolution technique is the DNS.
- An endpoint can issue a multicast GRQ to the gatekeeper discovery address (224.0.1.41) to discover the IP address of its gatekeeper. If the endpoint receives a GCF to the request, it uses the IP address to proceed with registration.

To initiate registration, an endpoint sends an RRQ to the gatekeeper. In the register request, the endpoint identifies itself with its ID and provides its IP address. Optionally, the endpoint lists the prefixes (for example, telephone numbers) that it supports. These prefixes are gleaned from the plain old telephone service (POTS) dial-peer destination patterns associated with any FXS port.

With this procedure, a gatekeeper determines the location and identity of endpoints and the identities of SCN endpoints from gateway registrations.

Call Flows with a Gatekeeper

This topic discusses call setup scenarios with a gatekeeper.



The exchanges in the figure illustrate the use of a gatekeeper by both endpoints. In this example, both endpoints have registered with the same gatekeeper. Call flow with a gatekeeper proceeds as follows:

1. The gateway sends an ARQ to the gatekeeper to initiate the procedure. The gateway is configured with the domain or address of the gatekeeper.

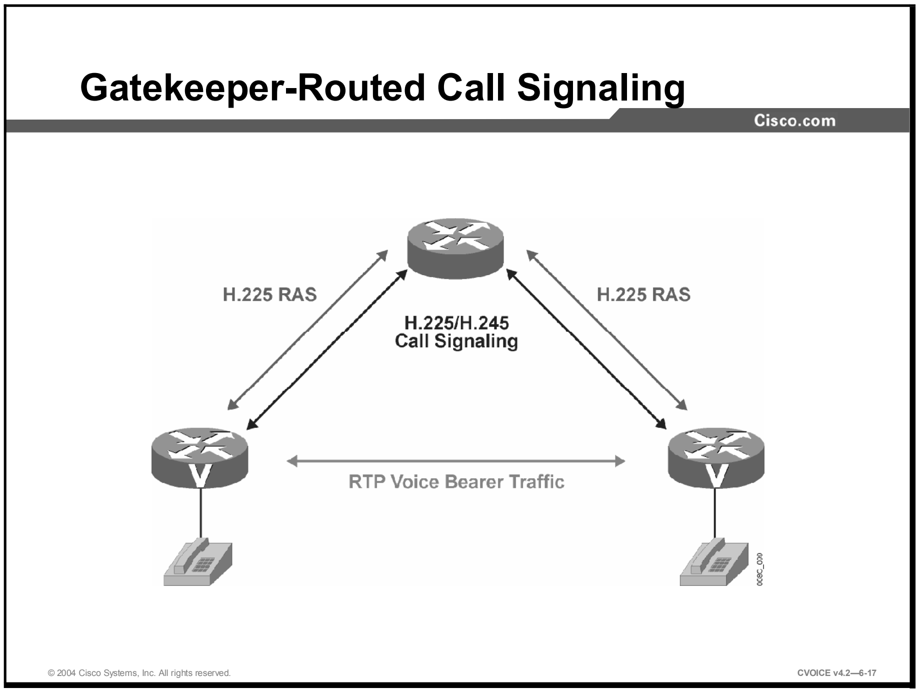
2. The gatekeeper responds to the admission request with an ACF. In the confirmation, the gatekeeper provides the IP address of the remote endpoint.

3. When the originating endpoint identifies the terminating endpoint, it initiates a basic call setup.

4. Before the terminating endpoint accepts the incoming call, it sends an ARQ to the gatekeeper to gain permission.

5. The gatekeeper responds affirmatively, and the terminating endpoint proceeds with the call setup procedure.

During this procedure, if the gatekeeper responds to either endpoint with an ARJ to the admission request, the endpoint that receives the rejection terminates the procedure.

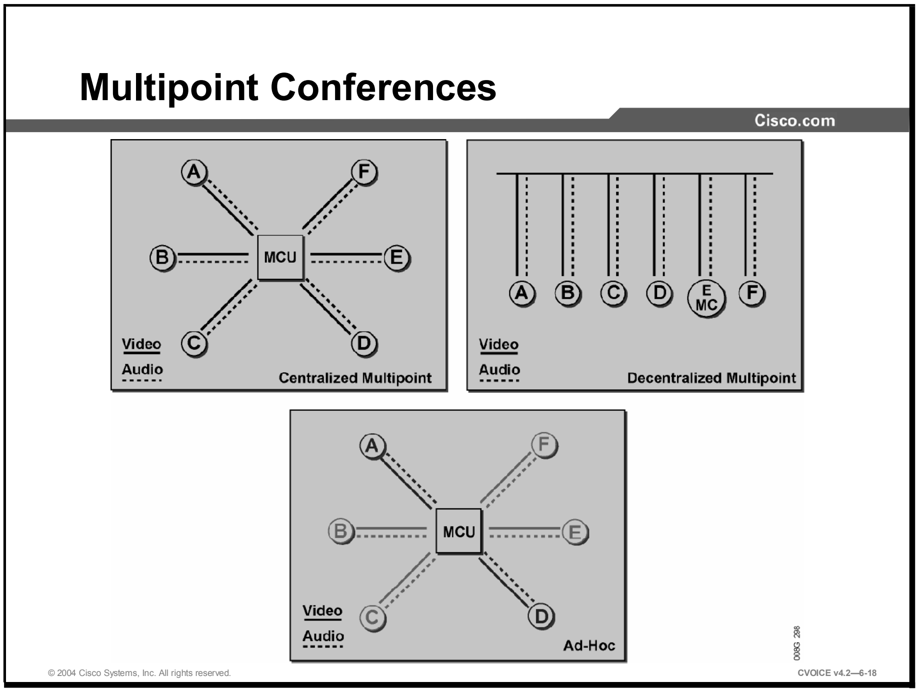


In the previous examples, the call-signaling channel is created from endpoint to endpoint. In some cases, it is desirable to have the gatekeeper represent the other endpoint for signaling purposes. This method is called gatekeeper-routed call signaling. The process for gatekeeper-routed call signaling is as follows:

1. The gatekeeper responds to an admission request and advises the endpoint to perform the call setup procedure with the gatekeeper, not with the terminating endpoint.
2. The endpoint initiates the setup request with the gatekeeper.
3. The gatekeeper sends its own request to the terminating endpoint and incorporates some of the details acquired from the originating request.
4. When a connect message is received from the terminating endpoint, the gatekeeper sends a connect to the originating endpoint.
5. The two endpoints establish an H.245 control channel between them. The call procedure continues normally from this point.

Multipoint Conferences

H.323 defines three types of multipoint conferences: centralized, distributed, and ad-hoc. H.323 also defines a hybrid of the first two. This topic describes the multipoint conference control components used to support these conferences.



All types of multipoint conferences rely on a single MC to coordinate the membership of a conference. Each endpoint has an H.245 control channel connection to the MC. Either the MC or the endpoint initiates the control channel setup. H.323 defines the following three types of conferences:

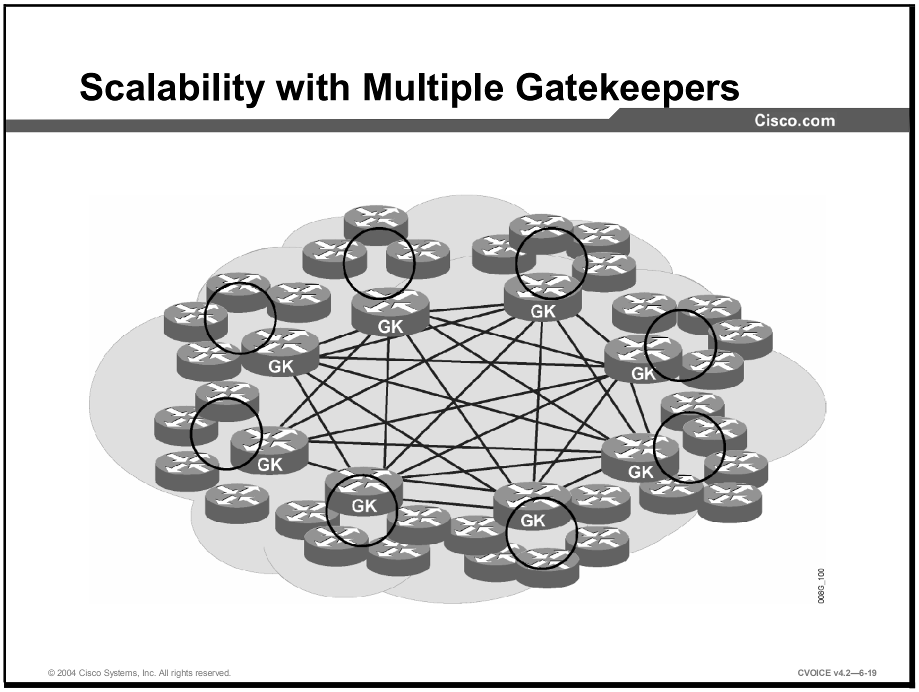
- **Centralized multipoint conference:** The endpoints must have their audio, video, or data channels connected to an MP. The MP performs mixing and switching of the audio, video, and data, and if the MP supports the capability, each endpoint can operate in a different mode.
- **Distributed multipoint conference:** The endpoints do not have a connection to an MP. Instead, endpoints multicast their audio, video, and data streams to all participants in the conference. Because an MP is not available for switching and mixing, any mixing of the conference streams is a function of the endpoint, and all endpoints must use the same communication parameters.

To accommodate situations in which two streams (audio and video) would be handled by the different multipoint conference models, H.323 defines a “hybrid.” A hybrid describes a situation in which the audio and video streams are managed by a single H.245 control channel with the MC, but where one stream relies on multicast (according to the distributed model) and the other uses the MP (as in the centralized model).

- **Ad-hoc multipoint conference:** Any two endpoints in a call can convert their relationship into a point-to-point conference. If neither of the endpoints has a colocated MC, then the services of a gatekeeper are used. When the point-to-point conference is created, other endpoints become part of the conference by accepting an invitation from a current participant, or the endpoint can request to join the conference.

Call Flows with Multiple Gatekeepers

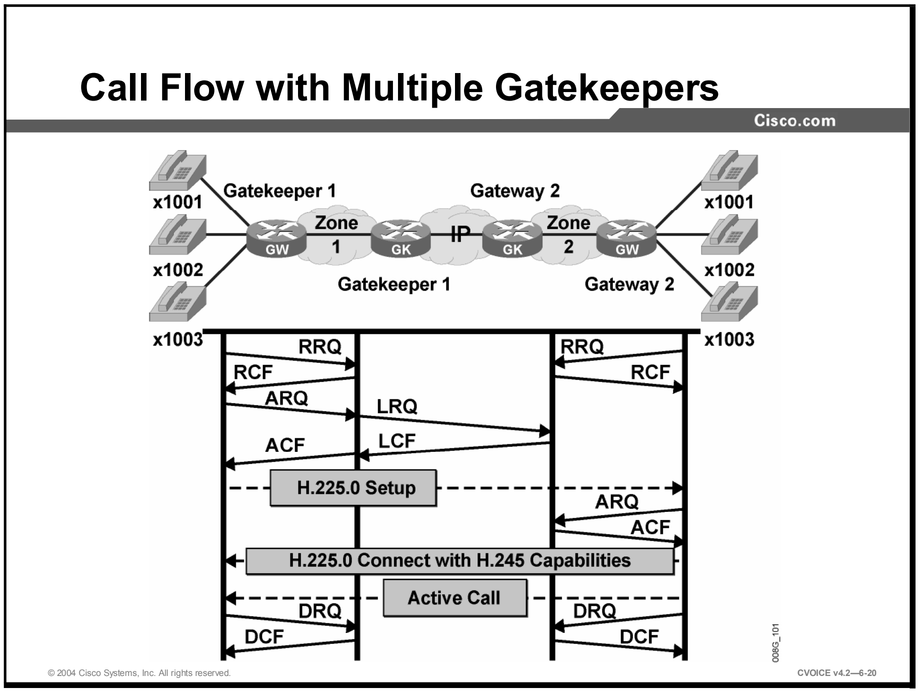
By simplifying configuration of the endpoints, gatekeepers aid in building large-scale VoIP networks. As the VoIP network grows, incorporating additional gatekeepers enhances the network scalability. This topic discusses the use of multiple gatekeepers for scalability and illustrates call flow in a multiple gatekeeper environment.



Without a gatekeeper, endpoints must find each other by any means available. This limits the growth potential of the VoIP network. Through the registration and address resolution services of a gatekeeper, growth potential improves significantly.

A single gatekeeper design may not be appropriate for several reasons. A single gatekeeper can become overloaded, or it can have an inconvenient network location, necessitating a long and expensive round trip to it.

Deploying multiple gatekeepers offers a more scalable and robust environment.

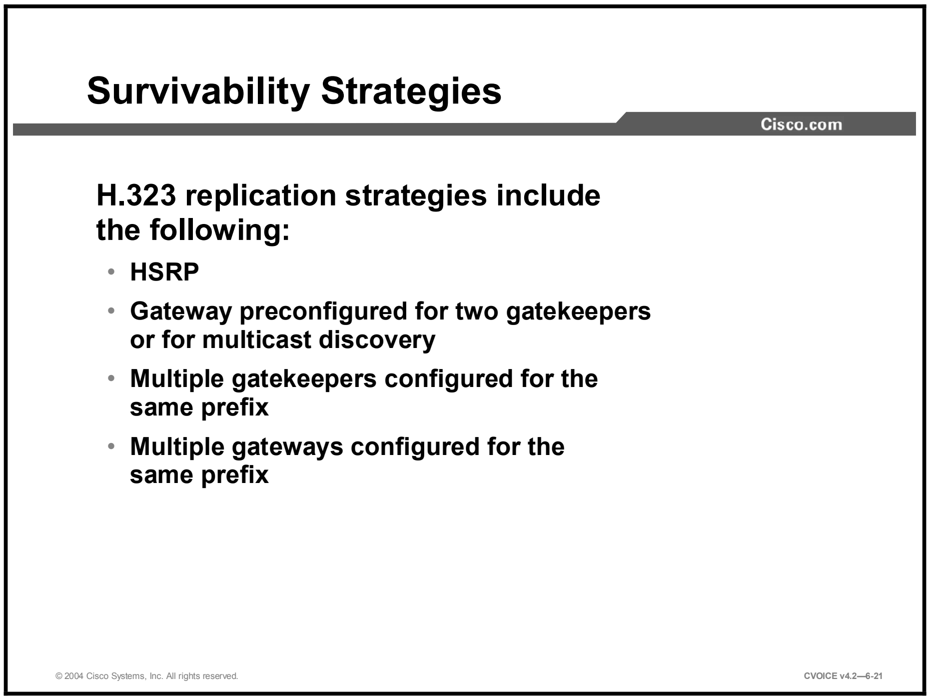


The figure illustrates a call setup involving two gatekeepers. In this example, each endpoint is registered with a different gatekeeper. Notice the changes in the following call setup procedure:

1. The originating endpoint sends an admission request to its gatekeeper requesting permission to proceed and asking for the session parameters for the terminating endpoint.
2. The gatekeeper for the originating endpoint (gatekeeper 1) determines from its configuration or from a directory resource that the terminating endpoint is potentially associated with gatekeeper 2. Gatekeeper 1 sends an LRQ to gatekeeper 2.
3. Gatekeeper 2 recognizes the address and sends back an LCF. In the confirmation, gatekeeper 2 provides the IP address of the terminating endpoint.
4. If gatekeeper 1 considers the call acceptable for security and bandwidth reasons, it maps the LCF to an ARQ and sends the confirmation back to the originating endpoint.
5. The endpoint initiates a call setup to the remote endpoint.
6. Before accepting the incoming call, the remote endpoint sends an ARQ to gatekeeper 2 requesting permission to accept the incoming call.
7. Gatekeeper 2 performs admission control on the request and responds with a confirmation.
8. The endpoint responds to the call setup request.
9. The call setup progresses through the H.225.0 call function and H.245 control function procedures until the RTP sessions are initiated.
10. At the conclusion of the call, each endpoint sends a disconnect request to its gatekeeper to advise the gatekeeper that the call is complete.
11. The gatekeeper responds with a confirmation.

Survivability Strategies

Maintaining high availability in an H.323 environment requires a design that accommodates failure of a critical component. This topic describes strategies for maintaining VoIP service.



In any environment that depends on common control components, the vulnerability of the environment is directly proportional to the probability of common control component failure. In a classical telephone application, fault tolerance is accommodated by incorporating extra common control technology. One strategy replicates all critical components. This expensive approach is often replaced with the more cost-effective solution of "*n* out of *n* + 1" redundancy; a single spare component is available to step in when any one of the active *n* components fails. The essential part of either strategy is the replication of key components.

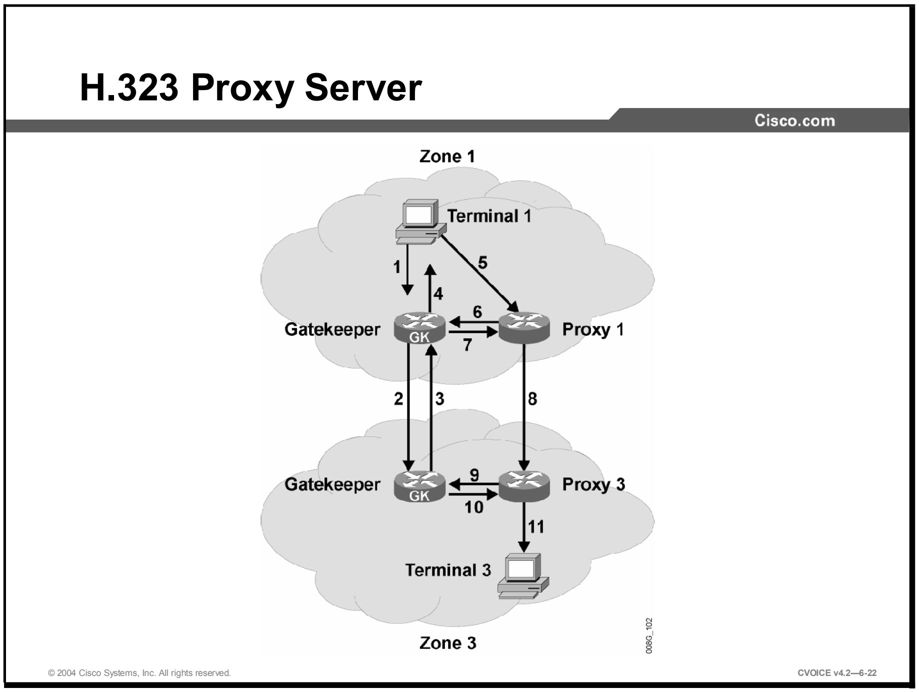
In H.323, the key components are the gateways and the gatekeeper. H.323 can employ any of the following strategies:

- **Hot Standby Router Protocol:** The Hot Standby Router Protocol (HSRP) allows two gatekeepers to share both an IP address and access to a common LAN; however, at any time, only one is active. Endpoints are configured with the name of the gatekeeper, which they can resolve using DNS or the IP address of the gatekeeper.
- **Multiple gatekeepers with gatekeeper discovery:** Deployment of multiple gatekeepers reduces the probability of total loss of gatekeeper access. However, adding new gatekeepers presents a new challenge. Each gatekeeper creates a unique H.323 zone. Because an H.323 endpoint is associated with only one gatekeeper at a time (in only one zone at a time), endpoints are configured to find only one of several working gatekeepers. Fortunately, a gateway can be configured with an ordered list of gatekeepers, or to use IP multicast to locate a gatekeeper.

- **Multiple gatekeepers configured for the same prefix:** Gatekeepers send location request messages to other gatekeepers when locating an endpoint. By supporting the same prefix on multiple gatekeepers, the location request can be resolved by multiple gatekeepers. This strategy makes the loss of one gatekeeper less significant.
- **Multiple gateways configured for the same prefix:** Survivability is enhanced at the gateway with multiple gateways that are configured to reach the same SCN destination. By configuring the same prefix of destinations in multiple gateways, the gatekeeper sees the same prefix more than once as each gateway registers with its gatekeeper.

H.323 Proxy Server

This topic describes a call setup scenario involving a proxy server.



An H.323 proxy server can circumvent the shortcomings of a direct path in cases where the direct path between two H.323 endpoints is not the most appropriate; for example, when the direct path has poor throughput and delay characteristics, is not easily available because of a firewall, or zones are configured as inaccessible on the gatekeepers in order to isolate addressing information in different zones.

When a proxy server is involved, two sessions are typically established as follows:

- Originating endpoint to the proxy server
- Proxy server to the terminating endpoint

However, when a proxy server also represents the terminating endpoint, a third session is required, as follows:

- Originating endpoint to proxy server 1
- Proxy server 1 to proxy server 2
- Proxy server 2 to terminating endpoint

Example: H.323 Proxy

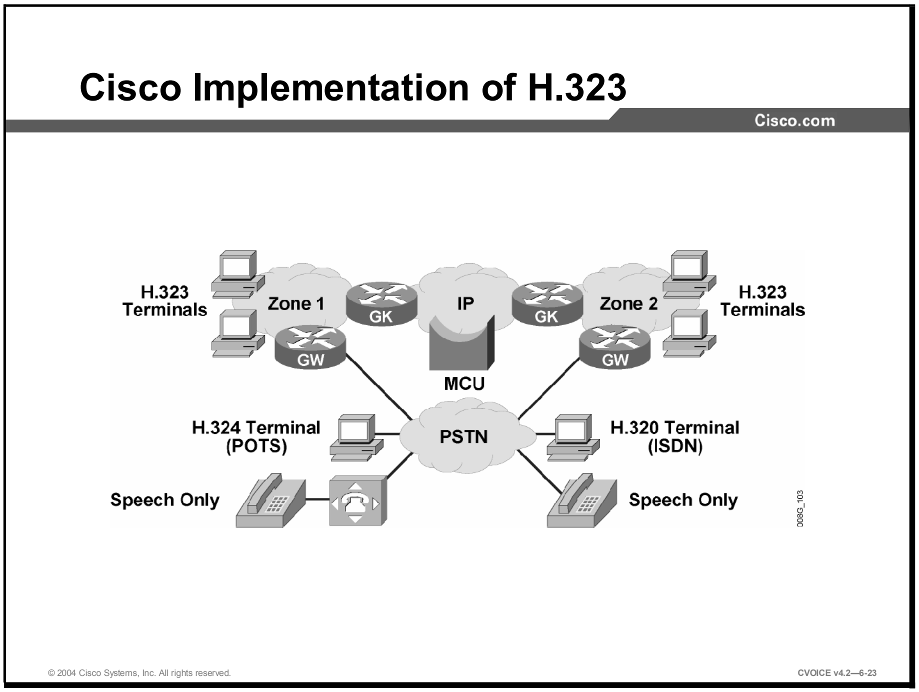
The figure illustrates an example with three sessions. The objective in this scenario is for terminal 1 and terminal 3 to establish an end-to-end relationship for multimedia communications. The following sequence of events occurs:

1. Terminal 1 asks gatekeeper 1 for permission to call terminal 3.
2. Gatekeeper 1 locates gatekeeper 3 as the terminal 3 gatekeeper. Gatekeeper 1 asks gatekeeper 3 for the address of terminal 3.
3. Gatekeeper 3 responds with the address of proxy 3 (instead of the address of terminal 3) to hide the identity of terminal 3.
4. Gatekeeper 1 is configured to get to proxy 3 by way of proxy 1, so gatekeeper 1 returns the address of proxy 1 to terminal 1.
5. Terminal 1 calls proxy 1.
6. Proxy 1 consults gatekeeper 1 to discover the true destination of the call, which is terminal 3 in this example.
7. Gatekeeper 1 instructs proxy 1 to call proxy 3.
8. Proxy 1 calls proxy 3.
9. Proxy 3 consults gatekeeper 3 for the true destination, which is terminal 3.
10. Gatekeeper 3 gives the address of terminal 3 to proxy 3.
11. Proxy 3 completes the call to terminal 3.

Notice that the resulting path between terminal 1 and terminal 3 involves three separate legs; one between terminal 1 and proxy 1, one between proxy 1 and proxy 3, and one between proxy 3 and terminal 3. Both the media and any signaling are carried over these three legs.

Cisco Implementation of H.323

This topic discusses how Cisco implements H.323.

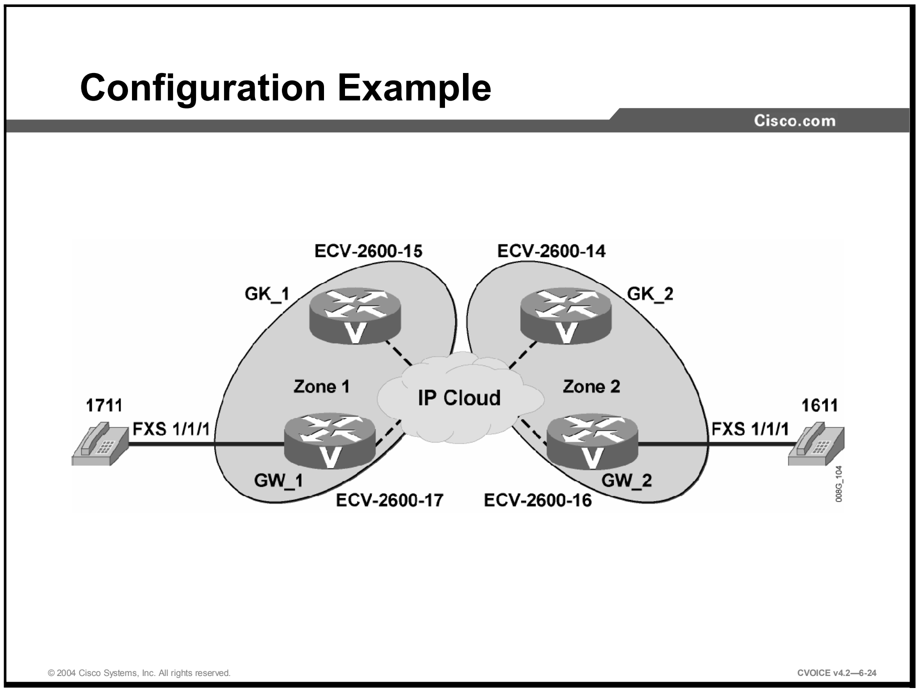


Cisco provides support for all H.323 components. These H.323 components include the following:

- **H.323 terminals:** Cisco provides support for H.323 terminals in Cisco IP Phone.
- **Gateways:** Cisco implements H.323 gateway support in:
  - Cisco voice-enabled routers (first available in Cisco IOS Release 11.3)
  - Cisco SC2200 Signaling Controllers
  - Cisco PGW 2200 PSTN gateways
  - Voice-enabled Cisco AS5*xx*0 access servers
  - Cisco BTS 10200 Softswitch
- **Gatekeepers:** Cisco implements gatekeeper support in:
  - Cisco Multimedia Conference Manager
  - Cisco CallManager
  - Routers (first available in Cisco IOS Release 11.3)
- **Multipoint control unit:** The MC and MP of the Cisco IP/VC 3500 Series MCU support all H.323 conference types. The IP/VC 3500 also incorporates a gatekeeper.
- **Other support:** Cisco PIX 500 Series firewalls and Context-Based Access Control (CBAC) support in the Cisco Secure Integrated Software monitor the logical channel handshaking of the H.245 control function and dynamically open conduits for the RTP sessions.

Configuring H.323 Gateways

This topic illustrates and describes the configuration commands used to create a two-zone, two-gatekeeper scenario.



The figure illustrates the scenario on which the gateway configurations are based.

Configuring the Gateways

Cisco.com

Gateway 1

```
hostname ECV-2610-17
!
interface Ethernet0/0
 ip address 10.52.218.49 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id gk-zone1.test.com ipaddr 10.52.218.47 1718
 h323-gateway voip h323-id gw_1
h323-gateway voip bind srcaddr 10.52.218.49
!
dial-peer voice 1 voip
 destination-pattern 16..
 session target ras
!
dial-peer voice 2 pots
 destination-pattern 911
 port 1/1/1
 no register e164
!
Gateway
!
end
```

008G_503

 CVOICE v4.2—6-25

To use a gatekeeper, the user must complete the following three tasks on the gateway:

1. Enable the gateway with the **gateway** command.
2. Configure the relationship with the gatekeeper. This requires three interface subcommands:
   - **h323-gateway voip interface:** Tells the router that this interface should be enabled for H.323 packet processing.
   - **h323-gateway voip id:** Identifies the ID of the gatekeeper.
   - **h323-gateway voip h323-id:** Configures the ID of this router. When the router registers with the gatekeeper, the gatekeeper recognizes the gateway by this ID.
3. Configure a dial peer to use the gatekeeper with the **ras** parameter on the dial peer subcommand **session target**.

Configuring the Gateways (Cont.)

Cisco.com

Gateway 2

```
hostname ECV-2610-16
!
interface Ethernet0/0
 ip address 10.52.218.48 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id gk-zone2.test.com ipaddr 10.52.218.46 1718
 h323-gateway voip h323-id gw_2
 h323-gateway voip bind srcaddr 10.52.218.48
!
dial-peer voice 1 voip
 destination-pattern 17..
 session target ras
!
dial-peer voice 2 pots
 destination-pattern 911
 port 1/1/1
 no register e164
!
Gateway
!
end
```

008G_504

 CVOICE v4.2—6-26

You can use other interface subcommands, one of which is illustrated in the configuration for both gateways. This command performs the following function:

- **h323-gateway voip tech-prefix 1#:** Registers a technology prefix
- **h323-gateway voip bind srcaddr 10.52.218.48:** Sets the source address in H.323 packets

A technology prefix advises the gatekeeper that this gateway can handle type 1# destinations. For routing purposes, a technology prefix may be assigned to a multimedia type, such as video. By registering type 1# support, the gateway supports the video applications.

In the dial peer, the **no register e164** subcommand causes the gateway not to register the destination pattern when communicating with the gatekeeper. When the dial peer does not register its prefix, the dial peer requires an alternative mechanism for the gatekeeper to acquire this information.

Configuring H.323 Gatekeepers

This topic illustrates the gatekeeper configuration for a two-zone, two-gatekeeper scenario.

Configuring the Gatekeepers

Cisco.com

Gatekeeper 1

```
hostname ECV-2610-15
!
interface Ethernet0/0
 ip address 10.52.218.47 255.255.255.0
!
Gatekeeper
 zone local gk-zone1.test.com test.com 10.52.218.47
 zone remote gk-zone2.test.com test.com 10.52.218.46 1719
 zone prefix gk-zone2.test.com 16..
 zone prefix gk-zone1.test.com 17..
 gw-type-prefix 1#* default-technology
 no shutdown
!
end
```

008G_505

 CVOICE v4.2—6-27

The gatekeeper application is enabled with the **gatekeeper** command.

For this example, the gateways are configured to withhold their E.164 addresses, so the gatekeepers must define the addresses locally. This is done with the **zone prefix** command. In the example, each gatekeeper has two **zone prefix** commands, the first pointing to the other gatekeeper and the second pointing to the local zone (meaning the prefix is in the local zone). The **zone prefix** command that points to itself is configured with the name of the gateway used to direct traffic to the destination. The address of the gateway is not required, because it is determined automatically when the gateway registers. The commands in the figure perform the following functions:

- **zone local gk-zone1.test.com test.com 10.52.218.47:** Defines the ID of the local gatekeeper
- **zone remote gk-zone2.test.com test.com 10.52.218.46 1719:** Defines the identity and IP address of neighboring gatekeepers

Configuring the Gatekeepers (Cont.)

Cisco.com

Gatekeeper 2

```
hostname ECV-2610-14
!
interface Ethernet0/0
 ip address 10.52.218.46 255.255.255.0
!
Gatekeeper
 zone local gk-zone2.test.com test.com 10.52.218.46
 zone remote gk-zone1.test.com test.com 10.52.218.47 1719
 zone prefix gk-zone2.test.com 16..
 zone prefix gk-zone1.test.com 17..
 gw-type-prefix 1#* default-technology
 no shutdown
!
end
```


CVOICE v4.2—6-28

Because the gateways register their technology prefixes, the gatekeeper does not need to be configured. If a technology prefix is required, the **gw-type-prefix** defines a technology prefix, and can manually update technology prefix knowledge in the gatekeeper. In the example configuration in this figure, the gatekeeper attempts to define a technology prefix as the default with the command **gw-type-prefix 1#* default-technology**. Any unknown destination is assumed to be of the default technology type, and calls are forwarded to any gateway that registered the default technology type.

Monitoring and Troubleshooting

The **show** and **debug** commands are valuable when examining the status of the H.323 components and during troubleshooting. This topic lists many **show** and **debug** commands that are used to provide support for monitoring and troubleshooting H.323.

Example: show Command

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```
Router# show gatekeeper calls
Total number of active calls = 1.
                         GATEKEEPER CALL INFO
                         ====================
LocalCallID                         Age(secs)   BW
12-3339                             94          768(Kbps)
 Endpt(s):Alias        E.164Addr     CallSignalAddr  Port
RASSignalAddr   Port
   src EP:epA                          90.0.0.11       1720
90.0.0.11       1700
   dst EP:epB@zoneB.com
   src PX:pxA                          90.0.0.01       1720
90.0.0.01       24999
   dst PX:pxB                          172.21.139.90   1720
172.21.139.90   24999
```

006G_507

 CVOICE v4.2—6-29

Following are some of the **show** commands used for H.323:

- **show call active voice** [**brief**]: Displays the status, statistics, and parameters for all active voice calls
- **show call history voice** [**last** *n*|**record**|**brief**]: Displays call records from the history buffer
- **show gateway**: Displays the current status of the H.323 gateway configured in the router
- **show gatekeeper calls**: Displays the active calls for which the gatekeeper is responsible (illustrated in the figure)
- **show gatekeeper endpoints**: Lists the registered endpoints with ID and supported prefixes
- **show gatekeeper gw-type-prefix**: Displays the current technology prefix table
- **show gatekeeper status**: Displays the current status of the gatekeeper
- **show gatekeeper zone prefix**: Displays the gateways and their associated E.164 prefixes
- **show gatekeeper zone status**: Displays the status of the connections to gateways in the local zone and the status of the connections to gatekeepers in other zones

Selected debug Commands

The **debug** commands used for H.323 include the following:

- **debug voip ccapi inout:** Shows every interaction with the call control application programming interface (API) on the telephone interface and the VoIP side. Monitoring the **debug voip ccapi inout** command output allows users to follow the progress of a call from the inbound interface or VoIP peer to the outbound side of the call. Because this debug is highly active, use it sparingly in a live network.
- **debug cch323 h225:** Traces the transitions in the H.225.0 state machine during the establishment of the call control channel. The first step in establishing a relationship between any two components is to bring up the call control channel. Monitoring the output of the **debug cch323 h225** command allows users to follow the progress and determine if the channel is established correctly.
- **debug cch323 h245**: Traces the state transitions in the H.245 state machine during the establishment of the H.245 control channel. Monitoring the output of the **debug cch323 h245** command allows users to follow the progress to see if the channel is established correctly.
- **debug cch323 ras**: Traces the state transition in the establishment of the RAS control channel. Monitoring the output of the **debug cch323 ras** command allows users to determine if the channel is established correctly.
- **debug h225 asn1**: Displays an expansion of the ASN.1-encoded H.225.0 messages. When investigating VoIP peer association problems, this **debug** command helps users monitor the activity of the call-signaling channel. Because H.225.0 encapsulates H.245, this is a useful approach for monitoring both H.225.0 and H.245.
- **debug h225 events**: Similar to the ASN.1 version of the command but does not expand the ASN.1. Debugging events usually imposes a lighter load on the router.
- **debug h245 asn1**: Similar to the H.225.0 variant except that it displays only the H.245 messages.
- **debug h245 events**: Similar to the H.225.0 variant except that it displays only the H.245 messages.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

Cisco.com

- H.323 includes recommendations for capabilities exchange, call setup, and call routing.
- H.323 components perform the call signaling and control functions.
- An H.323 environment includes terminals, gateways, gatekeepers, and components for multipoint conferences.
- A basic or Fast Connect H.323 call setup does not require a gatekeeper.
- IP-to-IP gateways facilitate the connection between two VoIP networks.

CVOICE v4.2—6-30

Summary (Cont.)

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- H.323 call flows can be established using gatekeepers.
- H.323 defines three types of multipoint conferences: centralized, decentralized, and ad hoc.
- Multiple gatekeepers can be used to build large VoIP networks.
- H.323 uses HSRP and multiple gateways and gatekeepers to mitigate failures.
- An H.323 proxy server can be placed between components to enhance security or to take advantage of routing features.

CVOICE v4.2—6-31

Summary (Cont.)

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- **To configure a gateway, you must enable the gateway, configure the relationship with the gatekeeper, and configure a dial-peer to use the gatekeeper.**
- **To configure a gatekeeper, you must enable the gatekeeper, define the ID of the local gatekeeper and the ID and IP address of neighboring gatekeepers, and if necessary, define technology prefixes and zone prefixes.**
- **Several** show **and** debug **commands can be used to monitor and troubleshoot H.323.**

 CVOICE v4.2—6-32

References

For additional information, refer to these resources:

- ITU-T Recommendation H.323 (Version 4)
  http://www.itu.int/rec/recommendation.asp?type=items&lang=E&parent=T-REC-H.323-200011-S
- *Cisco IOS Voice, Video, and Fax Configuration Guide*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_configuration_guide_book09186a0080080ada.html
- *Cisco IOS Voice, Video, and Fax Command Reference*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_command_reference_book09186a0080080c8b.html

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 6-1: VoIP with H.323

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which two tasks are performed by the RAS signaling function of H.225.0? (Choose two.)

A) performs bandwidth changes

B) transports audio messages between endpoints

C) performs disengage procedures between endpoints and a gatekeeper

D) allows endpoints to create connections between call agents

E) defines call setup procedures based on ISDN call setup

Q2) Match the H.245 control function with its description.

A) Logical channel signaling

B) Capabilities exchange

C) Master/slave determination

D) Mode request

_____ 1. opens and closes the channel that carries the media stream

_____ 2. requests a change in capability of the media stream

_____ 3. negotiates audio, video, and codec capability between the endpoints

_____ 4. is used to resolve conflicts during the call

Q3) Which H.323 component can be colocated with another H.323 component?

A) H.323 terminal

B) H.323 gateway

C) H.323 gatekeeper

D) multipoint control unit

Q4) What are the functions of an H.323 gateway?

A) converts an alias address to an IP address

B) responds to bandwidth requests and modifications

C) transmits and receives G.711 PCM-encoded voice

D) performs translation between audio, video, and data formats

E) receives and processes multiple streams of multimedia input

Q5) In H.323 call establishment, which channel do endpoints use to communicate with the gatekeeper?

A) B channel

B) RAS channel

C) forward channel

D) in-band control channel

Q6) Which **ras** message does a gatekeeper use to determine the status of an endpoint?

A) ARQ

B) IRQ

C) LRQ

D) RRQ

E) URQ

Q7) Put in the correct order the steps involved in H.323 basic call setup without a gatekeeper.

_____ 1. Call setup procedures based on Q.931 create a call-signaling channel between the endpoints.

_____ 2. The H.245 control function negotiates capabilities and exchanges logical channel descriptions.

_____ 3. The gateway determines the IP address of the destination gateway internally.

_____ 4. The logical channel descriptions open RTP sessions.

_____ 5. The endpoints open another channel for the H.245 control function.

_____ 6. The originating gateway initiates an H.225.0 session with the destination gateway on registered TCP port 1720.

_____ 7. The endpoints exchange multimedia over the RTP sessions.

Q8) How does the abbreviated call-setup procedure in version 2 of Recommendation H.323 provide fast setup?

A) The gateway knows a DNS-resolvable domain name for the destination.

B) Endpoints use a separate channel for H.245 control functions to speed up signaling.

C) Capability exchange and logical channel assignments are completed in one round trip.

D) Endpoints and gateways use the same call control model so that no translation is required.

Q9) In gatekeeper-routed call signaling, what is the role of the gatekeeper?

A) It establishes an H.245 control channel with both endpoints.

B) It performs call setup, call function, and call control functions.

C) It represents the other endpoint for call signaling only.

D) It passes on the originating endpoint's request to the terminating endpoint.

Q10) What information does the gatekeeper include in the ACF message?

A) the IP address of the remote endpoint

B) the admission request from the originating endpoint

C) the call setup request from the gateway

D) network statistics

Q11) How are audio and video streams managed in hybrid multipoint conferences?

A) the audio and video streams use separate control channels

B) one stream relies on multicast and the other stream uses the MP

C) the audio, video, and data are mixed and switched by the MP

D) the audio, video, and data streams are multicast to all conference participants

Q12) In which type of multipoint conference can two endpoints convert to a point-to-point conference?

A) centralized

B) distributed

C) ad hoc

D) hybrid

Q13) Which two gatekeeper services contribute to scalability of a network? (Choose two.)

A) address resolution

B) authentication

C) call control

D) call routing

E) call signaling

F) registration

Q14) Which gatekeeper in a multiple gatekeeper call flow allows the remote endpoint to accept the incoming call?

A) the gatekeeper that the originating endpoint is associated with

B) the gatekeeper that the remote endpoint is associated with

C) the gatekeeper that first received the admission request

D) the gatekeeper that is available to set up the call

Q15) In H.323, which survivability strategy allows two gatekeepers to share an IP address?

A) HSRP

B) multiple gateways configured for the same prefix

C) multiple gatekeepers configured for the same prefix

D) multiple gatekeepers with gatekeeper discovery

Q16) What is the problem with having multiple gatekeepers in a network?

A) Only one of the gatekeepers can be active at any given time.

B) Endpoints are configured to find only one of several working gatekeepers in the network.

C) Network components are hard to configure.

D) The H.323 zones may overlap.

Q17) What is the role of an H.323 proxy server?

A) provides an alternate path when the direct path is not the most appropriate

B) performs MC services for a multipoint conference

C) substitutes for a gatekeeper if that gatekeeper goes down

D) provides the control channel for audio and video data streams

Q18) When H.323 proxy servers are being used on a network, which situation would require three sessions to set up a call?

A) when authentication is required

B) when the direct path has poor throughput

C) when the proxy server also represents the terminating endpoint

D) when zones are configured as inaccessible on the gatekeeper

Q19) Which Cisco application supports H.323 gatekeepers?

A) Cisco IP Phone

B) Cisco CallManager

C) Cisco Secure Integrated Software

D) Cisco voice-enabled switches

Q20) Which H.323 component is supported by Cisco SC2200 Signaling Controllers?

A) terminals

B) gateways

C) gatekeepers

D) MCU

Q21) What is the function of the **session target ras** subcommand in H.323 gateway configuration?

A) configures a dial peer to use the gatekeeper

B) identifies the ID of the gatekeeper

C) registers the destination gatekeeper

D) configures the remote gatekeeper

Q22) When you are configuring an H.323 gateway, which three tasks do you need to do to configure the relationship with the gatekeeper? (Choose three.)

A) enable the gateway

B) tell the router which interface should be enabled for H.323 packet processing

C) configure a dial peer to use the gateway

D) identify the gatekeeper ID

E) configure the gateway ID

F) tell the gateway not to register the destination pattern

Q23) Which command tells the gatekeepers to define addresses locally?

A) **zone prefix**

B) **zone local**

C) **zone remote**

D) **zone default**

Q24) This command is configured on a gatekeeper:

zone remote gk-zone2.test.com test.com 10.52.218.46 1719

What is the function of this command?

A) defines a technology prefix

B) defines the identity and IP address of neighboring gatekeepers

C) defines the identity and IP address of the local gatekeeper

D) defines the IP address of the gateway to which default calls will be forwarded

Q25) Which **show** command is used to list the registered endpoints with ID and supported prefixes?

A) **show gatekeeper gw-type-prefix**

B) **show gatekeeper zone prefix**

C) **show gatekeeper endpoints**

D) **show gatekeeper status**

Q26) Which **debug** command should you use to monitor the activity of the H.225.0 and H.245 call signaling channels on a busy network?

A) **debug h225 asn1**

B) **debug h225 events**

C) **debug h245 asn1**

D) **debug h245 events**

Quiz Answer Key

Q1) A, C

Relates to: H.323 and Associated Recommendations

Q2) 1-A, 2-D, 3-B, 4-C

Relates to: H.323 and Associated Recommendations

Q3) C

Relates to: Functional Components of H.323

Q4) D

Relates to: Functional Components of H.323

Q5) B

Relates to: H.323 Call Establishment and Maintenance

Q6) B

Relates to: H.323 Call Establishment and Maintenance

Q7)

Step 1 The originating gateway initiates an H.225.0 session with the destination gateway on registered TCP port 1720.

Step 2 The gateway determines the IP address of the destination gateway internally.

Step 3 Call setup procedures based on Q.931 create a call-signaling channel between the endpoints.

Step 4 The endpoints open another channel for the H.245 control function.

Step 5 The H.245 control function negotiates capabilities and exchanges logical channel descriptions.

Step 6 The logical channel descriptions open RTP sessions.

Step 7 The endpoints exchange multimedia over the RTP sessions.

Relates to: Call Flows Without a Gatekeeper

Q8) C

Relates to: Call Flows Without a Gatekeeper

Q9) C

Relates to: Call Flows with a Gatekeeper

Q10) A

Relates to: Call Flows with a Gatekeeper

Q11) B

Relates to: Multipoint Conferences

Q12) C

Relates to: Multipoint Conferences

Q13) A, F

Relates to: Call Flows with Multiple Gatekeepers

Q14) B

Relates to: Call Flows with Multiple Gatekeepers

Q15) A

Relates to: Survivability Strategies

Q16) B

Relates to: Survivability Strategies

Q17) A

Relates to: H.323 Proxy Server

Q18) C

Relates to: H.323 Proxy Server

Q19) B

Relates to: Cisco Implementation of H.323

Q20) B

Relates to: Cisco Implementation of H.323

Q21) A

Relates to: Configuring H.323 Gateways

Q22) B, D, E

Relates to: Configuring H.323 Gateways

Q23) A

Relates to: Configuring H.323 Gatekeepers

Q24) B

Relates to: Configuring H.323 Gatekeepers

Q25) C

Relates to: Monitoring and Troubleshooting

Q26) A

Relates to: Monitoring and Troubleshooting

Configuring SIP

Overview

This lesson describes how to configure the session initiation protocol (SIP) and explores the features and functions of the SIP environment, including its components, how these components interact, and how to accommodate scalability and survivability.

Relevance

An understanding of the features and functions of SIP components, and the relationships the components establish with each other, is important in implementing a scalable, resilient, and secure SIP environment.

Objectives

Upon completing this lesson, you will be able to configure, monitor, and troubleshoot SIP on a Cisco router. This includes being able to meet these objectives:

- Describe three IETF standards that help SIP in the establishment, maintenance, and termination of multimedia sessions
- List the types of user agents and servers that are used by SIP and describe their functions
- List six examples of SIP request and response messages
- Identify three types of SIP addresses and the servers that are involved in address registration and resolution
- Describe three SIP call setup procedures and list their advantages and disadvantages
- Illustrate two strategies that are used by SIP to provide fault tolerance
- List SIP gateway and network server devices that are supported by Cisco Systems
- Use the **sip-ua** command with subcommands to configure SIP on a Cisco router
- Use **show** and **debug** commands to monitor and troubleshoot SIP

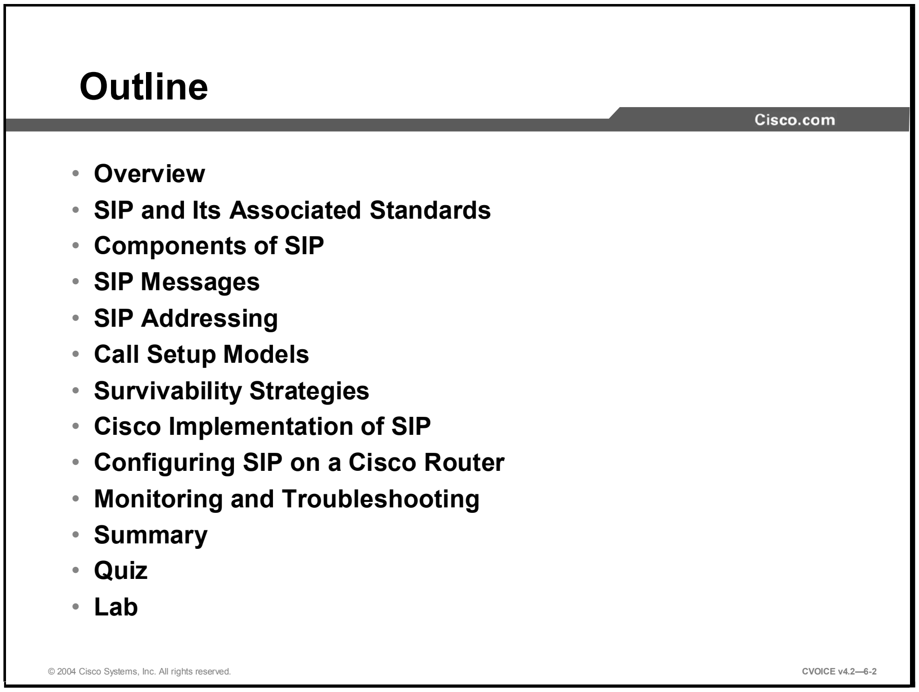
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- An understanding of the objectives and principles of signaling and call control in the context of VoIP

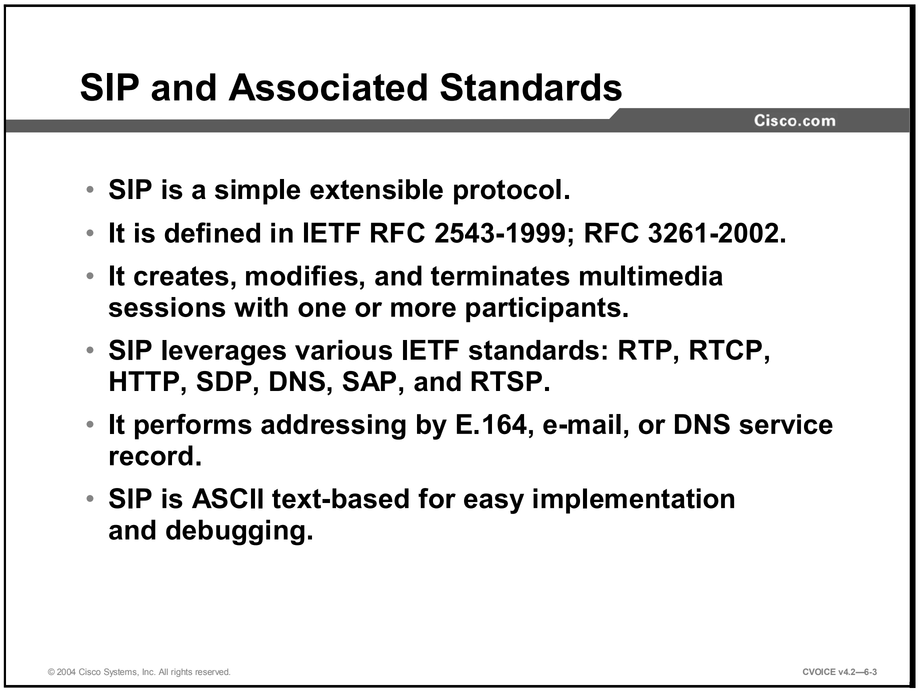
Outline

The outline lists the topics included in this lesson.



SIP and Its Associated Standards

SIP provides another framework for establishing and maintaining VoIP calls. This topic describes SIP and its standards.



SIP is a signaling and control protocol for the establishment, maintenance, and termination of multimedia sessions with one or more participants. SIP multimedia sessions include Internet telephone calls, multimedia conferences, and multimedia distribution. Session communications may be based on multicast, unicast, or both.

SIP operates on the principle of session invitations. Through invitations, SIP initiates sessions or invites participants into established sessions. Descriptions of these sessions are advertised by any one of several means, including the SAP defined in RFC 2974, which incorporates a session description according to the Session Description Protocol (SDP) defined in RFC 2327.

SIP uses other IETF protocols to define other aspects of VoIP and multimedia sessions; for example, URLs for addressing, DNS for service location, and Telephony Routing over IP (TRIP) for call routing.

SIP supports personal mobility and other Intelligent Network (IN) telephony subscriber services through name mapping and redirection services. Personal mobility allows a potential participant in a session to be identified by a unique personal number or name.

IN provides carriers with the ability to rapidly deploy new user services on platforms that are external to the switching fabric. Access to the external platforms is by way of an independent vendor and standard user interface. Calling-card services, 800 services, and local number portability are just three of these services.

Multimedia sessions are established and terminated by the following services:

- **User location services:** Locate an end system
- **User capabilities services:** Select the media type and parameters
- **User availability services:** Determine the availability and desire for a party to participate
- **Call setup services:** Establish a session relationship between parties and manage call progress
- **Call handling services:** Transfer and terminate calls

Although the IETF has made great progress in defining extensions that allow SIP to work with legacy voice networks, the primary motivation behind the protocol is to create an environment that supports next-generation communication models that use the Internet and Internet applications.

SIP is described in IETF RFC 3261 (June 2002), which renders obsolete RFC 2543 (March 1999).

Example: Cisco SIP Support

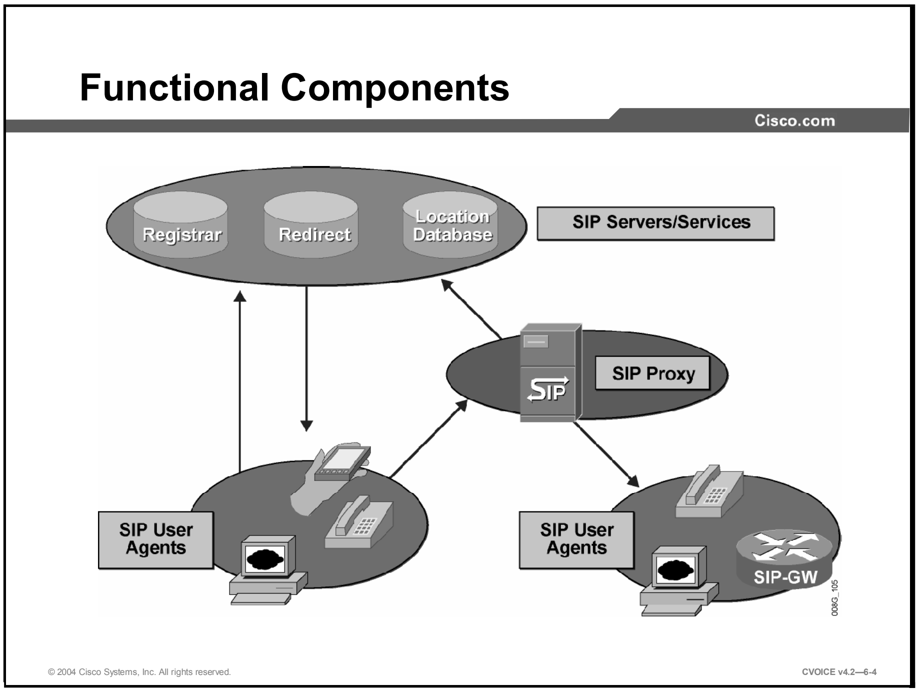
The Cisco SIP-enabled product portfolio encompasses all components of a SIP network infrastructure, from IP Phones and access devices to call control and PSTN interworking. The first Cisco SIP products were deployed with live traffic several years ago. All of the following products are deployed in live networks spanning a variety of applications and continents.

- **Cisco IP Phones:** The Cisco IP Phone series, including the Cisco IP Phone 7960 and the Cisco IP Phone 7940, support SIP user-agent functionality. These IP Phones deliver functionality such as inline-power support and dual-Ethernet ports, and deliver traditional desktop functionality such as call hold, transfer, conferencing, Caller ID, call waiting, and a lighted message-waiting indicator.
- **Cisco ATA 186 Analog Telephone Adaptor:** The Cisco ATA 186 supports SIP user-agent functionality. With two FXS ports and a single Ethernet port, the ATA 186 provides a low-cost means to connect analog phones to a SIP network. It also delivers traditional desktop functionality such as call hold, transfer, conferencing, Caller ID, and lighted call-waiting and message-waiting indicators.
- **Cisco packet voice gateways:** The Cisco 1700 modular access routers that are voice-capable, Cisco 2600 Series multiservice platforms, Cisco 3700 Series multiservice platforms, Cisco AS5000 universal gateways, and Cisco 7200 Series voice gateways all support SIP user-agent functionality. They provide a means of connecting SIP networks to traditional TDM networks via T1, E1, DS3, channel associated signaling (CAS), PRI/BRI, R2 signaling, FXS, FXO, or ear and mouth (E&M) interfaces. Cisco packet voice gateways are used to build the largest packet telephony networks in the world.
- **Cisco SIP Proxy Server:** The Cisco SIP Proxy Server provides the functionality of a SIP proxy, SIP redirect, SIP registrar, and SIP location services server. The Cisco SIP Proxy Server provides the foundation for call routing within SIP networks; it can interwork with traditional SIP location services such as DNS or ENUM, with feature servers via a SIP redirect message, and with H.323 location services using standard LRQ messages. The Cisco SIP Proxy Server runs on either Solaris or Linux operating systems.

- **Cisco BTS 10200 Softswitch:** The Cisco BTS 10200 provides softswitch functionality to Class 4 and Class 5 networks, and provides SIP-to-SS7 gateway functionality for American National Standards Institute (ANSI) standardized networks. The BTS 10200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as a Cisco AS5000 Universal Gateway or Cisco MGX 8000 Series Voice Gateway.
- **Cisco PGW 2200 PSTN Gateway:** The Cisco PGW 2200 provides softswitch functionality for Class 4 networks, as well as Internet offload and SIP-to-SS7 gateway functionality for international networks. The PGW 2200 supports ISDN User Part (ISUP) certification in over 130 countries. The PGW 2200 supports SIP user-agent functionality in conjunction with a Cisco packet voice media gateway such as an AS5000 Universal Gateway or MGX 8000 Series Voice Gateway.
- **Cisco Secure PIX Firewall:** The Cisco Secure PIX Firewall is a SIP-aware networking device that provides firewall and NAT functionality. Because it is SIP aware, it is able to dynamically allow SIP signaling to traverse network and addressing boundaries without compromising overall network security. A Cisco Secure PIX Firewall functioning in this capacity is called an application layer gateway (ALG).

Components of SIP

SIP is modeled on the interworking of user agents (UAs) and network servers. This topic describes the functional and physical components of a UA.



SIP is a peer-to-peer protocol. The peers in a session are called UAs. A UA consists of two functional components:

- **User agent client (UAC):** A client application that initiates a SIP request.
- **User agent server (UAS):** A server application that contacts the user when a SIP invitation is received and then returns a response on behalf of the user to the invitation originator.

Typically, a SIP UA can function as a UAC *or* a UAS during a session, but not both in the same session. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request: the initiating UA uses a UAC and the terminating UA uses a UAS.

From an architectural standpoint, the physical components of a SIP network are grouped into the following two categories:

- **User agents:** SIP user agents include the following devices:
  - **IP telephone:** Acts as a UAS or UAC on a session-by-session basis. Software telephones and Cisco Systems SIP IP Phones initiate SIP requests and respond to requests.
  - **Gateway:** Acts as a UAS or UAC and provides call control support. Gateways provide many services, the most common being a translation function between SIP user agents and other terminal types. This function includes translation between transmission formats and between communications procedures. A gateway translates between audio and video signals and performs call setup and clearing on both the IP side and the SCN side.

- **SIP servers:** SIP servers include the following types:
  - **Proxy server:** Intermediate component that receives SIP requests from a client, then forwards the requests on behalf of the client to the next SIP server in the network. The next server can be another proxy server or a UAS. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request transmissions, and security.
  - **Redirect server:** Provides a UA with information about the next server that the UA should contact. The server can be another network server or a UA. The UA redirects the invitation to the server identified by the redirect server.
  - **Registrar server:** Requests from UACs for registration of their current location. Registrar servers are often located near or even colocated with other network servers, most often a location server.
  - **Location server:** An abstraction of a service providing address resolution services to SIP proxy or redirect servers. A location server embodies mechanisms to resolve addresses. These mechanisms can include a database of registrations or access to commonly used resolution tools such as finger, **rwhois**, Lightweight Directory Access Protocol (LDAP), or operating-system-dependent mechanisms. A registrar server can be modeled as one subcomponent of a location server; the registrar server is partly responsible for populating a database associated with the location server.

Note Except for the REGISTER mode request, communication between SIP components and a location server is not standardized.

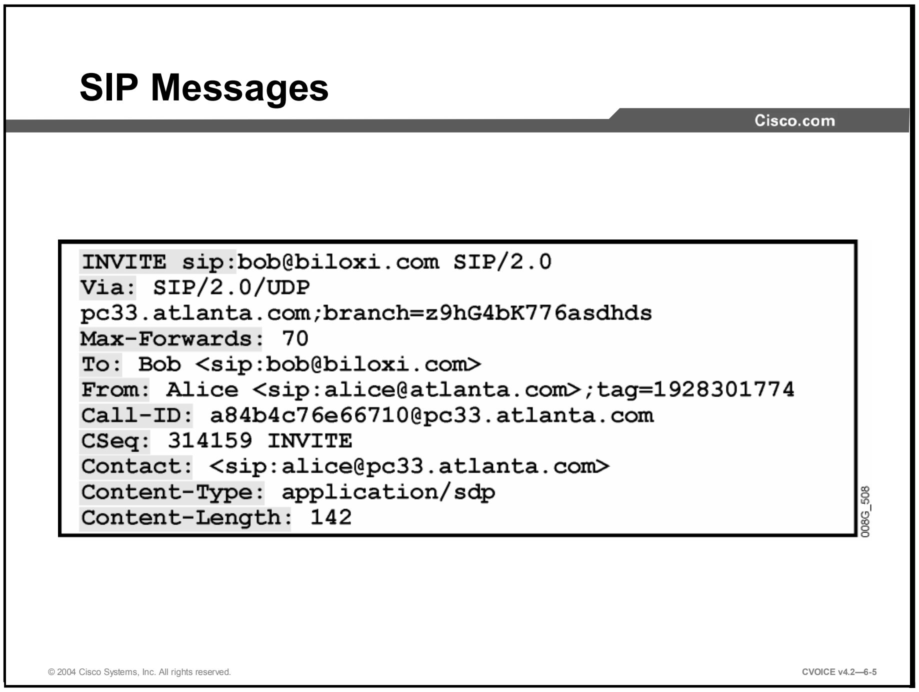
Example: SIP Applications

Leaders in the communications industry are developing new products and services that rely on SIP, and they are offering attractive new communications services to their customers. Microsoft recently added support for SIP clients in core product offerings—Windows XP and Windows Messenger—a step that will proliferate SIP clients on personal computers worldwide. SIP is gaining momentum in every market, and carriers that have deployed SIP in their networks include interexchange carriers such as Worldcom and Genuity, telephony application service providers and communications service providers such as TalkingNets and TellMe, alternate carriers such as Vonage, Internet telephony service providers such as deltathree, and advanced service providers such as Microsoft—all Cisco customers.

Cisco is enabling the advance of new communications services with a complete SIP-enabled portfolio, including proxy servers, packet voice gateways, call control and signaling, IP Phones, and firewalls. These products are available today. Only Cisco is dedicated to providing ubiquitous and seamless protocol interoperability in its packet-telephony solutions. Cisco solutions support a variety of call control and standard protocols, including H.323, MGCP, and SIP, which can coexist in the same customer network.

SIP Messages

Communication between SIP components uses a request and response message model. This topic describes the types, use, and structure of these messages.



SIP communication involves the following two messages:

- **Request from a client to a server:** Consists of a request line, header lines, and a message body
- **Response from a server to a client:** Consists of a status line, header lines, and a message body

All SIP messages are text-based and modeled on RFC 822, *Standard for ARPA Internet Text Messages*, and RFC 2068, *Hypertext Transfer Protocol — HTTP/1.1*.

SIP defines four types of headers: a general header, an entity header, a request header, and a response header. The first two appear on both message types. The latter two are specific to request and response, respectively.

SIP Request Messages

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- **INVITE—Indicates that a user or service is being invited to participate in a call session**
- **ACK—Confirms that a client has received a final response to an INVITE request**
- **BYE—Terminates an existing call; can be sent by either user agent**
- **CANCEL—Cancels pending searches; does not terminate calls that have been accepted**
- **OPTIONS—Queries the capabilities of servers**
- **REGISTER—Registers the user agent with the registrar server of a domain**

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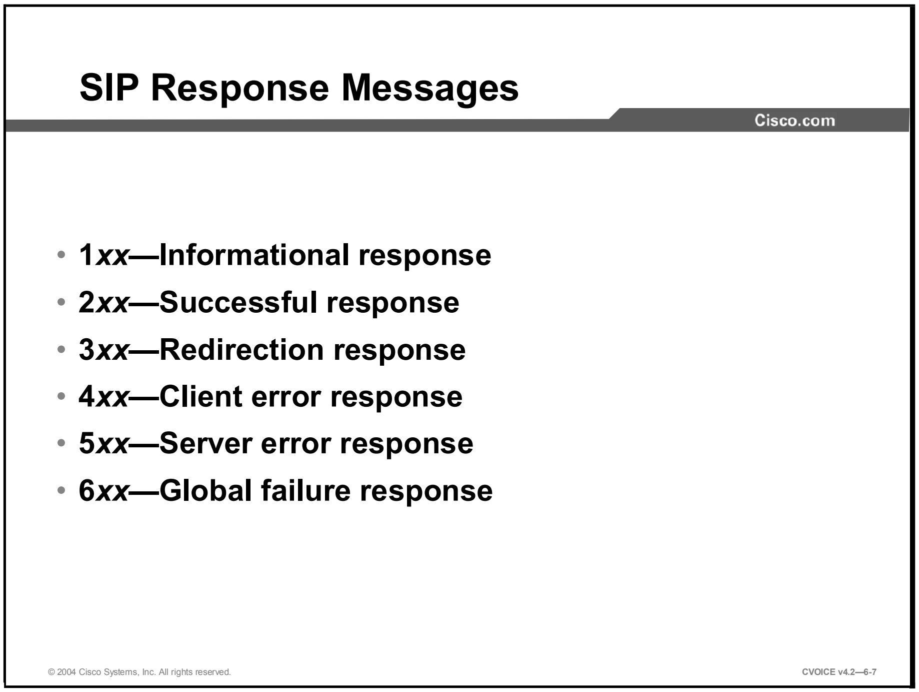
In the request line, SIP uses a “method” to indicate the action to be taken by the responding component (usually a server).

The following request methods indicate the action that the responding component should take:

- **INVITE:** A client originates the INVITE method to indicate that the server is invited to participate in a session. An invitation includes a description of the session parameters.
- **ACK:** A client originates the ACK method to indicate that the client has received a response to its earlier invitation.
- **BYE:** A client or server originates the BYE method to initiate call termination.
- **CANCEL:** A client or server originates the CANCEL method to interrupt any request currently in progress. CANCEL is *not* used to terminate active sessions.
- **OPTIONS:** A client uses the OPTIONS method to solicit capabilities information from a server. This method is used to confirm cached information about a UA or to check the ability of a UA to accept an incoming call.
- **REGISTER:** A UA uses the REGISTER method to provide information to a network server. Registrations have a finite life and must be renewed periodically. This prevents the use of stale information when a UA moves.



SIP response messages are sent in response to a request and indicate the outcome of request interpretation and execution. Responses take one of three basic positions: success, failure, or provisional. A status code reflects the outcome of the request.

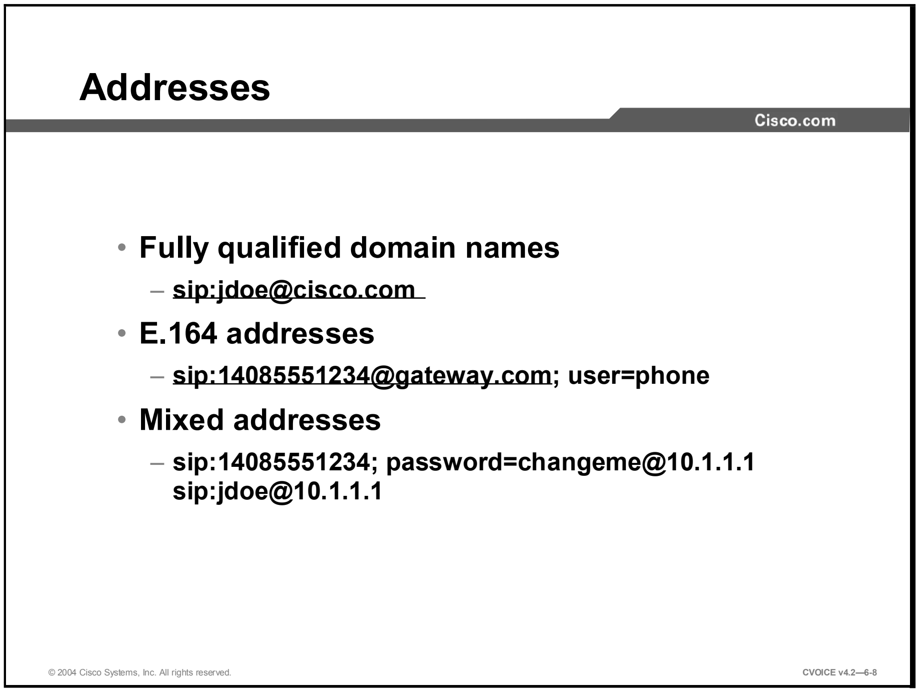
Status Codes

The following response messages indicate the status of a request:

- **1*xx*—Informational:** Provisional response. Indicates that the request is still being processed.
- **2*xx*—Successful:** Indicates that the requested action is complete and successful.
- **3*xx*—Redirection:** Indicates that the requestor requires further action; for example, a redirect server responds with "moved" to advise the client to redirect its invitation.
- **4*xx*—Client error:** Fatal response. Indicates that the client request is flawed or impossible to complete.
- **5*xx*—Server error:** Fatal response. Indicates that the request is valid but the server failed to complete it.
- **6*xx*—Global failure:** Fatal response. Indicates that the request cannot be fulfilled by any server.

SIP Addressing

To obtain the IP address of a SIP UAS or a network server, a UAC performs address resolution of a user identifier. This topic describes address formats, address registration, and address resolution.



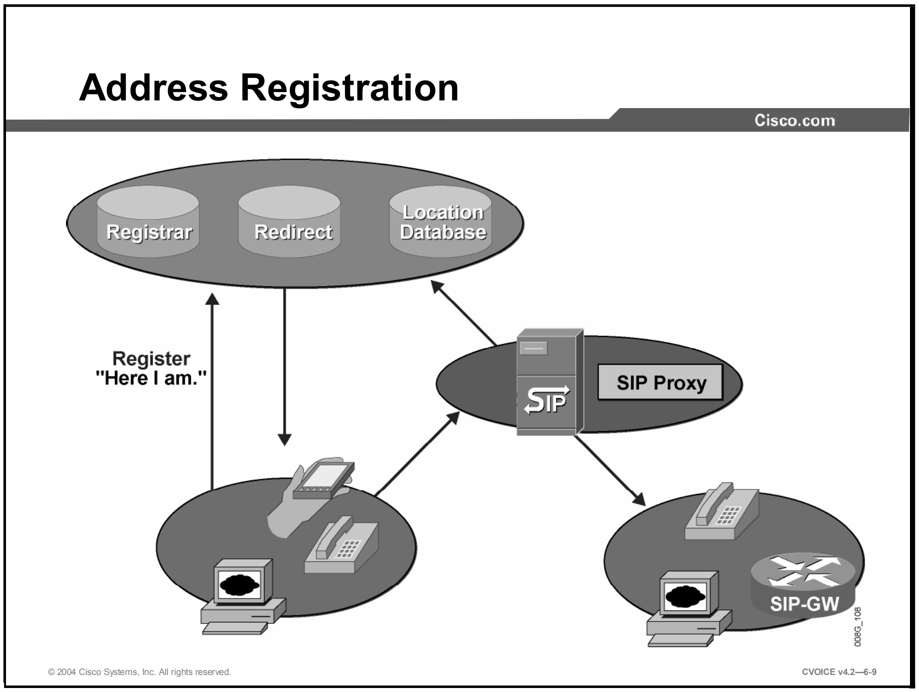
An address in SIP is defined in the syntax for a URL with **sip:** or **sips:** (for secure SIP connections) as the URL type. SIP URLs are used in SIP messages to identify the originator, the current destination, the final recipient, and any contact party. When two UAs communicate directly with each other, the current destination and final recipient URLs are the same. However, the current destination and the final recipient are different if a proxy or redirect server is used.

An address consists of an optional user ID, a host description, and optional parameters to qualify the address more precisely. The host description may be a domain name or an IP address. A password is associated with the user ID, and a port number is associated with the host description.

Example: SIP Addressing Variants

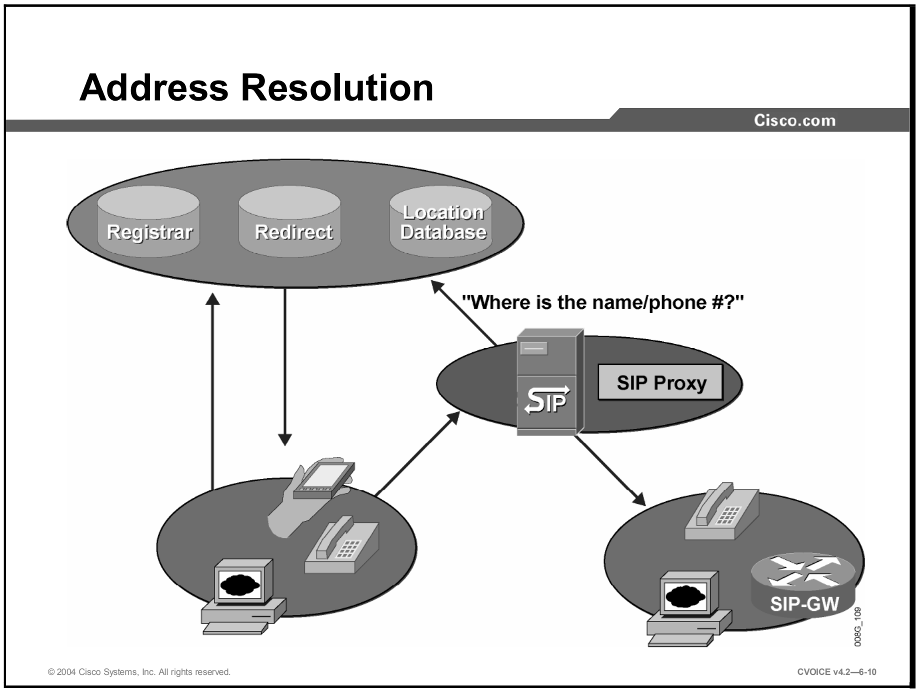
The figure provides examples of SIP addresses.

In the example **sip:14085551234@gateway.com; user=phone**, the **user=phone** parameter is required to indicate that the user part of the address is a telephone number. Without the **user=phone** parameter, the user ID is taken literally as a numeric string. The **14085559876** in the URL **sip:14085559876@10.1.1.1** is an example of a numeric user ID. In the same example, the password **changeme** is defined for the user.



A SIP address is acquired in several ways: by interacting with a user, by caching information from an earlier session, or by interacting with a network server. For a network server to assist, it must recognize the endpoints in the network. This knowledge is abstracted to reside in a location server and is dynamically acquired by its registrar server.

To contribute to this dynamic knowledge, an endpoint registers its user addresses with a registrar server. The figure illustrates a REGISTER mode request to a registrar server.

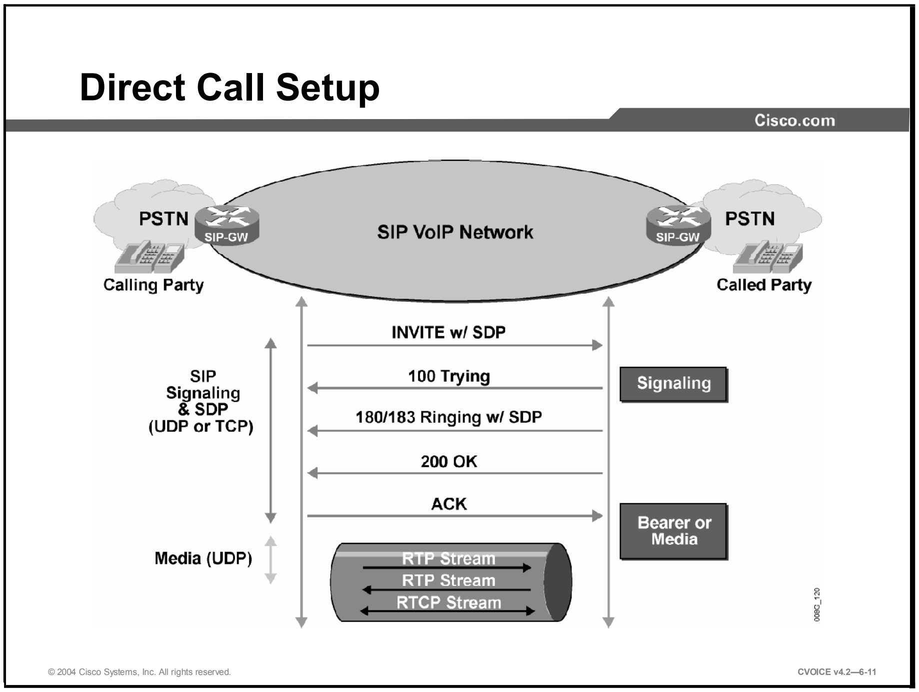


To resolve an address, a UA uses a variety of internal mechanisms such as a local host table, DNS lookup, finger, **rwhois**, or LDAP, or it leaves that responsibility to a network server. A network server uses any of the tools available to a UA or interacts through a nonstandard interface with a location server.

The figure illustrates a SIP proxy server resolving the address by using the services of a location server.

Call Setup Models

If a UAC recognizes the destination UAS, the client communicates directly with the server. In situations in which the client is unable to establish a direct relationship, the client solicits the assistance of a network server. This topic illustrates three interworking models for call setup: direct, using a proxy server, and using a redirect server.



When a UA recognizes the address of a terminating endpoint from cached information, or has the capacity to resolve it by some internal mechanism, the UAC may initiate direct (UAC-to-UAS) call setup procedures.

Direct setup is the fastest and most efficient of the call setup procedures. However, direct setup has some disadvantages. It relies on cached information or internal mechanisms to resolve addresses, which can become outdated if the destination is mobile. In addition, if the UA must keep information on a large number of destinations, management of the data can become prohibitive. This makes the direct method nonscalable.

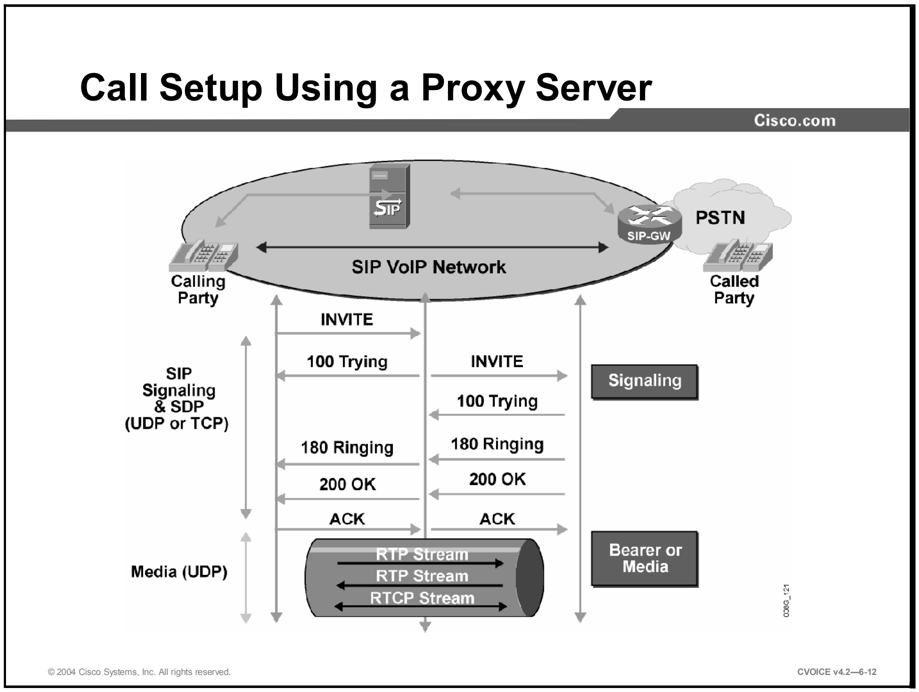
Direct call setup proceeds as follows:

1. The originating UAC sends an invitation (INVITE) to the UAS of the recipient. The message includes an endpoint description of the UAC and SDP.

2. If the UAS of the recipient determines that the call parameters are acceptable, it responds positively to the originator UAC.

3. The originating UAC issues an ACK.

At this point, the UAC and UAS have all the information that is required to establish RTP sessions between them.



The proxy server procedure is transparent to a UA. The proxy server intercepts and forwards an invitation to the destination UA on behalf of the originator.

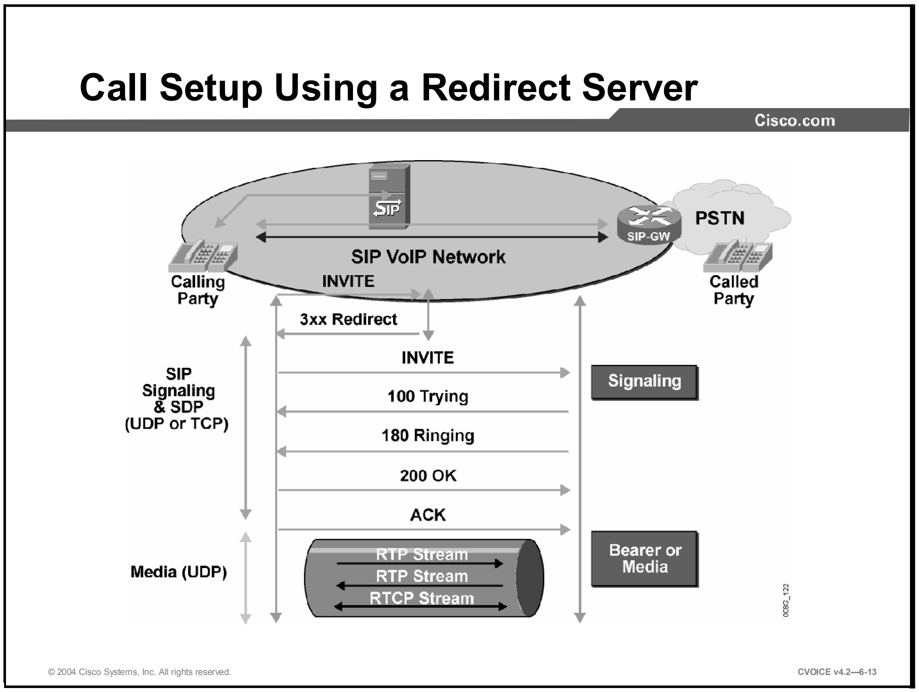
A proxy server responds to the issues of the direct method by centralizing control and management of call setup and providing a more dynamic and up-to-date address resolution capability. The benefit to the UA is that it does not need to learn the coordinates of the destination UA, yet can still communicate with the destination UA. The disadvantages of this method are that using a proxy server requires more messaging and creates a dependency on the proxy server. If the proxy server fails, the UA is incapable of establishing its own sessions.

Note Although the proxy server acts on behalf of a UA for call setup, the UAs establish RTP sessions directly with each other.

When a proxy server is used, the call setup procedure is as follows:

1. The originating UAC sends an invitation (INVITE) to the proxy server.
2. The proxy server, if required, consults the location server to determine the path to the recipient and its IP address.
3. The proxy server sends the invitation to the UAS of the recipient.
4. If the UAS of the recipient determines that the call parameters are acceptable, it responds positively to the proxy server.
5. The proxy server responds to the originating UAC.
6. The originating UAC issues an ACK.
7. The proxy server forwards the ACK to the recipient UAS.

The UAC and UAS now have all the information required to establish RTP sessions.



A redirect server is programmed to discover a path to the destination. Instead of forwarding the invitation to the destination, the redirect server reports back to a UA with the destination coordinates that the UA should try next.

A redirect server offers many of the advantages of the proxy server. However, the number of messages involved in redirection is fewer than with the proxy server procedure. The UA has a heavier workload because it must initiate the subsequent invitation.

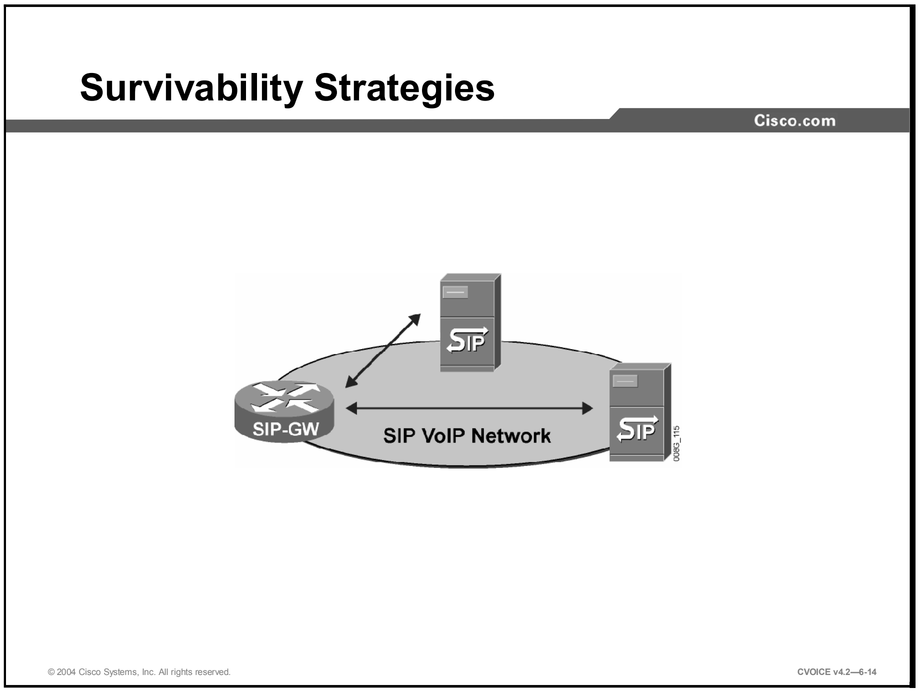
When a redirect server is used, the call setup procedure is as follows:

1. The originating UAC sends an invitation (INVITE) to the redirect server.
2. The redirect server, if required, consults the location server to determine the path to the recipient and its IP address.
3. The redirect server returns a "moved" response to the originating UAC with the IP address obtained from the location server.
4. The originating UAC acknowledges the redirection.
5. The originating UAC sends an invitation to the remote UAS.
6. If the UAS of the recipient determines that the call parameters are acceptable, it responds positively to the UAC.
7. The originating UAC issues an acknowledgment.

The UAC and UAS now have all the information required to establish RTP sessions between them.

Survivability Strategies

Maintaining high availability of a SIP environment requires a design that accommodates the failure of a network server. This topic describes strategies for maintaining VoIP service.



In a SIP environment, the failure of a network server cripples UAs that are dependent on that server. In SIP, the network servers are the proxy server, the redirect server, and the location server.

The most obvious way to preserve access to the critical components is to implement multiple instances of access.

For replication of a proxy or redirect server to be effective, a UA must have the ability to locate an active server dynamically. You can achieve this in any of the following ways:

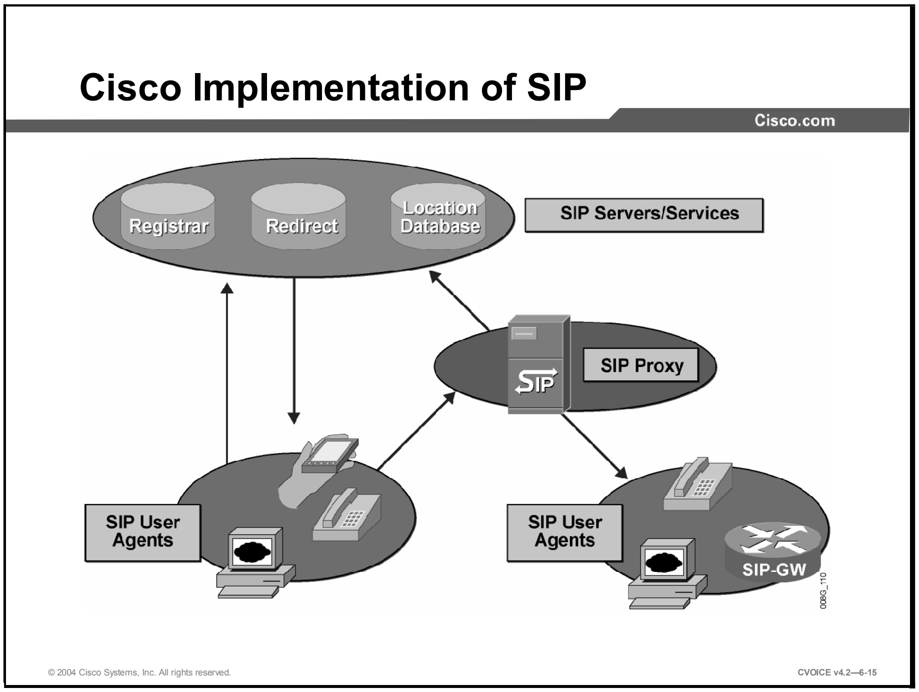
- Preconfigure a UA with the address of at least two of the servers. If access to its first choice fails, it shifts to the second.
- If all servers are configured with the same name, you must configure a UA to look up the name using DNS. The DNS query returns the addresses of all the servers matching the name, and the UA proceeds down the list until it finds one that works.

Example: SIP Survivability

The figure illustrates replication of SIP servers for survivability.

Cisco Implementation of SIP

This topic describes how Cisco implements SIP.



Cisco provides support for the following SIP components:

- **SIP user agents:** Cisco provides support for SIP UA in Cisco IP Phone.

  Cisco implements SIP UA (gateway) support in the following devices:

  — Cisco voice-enabled routers (first available in Cisco IOS Release 12.1)

  — Cisco PGW 2200 PSTN gateways

  — Voice-enabled AS5*xx*0 access servers

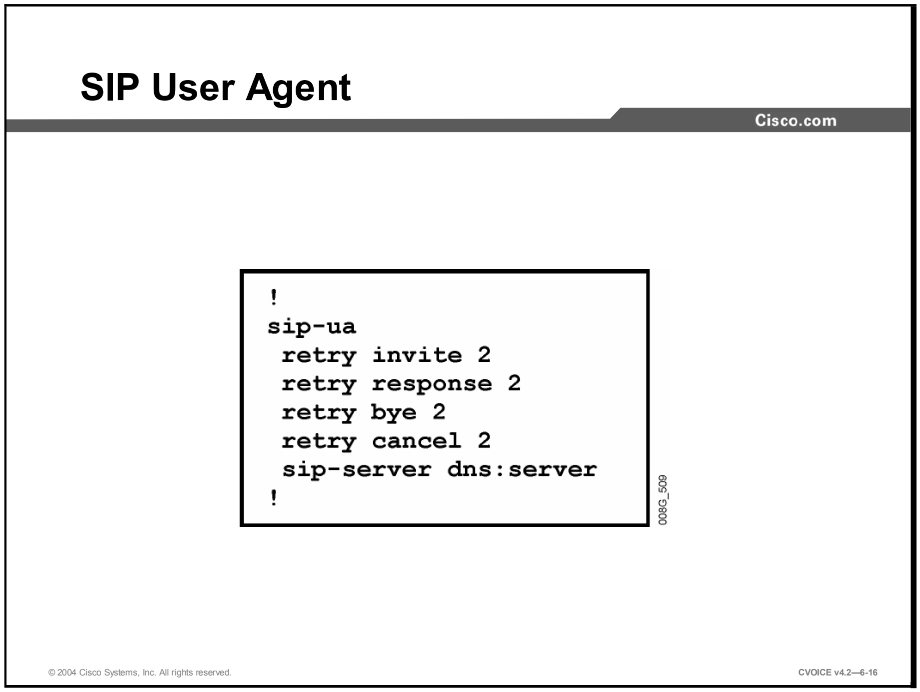
  — Cisco BTS 10200 Softswitch

- **Network servers:** Cisco implements SIP proxy and redirect server support in the Cisco SIP Proxy Server. The server is an application designed for a Linux (Redhat 7.3) or Solaris 8 operating environment.

- **Other support:** Cisco PIX 500 Series firewalls monitor the SIP handshaking to dynamically open conduits for the RTP sessions.

Configuring SIP on a Cisco Router

A SIP configuration consists of two parts: the SIP UA and the VoIP dial peers that select SIP as the session protocol. This topic illustrates and describes the configuration commands that you must use to implement SIP call setup models.



The SIP UA is one part of the SIP configuration. The figure shows an example of a SIP UA configuration.

Example: Configuring a SIP User Agent

The UA is enabled with the **sip-ua** command. Subcommands are optional. The example shows how you can change the value of four retry counters. The configuration also specifies the name of a SIP proxy or redirect server.

SIP Dial Peers

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```
!
dial-peer voice 444 voip
 destination-pattern 2339000
 session protocol sipv2
 session target ipv4:172.18.192.205
 !
dial-peer voice 111 voip
 destination-pattern 111
 session protocol sipv2
 session target sip-server
 !
```

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SIP is selected as the call control protocol from inside a dial peer. SIP is requested by the **session protocol sipv2** dial-peer subcommand. The example illustrates two dial-peer variations.

Example: SIP Dial Peers

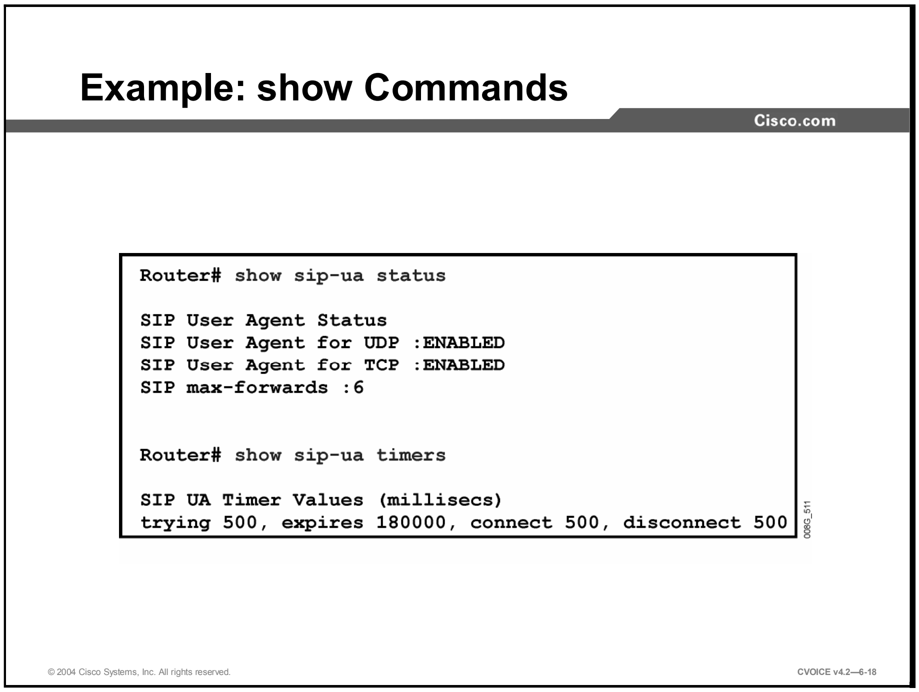
In the example, both dial peers include **session protocol sipv2**, and SIP is used when the destination pattern matches either dial peer. The session target distinguishes one session from the other.

In dial peer 444, the IP address of the server is provided as the session target. The address can be the address of a UA, proxy server, or redirect server.

In dial peer 111, the session target is **sip-server**. When **sip-server** is the target, the IP address of the actual server is taken from the **sip-server** subcommand in the SIP UA configuration. This means that from global configuration mode, the network administrator has entered the **sip-ua** command and the **sip-server dns:server** subcommand. The address represents the location of a proxy server or redirect server. In this example, the name of the SIP server is **server**.

Monitoring and Troubleshooting

This topic lists the **show** and **debug** commands used to provide support for monitoring and troubleshooting SIP.



The following **show** commands are valuable when examining the status of SIP components and troubleshooting:

- **show call active voice** [**brief**]**:** Displays the status, statistics, and parameters for all active voice calls
- **show call history voice** [**last** *n* | **record** | **brief**]**:** Displays call records from the history buffer
- **show sip-ua retry:** Displays the SIP protocol retry counts—high counts should be investigated
- **show sip-ua statistics:** Displays the SIP UA response, traffic, and retry statistics
- **show sip-ua status:** Displays the SIP UA listener status, which should be enabled (shown in the figure)
- **show sip-ua timers:** Displays the current value of the SIP UA timers (shown in the figure)

The following **debug** commands are valuable when examining the status of SIP components and troubleshooting:

- **debug voip ccapi inout:** Shows every interaction with the call control API on both the telephone interface and on the VoIP side. By monitoring the output you can follow the progress of a call from the inbound interface or VoIP peer to the outbound side of the call. This debug is very active; you must use it sparingly in a live network.
- **debug ccsip all:** Enables all **ccsip**-type debugging. This debug is very active; you must use it sparingly in a live network.
- **debug ccsip calls:** Displays all SIP call details as they are updated in the SIP call control block. You must use this debug to monitor call records for suspicious clearing causes.
- **debug ccsip errors:** Traces all errors encountered by the SIP subsystem.
- **debug ccsip events:** Traces events, such as call setups, connections, and disconnections. An **events** version of a debug is often the best place to start, because detailed debugs provide a great deal of useful information.
- **debug ccsip messages:** Shows the headers of SIP messages that are exchanged between a client and a server.
- **debug ccsip states:** Displays the SIP states and state changes for sessions within the SIP subsystem.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- SIP uses IETF protocols, including URL, DNS, and TRIP to define aspects of VoIP and multimedia sessions.
- The two basic components of SIP are user agents and network servers.
- SIP uses a request/response messaging model for communication. All messages are text-based and modeled on the HTTP syntax.
- SIP addresses follow the format and structure of a URL. Network components such as location and registrar servers record addresses and perform address resolution.

 CVOICE v4.2—6-19

Summary (Cont.)

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- Call setup between user agents is possible, but a proxy or redirect server may be used for scalability or to simplify user agent configuration.
- Multiple SIP proxy or redirect servers enhance reliability.
- Cisco supports standalone clients and gateway clients. Support for SIP proxy or redirect services is provided by the Cisco SIP proxy server.
- The sip-ua command can be used to configure SIP on a Cisco router.
- Several show and debug commands help in monitoring and troubleshooting SIP.

 CVOICE v4.2—6-20

References

For additional information, refer to these resources:

- IETF RFC 3261, *SIP: Session Initiation Protocol*
  http://www.faqs.org/rfcs/rfc3261.html
- *Cisco IOS Voice, Video, and Fax Configuration Guide*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_configuration_guide_book09186a0080080ada.html
- *Cisco IOS Voice, Video, and Fax Command Reference*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_command_reference_book09186a0080080c8b.html

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 6-2: Configuring VoIP with SIP

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which IETF protocol does SIP use for call routing?

A) BGP

B) OSPF

C) RIP

D) TRIP

Q2) Which SIP service selects the media type and parameters?

A) user location services

B) user capabilities services

C) user availability services

D) call setup services

E) call handling services

Q3) Which four of the following are SIP servers? (Choose four.)

A) registrar

B) gateway

C) redirect

D) location

E) proxy

Q4) Which SIP server is often colocated with the location server?

A) proxy

B) redirect

C) registrar

D) gateway

Q5) Which SIP method is used to provide information to a network server?

A) INVITE

B) ACK

C) OPTIONS

D) REGISTER

Q6) Which SIP response message is provisional?

A) 1*xx*—Informational

B) 2*xx*—Successful

C) 3*xx*—Redirection

D) 4*xx*—Client error

E) 5*xx*—Server error

F) 6*xx*—Global failure

Q7) How does a SIP UA resolve an address? (Choose three.)

A) uses a local host table

B) uses **rwhois**

C) lets the network server resolve it

D) relies on WINS

Q8) What type of SIP address is represented by **sip:19193631234@gateway.com;user=phone**?

A) fully qualified domain name

B) E.164 address

C) mixed address

D) URL address

Q9) What is a disadvantage of the direct call setup method?

A) It relies on cached information, which may be out of date.

B) It uses more bandwidth because it requires more messaging.

C) It must learn the coordinates of the destination UA.

D) It needs the assistance of a network server.

Q10) Which statement is true regarding call setup using a proxy server?

A) If the proxy server fails, the UA uses RTP to establish its sessions.

B) If the proxy server fails, the UA cannot establish its own sessions.

C) The proxy server sends fewer redirection messages than a redirect server.

D) The UAs establish RTP sessions through the proxy server.

Q11) Which three SIP components need to be replicated to provide fault tolerance? (Choose three.)

A) proxy server

B) redirect server

C) registrar server

D) location server

E) gateway server

Q12) What method can you use to replicate a proxy server?

A) configure two replication servers on the network

B) configure a redirect server to act as a proxy server

C) enable the UA to dynamically locate an active server

D) use HSRP

Q13) Cisco provides support for which three of the following SIP components? (Choose three.)

A) SIP user agent

B) SIP proxy server

C) SIP redirect server

D) SIP location server

Q14) Which operating environments can Cisco SIP proxy server be used with?

A) Linux (Redhat 7.3 or later)

B) Windows NT

C) Solaris 2.8

D) MacOS

Q15) Which command is required to enable SIP on a Cisco router?

A) **sip-ua** interface configuration subcommand

B) **sip-ua** dial-peer configuration subcommand

C) **sip-ua** global configuration command

D) No special command is required. SIP is on by default.

Q16) What does the **session target sip-server** dial-peer subcommand do?

A) It tells the router to use DNS to resolve **sip-server**.

B) It tells the router to use the server identified in the SIP UA configuration.

C) It tells the router to use SIP as the session protocol.

D) This is invalid syntax and an error will be generated.

Q17) Which **show** command displays SIP UA response and retry information?

A) **show sip-ua retry**

B) **show sip-ua statistics**

C) **show call active voice**

D) **show sip-ua status**

Q18) Which **debug** command would you use to trace call setups, connections, and disconnections?

A) **debug voip ccapi inout**

B) **debug ccsip calls**

C) **debug ccsip states**

D) **debug ccsip messages**

E) **debug ccsip events**

Quiz Answer Key

Q1) D

Relates to: SIP and Its Associated Standards

Q2) B

Relates to: SIP and Its Associated Standards

Q3) A, C, D, E

Relates to: Components of SIP

Q4) C

Relates to: Components of SIP

Q5) D

Relates to: SIP Messages

Q6) A

Relates to: SIP Messages

Q7) A, B, C

Relates to: SIP Addressing

Q8) B

Relates to: SIP Addressing

Q9) A

Relates to: Call Setup Models

Q10) B

Relates to: Call Setup Models

Q11) A, B, D

Relates to: Survivability Strategies

Q12) C

Relates to: Survivability Strategies

Q13) A, B, C

Relates to: Cisco Implementation of SIP

Q14) A

Relates to: Cisco Implementation of SIP

Q15) C

Relates to: Configuring SIP on a Cisco Router

Q16) B

Relates to: Configuring SIP on a Cisco Router

Q17) B

Relates to: Monitoring and Troubleshooting

Q18) E

Relates to: Monitoring and Troubleshooting

Configuring MGCP

Overview

The Media Gateway Control Protocol (MGCP) environment is an example of a centralized call control model. This lesson describes how to configure MGCP on a gateway, and the features and functions of the MGCP environment.

Relevance

An understanding of the features and functions of MGCP, its components, and the relationships that the components establish with each other is important to implement a scalable, resilient, and secure MGCP environment.

Objectives

Upon completing this lesson, you will be able to configure, monitor, and troubleshoot MGCP on a Cisco router. This includes being able to meet these objectives:

- Define MGCP and its functions
- List the basic components of MGCP
- Name eight types of MGCP endpoints defined in RFC 2705 and identify their functions
- Name seven types of MGCP gateways defined in RFC 2705 and identify their functions
- Describe the function of a call agent in an MGCP environment
- List the basic concepts of MGCP
- List the steps involved in the process of MGCP call establishment
- Describe the function of MGCP events and signals and give five examples of each
- List eight types of MGCP gateways and the packages that are associated with each gateway
- Explain how digit maps reduce the load on the network during call setup
- List nine MGCP control messages that are used to control and manage endpoints and their connections
- Describe MGCP call setup and control procedures
- Identify two strategies implemented by Cisco Systems to provide high availability in an MGCP environment

- Identify the Cisco devices that implement MGCP
- List the basic steps necessary to implement Cisco CallManager as an MGCP call agent
- Use the **mgcp** command and subcommands to configure an MGCP residential and trunk gateway on a Cisco router
- Use **show** and **debug** commands to monitor and troubleshoot MGCP

Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- An understanding of the objectives and principles of signaling and call control in the context of VoIP

Outline

The outline lists the topics included in this lesson.

Outline

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- Overview
- MGCP and Its Associated Standards
- Basic MGCP Components
- MGCP Endpoints
- MGCP Gateways
- MGCP Call Agents
- Basic MGCP Concepts
- MGCP Calls and Connections
- MGCP Events and Signals
- MGCP Packages

CVOICE v4.2—6-2

Outline (Cont.)

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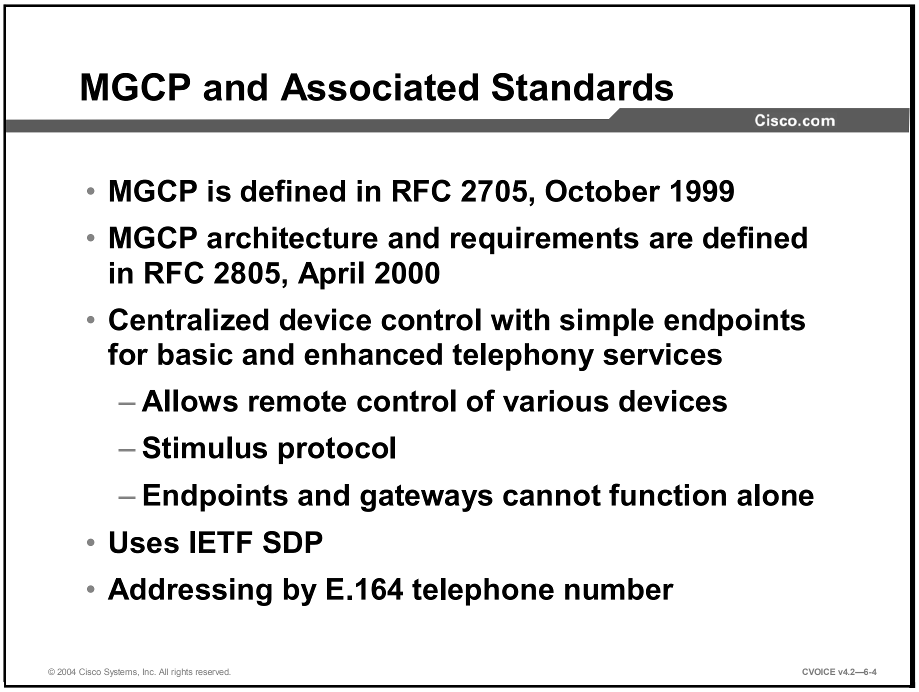
- MGCP Digit Maps
- MGCP Control Commands
- Call Flows
- Survivability Strategies
- Cisco Implementation of MGCP
- Understanding Basics of Cisco CallManager
- Configuring MGCP
- Monitoring and Troubleshooting MGCP
- Summary
- Quiz
- Lab

CVOICE v4.2—6-3

MGCP and Its Associated Standards

MGCP controls telephony gateways from a centralized call agent. This topic describes MGCP and identifies its associated standards.

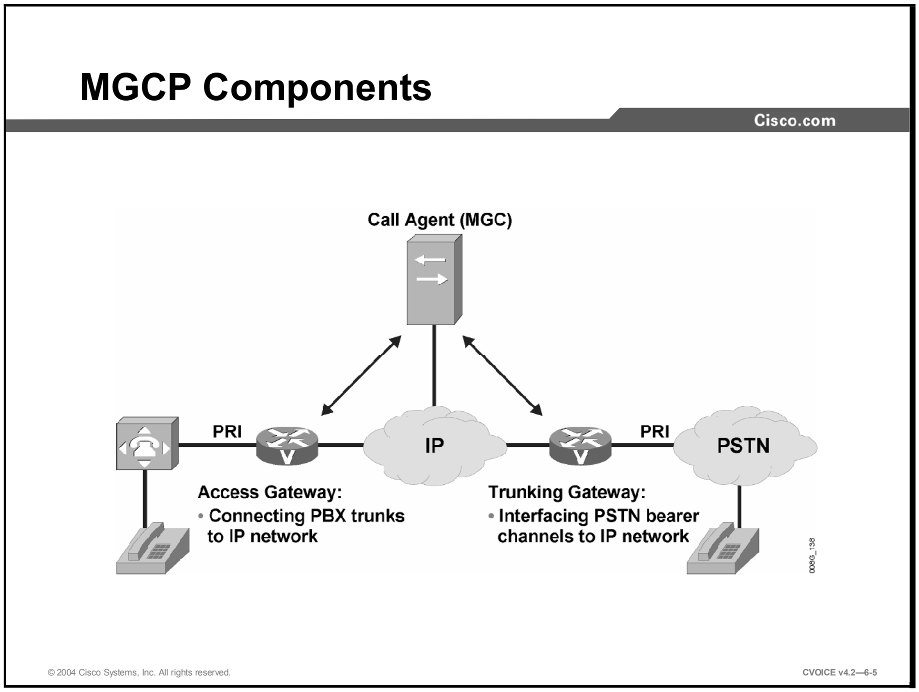


MGCP defines an environment for controlling telephony gateways from a centralized call control component known as a call agent. An MGCP gateway handles the translation of audio between the telephone SCN and the packet-switched network of the Internet. Gateways interact with a call agent that performs signaling and call processing.

IETF RFC 2705 defines MGCP. RFC 2805 defines an architecture for MGCP. These IETF standards describe MGCP as a centralized device control protocol with simple endpoints. The MGCP protocol allows a central control component, or call agent, to remotely control various devices. This protocol is referred to as a stimulus protocol because the endpoints and gateways cannot function alone. MGCP incorporates the IETF SDP to describe the type of session to initiate.

Basic MGCP Components

MGCP defines a number of components and concepts. You must understand the relationships between components and how the components use the concepts to implement a working MGCP environment. This topic describes the basic MGCP components.



The following components are used in an MGCP environment:

- **Endpoints:** Represent the point of interconnection between the packet network and the traditional telephone network
- **Gateways:** Handle the translation of audio between the SCN and the packet network
- **Call agent:** Exercises control over the operation of a gateway

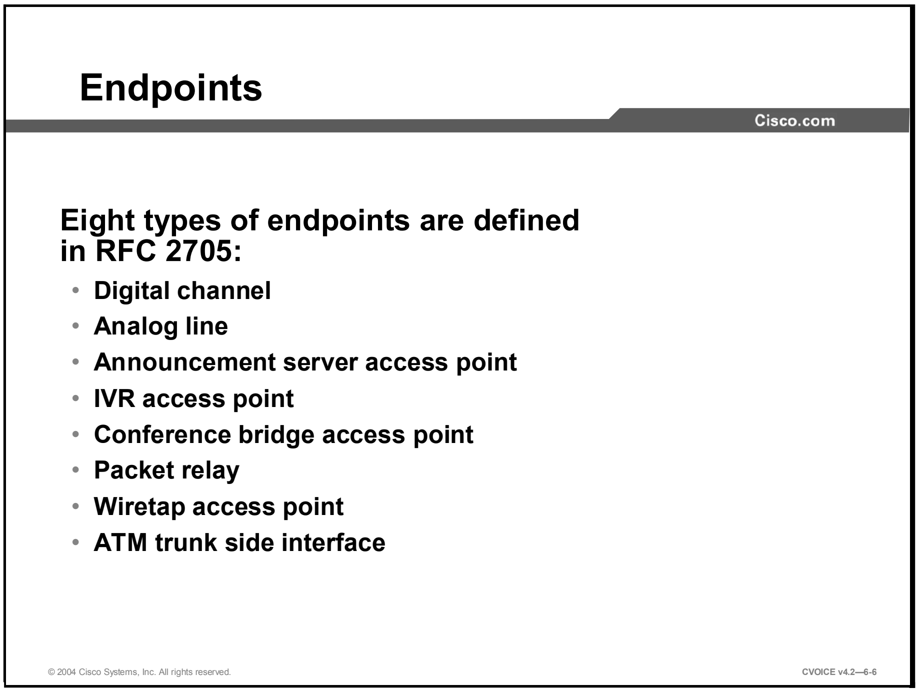
The figure shows an MGCP environment with all three components.

Example: Cisco MGCP Components

Cisco voice gateways can act as MGCP gateways. Cisco CallManager acts as an MGCP call agent.

MGCP Endpoints

This topic lists the standard endpoints and defines the way that identifiers are associated with an endpoint.



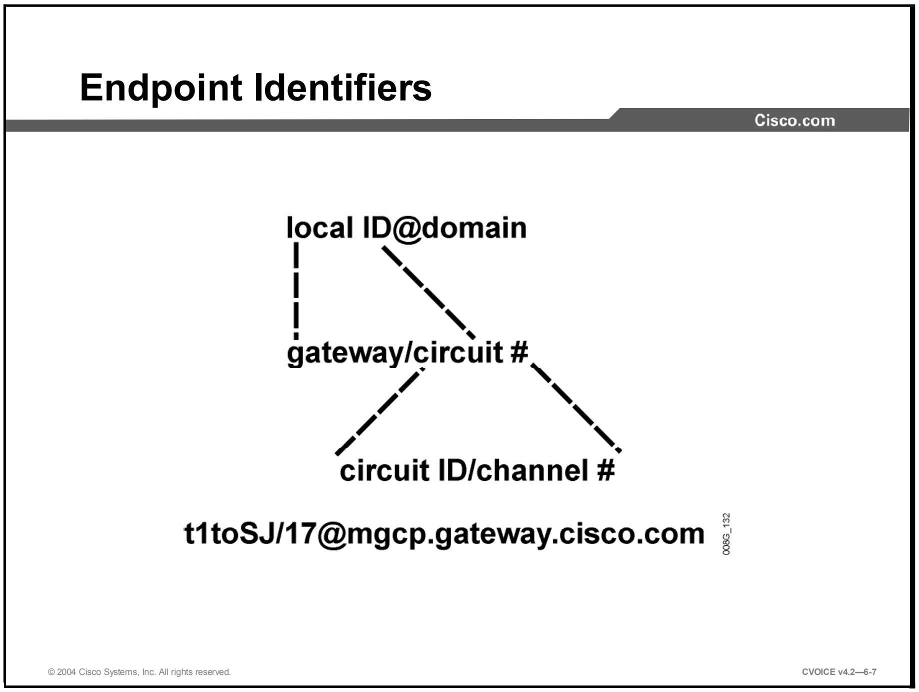
Endpoints represent the point of interconnection between the packet network and the traditional telephone network. Endpoints can be physical, representing an FXS port or a channel in a T1 or E1, or they can be logical, representing an attachment point to an announcement server.

To manage an endpoint, the call agent must recognize the characteristics of an endpoint. To aid in this process, endpoints are categorized into several types. The intent is to configure a call agent to manage a type of endpoint rather than to manage each endpoint individually.

There are several types of endpoints. RFC 2705 defines eight types, as follows:

- **Digital service level zero (DS0):** Represents a single channel (DS0) in the digital hierarchy. A digital channel endpoint supports more than one connection.
- **Analog line:** Represents the client-side interface, such as FXS, or switch-side interface, such as FXO, to the traditional telephone network. An analog line endpoint supports more than one connection.
- **Announcement server access point:** Represents access to an announcement server, for example, to play recorded messages. An announcement server endpoint may have only one connection. Multiple users of the announcement server are modeled to use different endpoints.
- **Interactive voice response (IVR) access point:** Represents access to an IVR service. An IVR endpoint has one connection. Multiple users of the IVR system are modeled to use different endpoints.
- **Conference bridge access point:** Represents access to a specific conference. Each conference is modeled as a distinct endpoint. A conference bridge endpoint supports more than one connection.

- **Packet relay:** Represents access that bridges two connections for interconnecting incompatible gateways or relaying them through a firewall environment. A packet relay endpoint has two connections.
- **Wiretap access point:** Represents access for recording or playing back a connection. A wiretap access point endpoint has one connection.
- **ATM trunk side interface:** Represents a single instance of an audio channel in the context of an ATM network. An ATM interface supports more than one connection.



When interacting with a gateway, the call agent directs its commands to the gateway for the express purpose of managing an endpoint or a group of endpoints. An endpoint identifier, as its name suggests, identifies endpoints.

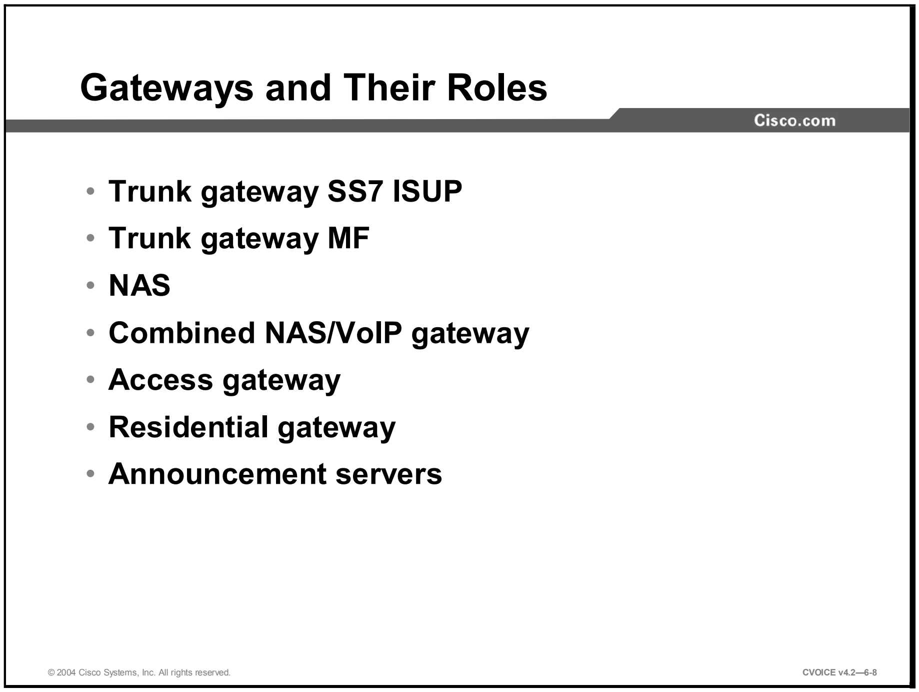
Endpoint identifiers consist of two parts: a local name of the endpoint in the context of the gateway and the domain name of the gateway itself. The two parts are separated by an "at" sign (@). If the local part represents a hierarchy, the subparts of the hierarchy are separated by a slash. In the graphic, the "local ID" may be representative of a particular "gateway/circuit #," and the "circuit #"may in turn be representative of a "circuit ID/channel #."

Example: Endpoint Identifiers

In the figure, **mgcp.gateway.cisco.com** is the domain name and **t1toSJ/17** refers to channel 17 in the T1 to San Jose.

MGCP Gateways

This topic lists several standard gateways and describes their functions.



Gateways are clustering points for endpoints. Gateways handle the translation of audio between the SCN and the packet network.

Although gateways are implemented in real systems, from a modeling point of view gateways are logical components. In this context, gateways represent a clustering of a single type and profile of endpoints.

A gateway interacts with one call agent only; therefore, it associates with one call agent at a time.

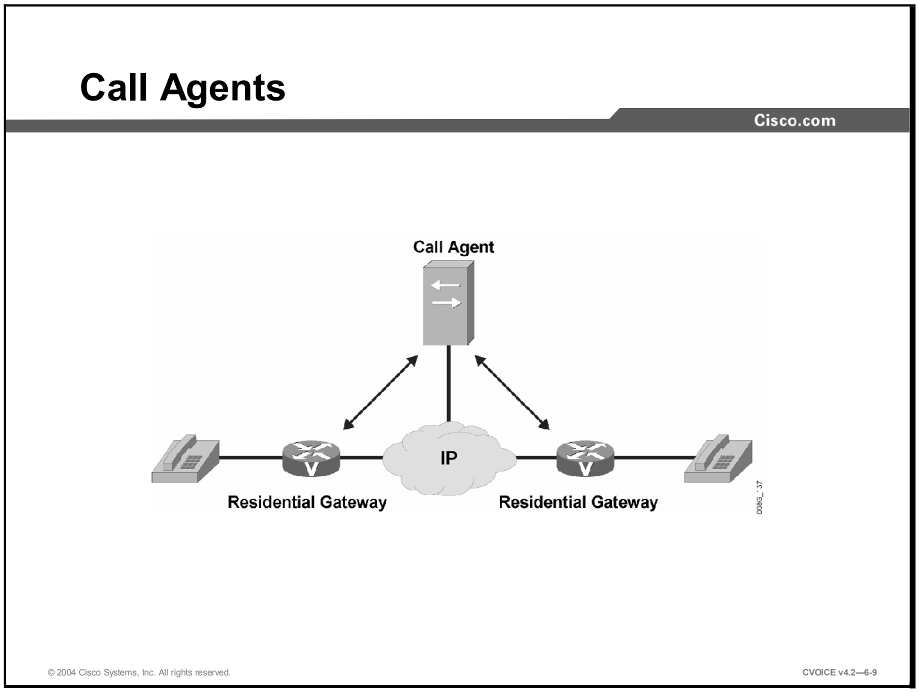
RFC 2705 identifies the following seven types of gateways:

- **Trunk gateway SS7 ISUP:** Supports digital circuit endpoints subject to ISDN signaling
- **Trunk gateway multifrequency (MF):** Typically supports digital or analog circuit endpoints that are connected to a service provider of an enterprise switch that is subject to MF signaling
- **Network access server (NAS):** Supports an interconnect to endpoints over which data (modem) applications are provided
- **Combined NAS/VoIP gateway:** Supports an interconnect to endpoints over which a combination of voice and data access is provided
- **Access gateway:** Supports analog and digital endpoints connected to a PBX
- **Residential gateway:** Supports endpoints connected to traditional analog interfaces
- **Announcement servers:** Supports endpoints that represent access to announcement services

Multiple gateway types, and multiple instances of the same type, can be incorporated into a single physical gateway implementation.

MGCP Call Agents

This topic describes how a call agent controls gateways and endpoints.



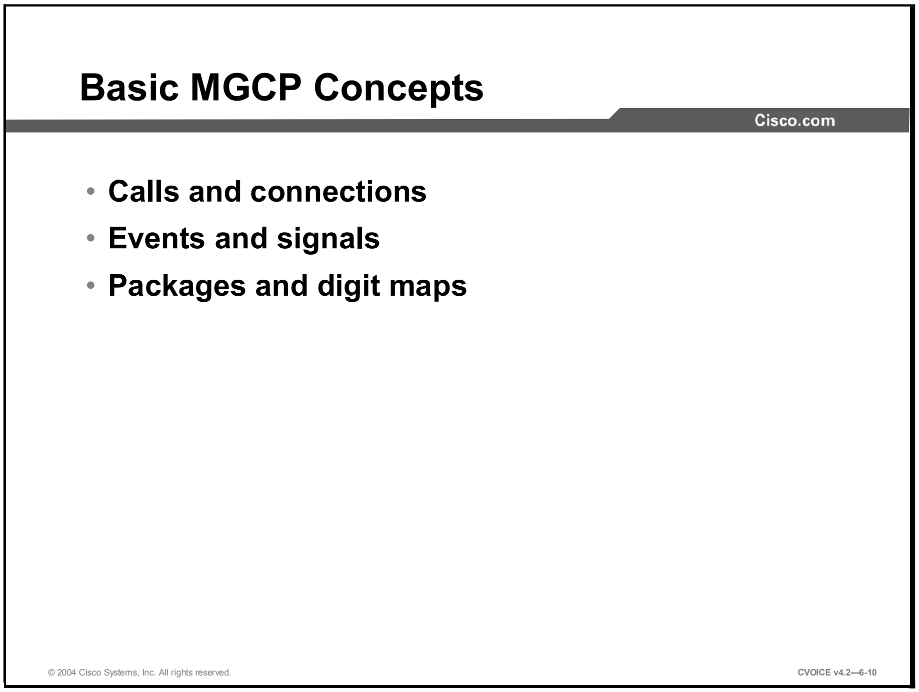
A call agent, or Media Gateway Controller (MGC), represents the central controller in an MGCP environment.

A call agent exercises control over the operation of a gateway and its associated endpoints by requesting that a gateway observe and report events. In response to the events, the call agent instructs the endpoint what signal, if any, the endpoint should send to the attached telephone equipment. This requires a call agent to recognize each endpoint type that it supports and the signaling characteristics of each physical and logical interface that is attached to a gateway.

A call agent uses its directory of endpoints and the relationship that each endpoint has with the dial plan to determine call routing. Call agents initiate all VoIP call legs.

Basic MGCP Concepts

This topic introduces the basic MGCP concepts.

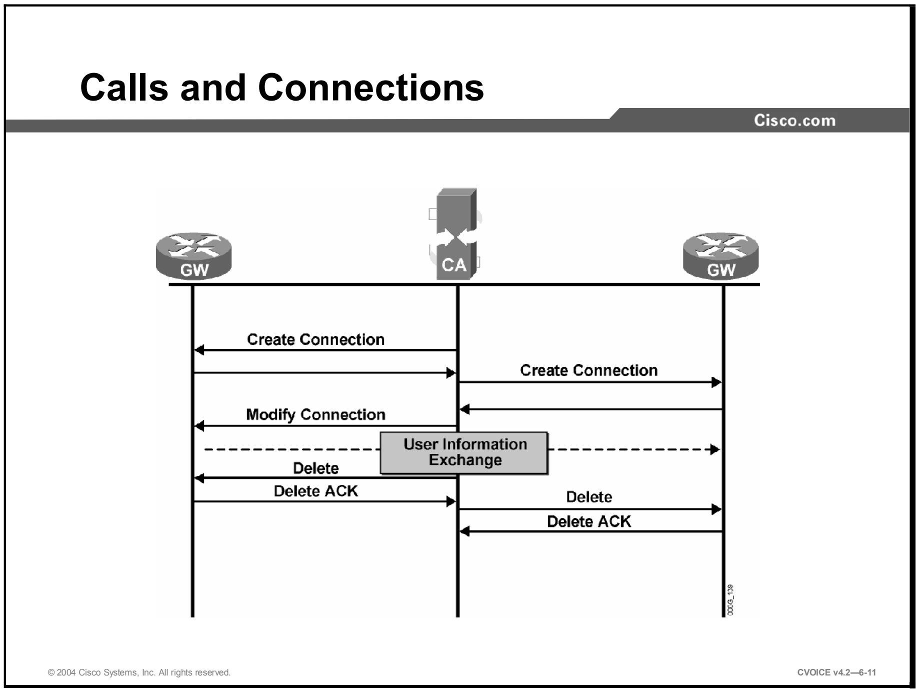


The basic MGCP concepts are listed below:

- **Calls and connections:** Allow end-to-end calls to be established by connecting two or more endpoints
- **Events and signals:** Fundamental MGCP concept that allows a call agent to provide instructions for the gateway
- **Packages and digit maps:** Fundamental MGCP concept that allows a gateway to determine the call destination

MGCP Calls and Connections

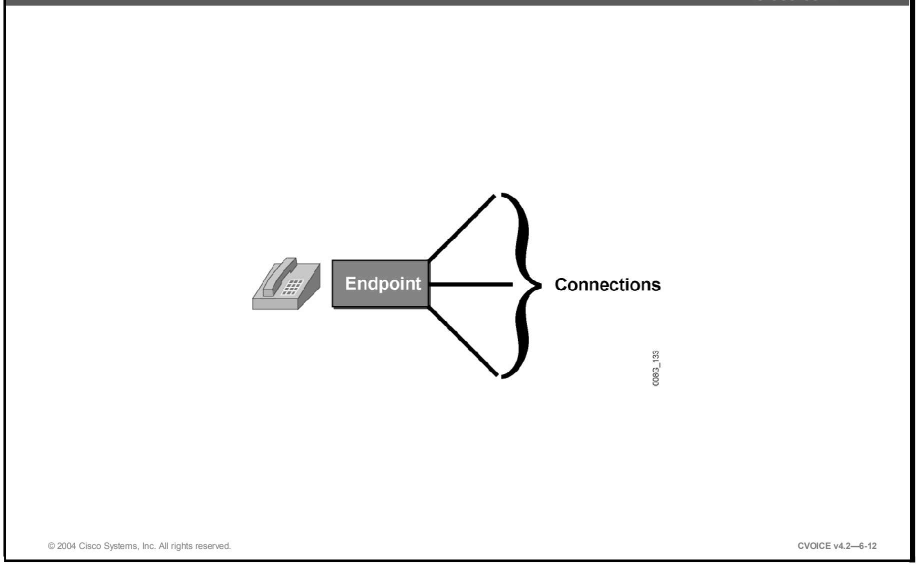
This topic discusses how end-to-end calls are established by connecting multiple endpoints.



End-to-end calls are established by connecting two or more endpoints. To establish a call, the call agent instructs the gateway that is associated with each endpoint to make a connection with a specific endpoint or an endpoint of a particular type. The gateway returns the session parameters of its connection to the call agent, which in turn sends these session parameters to the other gateway. With this method, each gateway acquires the necessary session parameters to establish RTP sessions between the endpoints. All connections that are associated with the same call will share a common call ID and the same media stream.

At the conclusion of a call, the call agent sends a delete connection request to each gateway.

Multipoint Calls



To create a multipoint call, the call agent instructs an endpoint to create multiple connections. The endpoint is responsible for mixing audio signals.

MGCP Events and Signals

This topic describes how a call agent uses events and signals to provide instruction to the gateway.

Events and Signals

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Events:

- **Continuity detection (as a result of a continuity test)**
- **Continuity tone**
- **DTMF digits**
- **Fax tones**
- **Hookflash**
- **Modem tones**
- **Off-hook transition**
- **On-hook transition**

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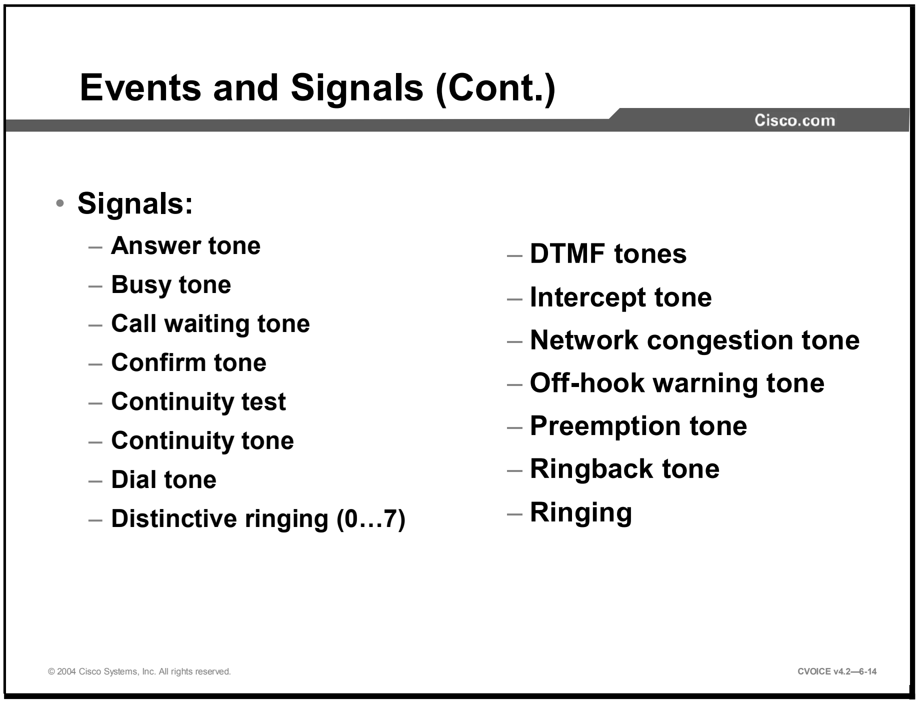
Events and signals help the call agent instruct the gateway on the call control and signaling procedures. The call agent has complete control over the gateway. The call agent informs the gateway of every action to take, including the following:

- Events that the gateway monitors on an endpoint
- What to do if an event occurs
- When to generate a notification to the call agent

An example of an event on an analog line is an off-hook condition. Using signals, the call agent requests that the gateway provide dial tone upon observing the off-hook event.

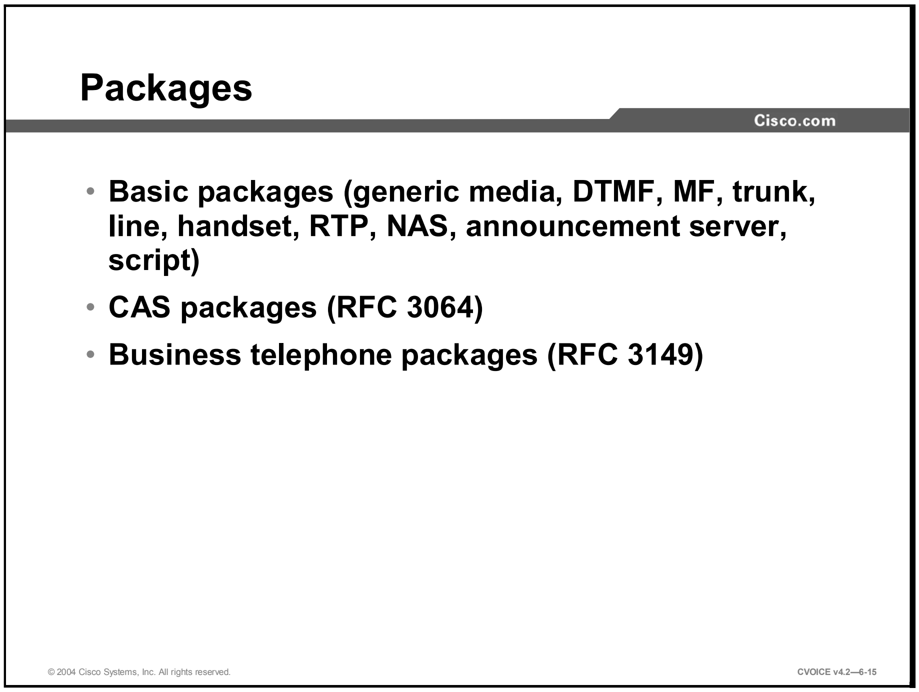
The figure lists examples of events used in MGCP environments. Events and signals are assigned simple, case-insensitive codes; for example, the code for an off-hook transition event is “hd”, and the code for the dial tone signal is “dl”.



The figure gives examples of signals used in MGCP environments.

MGCP Packages

This topic describes how events and signals are packaged in gateways and how they are used by digit maps and gateways.



Packages are collections of commonly occurring events and signals that are relevant to a specific type of endpoint; for example, "off hook," an event, and "dial tone," a signal, are unique to managing subscriber lines. Consequently, they are associated with the "line" package.

Each event and signal is placed in one particular package. The name of an event or signal acquires the code assigned to the package. The package code and the event code are separated by a slash. Therefore, the full specified name for the off-hook transition event is "L/hd".

RFC 2705 defines packages, as shown in the table.

Packages

Reference	Package Name
G	Generic media
M	Multifrequency
D	DTMF
T	Trunk
N	Network access server
L	Line
A	Announcement server
R	RTP
H	Handset
S	Script

RFC 3064 defines CAS packages, while RFC 3149 defines business telephone packages. The CAS packages contain definitions for various CAS media gateways, including support for emulated E&M interfaces, direct inward dialing (DID) interfaces, FXO interfaces, and others. Business telephone packages contain support for business telephone functions (and buttons) such as hold, transfer, forward, conference, and others.

Gateways and Their Packages

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Gateway Type	Packages
Trunk gateway (ISUP)	G, D, T, R
Trunk gateway (MF)	G, M, D, T, R
Network access server (NAS)	G, M, T, N
Combined NAS/VoIP gateway	G, M, D, T, N, R
Access gateway (VoIP)	G, D, M, R
Access gateway (VoIP & NAS)	G, D, M, N, R
Residential gateway	G, D, L, R
Announcement server	A, R

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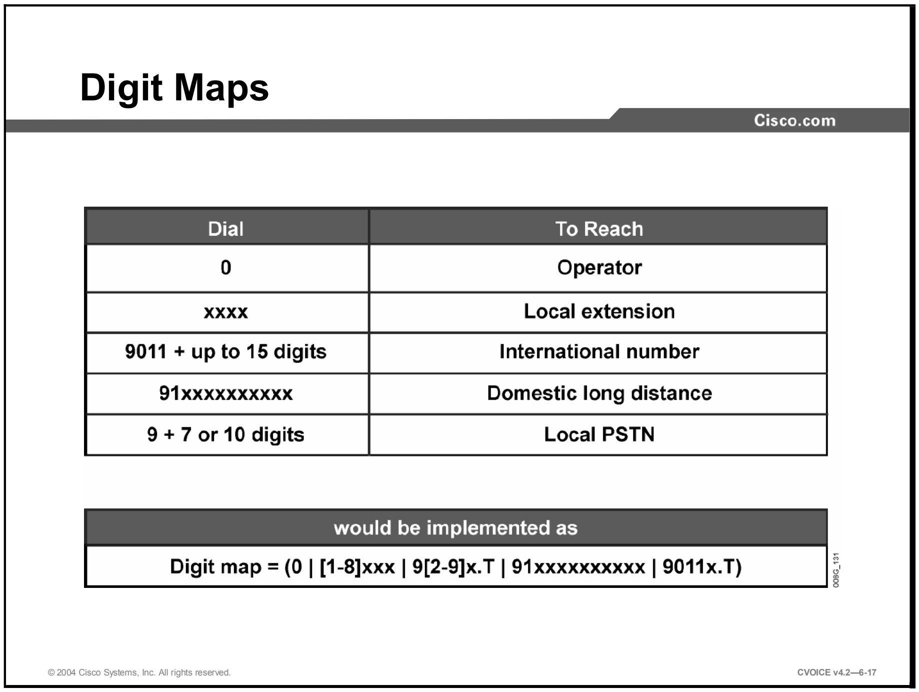
Packages cluster events and signals by their relevance to various types of endpoints. Conceptually, gateways also cluster endpoints of different types. It is appropriate then to associate packages with gateways. The table in the figure lists the gateways and identifies the packages that are associated with them.

MGCP Digit Maps

This topic describes the function of digit maps in MGCP.



Dial	To Reach
0	Operator
xxxx	Local extension
9011 + up to 15 digits	International number
91xxxxxxxxxx	Domestic long distance
9 + 7 or 10 digits	Local PSTN

would be implemented as
Digit map = (0 | [1-8]xxx | 9[2-9]x.T | 91xxxxxxxxxx | 9011x.T)

A digit map is a specification of the dial plan. When you download a digit map to a gateway for use on an endpoint or a group of endpoints, a digit map allows the gateway to collect digits until the gateway either finds a match or concludes that the dialed digits could not possibly match the specification. When either condition occurs, the gateway notifies the call agent.

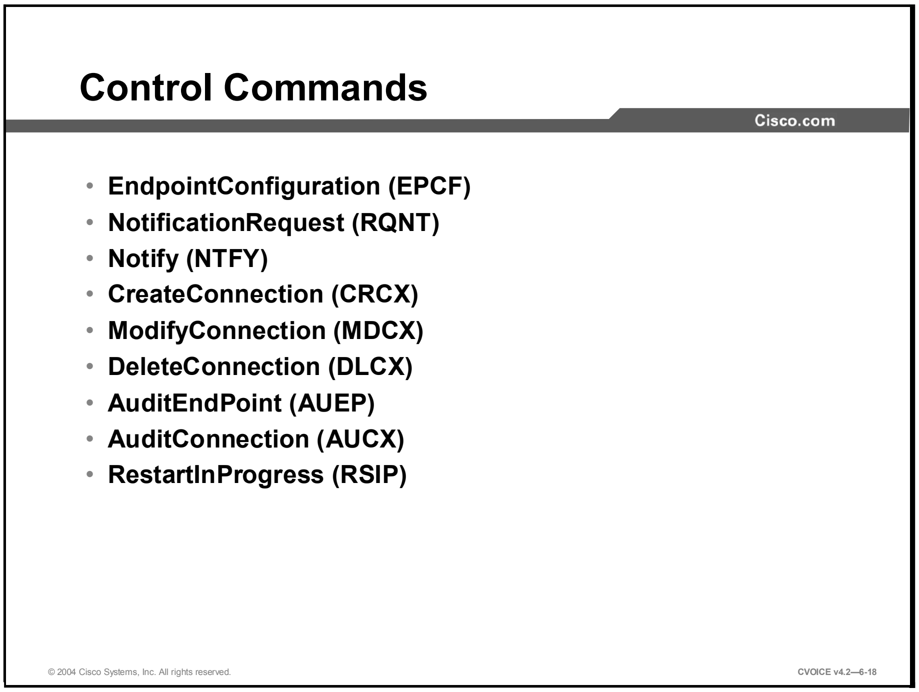
Without a digit map, a gateway must notify the call agent on each digit dialed, which places a heavy burden on the call agent and the network connecting the gateway and call agent. The figure shows an example of a digit map.

Example: Digit Map Translation

According to the figure, if a user dials 9 as the first digit, three of the possible digit maps may be invoked, depending on the next digit dialed. If the next digit is a 1, then the user is dialing domestic long distance, and the digit map 91*xxxxxxxxxx* will be used.

MGCP Control Commands

MGCP defines nine messages to control and manage endpoints and their connections. This topic describes these control messages.

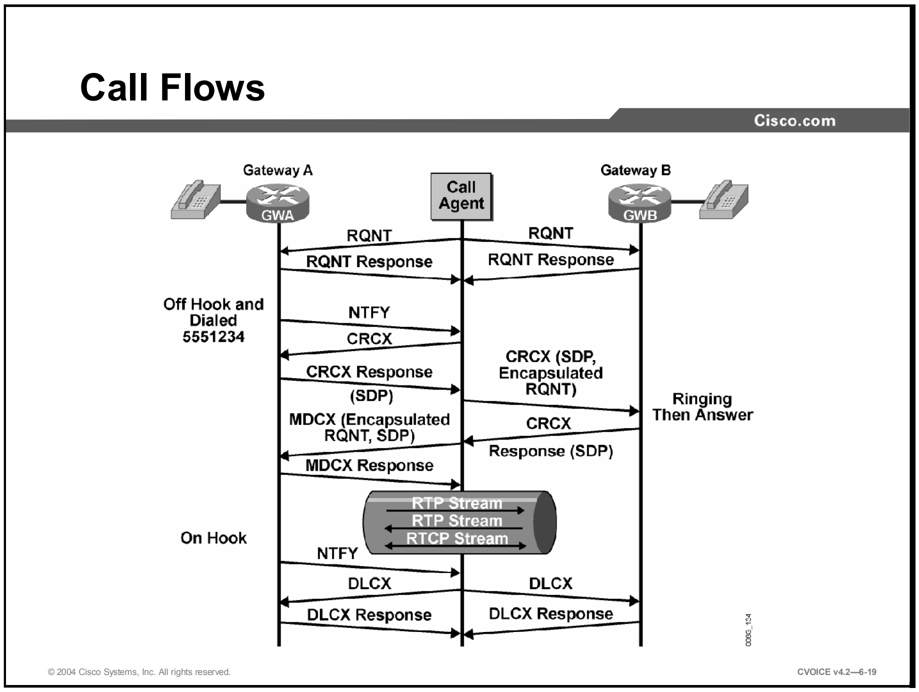


A call agent uses control commands or messages to direct its gateways and their operational behavior. Gateways use the following control commands in responding to requests from a call agent and notifying the call agent of events and abnormal behavior:

- **EndpointConfiguration (EPCF):** Identifies the coding characteristics of the endpoint interface on the line side of the gateway. The call agent issues the command.
- **NotificationRequest (RQNT):** Instructs the gateway to watch for events on an endpoint and the action to take when they occur. The call agent issues the command.
- **Notify (NTFY):** Informs the call agent of an event for which notification was requested. The gateway issues the command.
- **CreateConnection (CRCX):** Instructs the gateway to establish a connection with an endpoint. The call agent issues the command.
- **ModifyConnection (MDCX):** Instructs the gateway to update its connection parameters for a previously established connection. The call agent issues the command.
- **DeleteConnection (DLCX):** Informs the recipient to delete a connection. The call agent or the gateway can issue the command. The gateway or the call agent issues the command to advise that it no longer has the resources to sustain the call.
- **AuditEndpoint (AUEP):** Requests the status of an endpoint. The call agent issues the command.
- **AuditConnection (AUCX):** Requests the status of a connection. The call agent issues the command.
- **RestartInProgress (RSIP):** Notifies the call agent that the gateway and its endpoints are removed from service or are being placed back in service. The gateway issues the command.

Call Flows

This topic illustrates and explains the interactions between a call agent and its associated gateways.



The figure illustrates a dialog between a call agent and two gateways. Although the gateways in this example are both residential gateways, the following principles of operation are the same for other gateway types:

1. The call agent sends a notification request (RQNT) to each gateway. Because they are residential gateways, the request instructs the gateways to wait for an off-hook transition (event). When the off-hook transition event occurs, the call agent instructs the gateways to supply dial tone (signal). The call agent asks the gateway to monitor for other events as well. By providing a digit map in the request, the call agent can have the gateway collect digits before it notifies the call agent.

2. The gateways respond to the request. At this point, the gateways and the call agent wait for a triggering event.

3. A user on gateway A goes off hook. As instructed by the call agent in its earlier request, the gateway provides dial tone. Because the gateway is provided with a digit map, it begins to collect digits (as they are dialed) until either a match is made or no match is possible. For the remainder of this example, assume that the digits *match* a digit map entry.

4. Gateway A sends a notify (NTFY) to the call agent to advise the call agent that a requested event was observed. The notify identifies the endpoint, the event, and, in this case, the dialed digits.

5. After confirming that a call is possible based on the dialed digits, the call agent instructs gateway A to create a connection (CRCX) with its endpoint.

6. The gateway responds with a session description if it is able to accommodate the connection. The session description identifies at least the IP address and UDP port for use

in a subsequent RTP session. The gateway does not have a session description for the remote side of the call, and the connection enters a wait state.

7. The call agent prepares and sends a connection request to gateway B. In the request, the call agent provides the session description obtained from gateway A. The connection request is targeted to a single endpoint—if only one endpoint is capable of handling the call—or to any one of a set of endpoints. The call agent also embeds a notification request that instructs the gateway about the signals and events that it should now consider relevant. In this example, in which the gateway is residential, the signal requests ringing and the event is an off-hook transition.

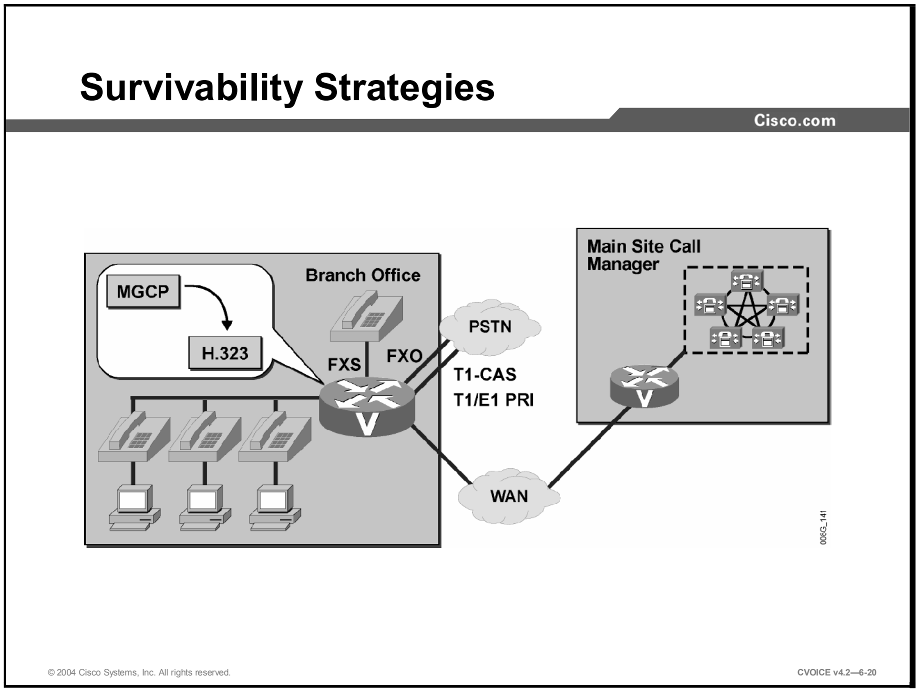
Note The interaction between gateway B and its attached user has been simplified.

---

8. Gateway B responds to the request with its session description. Notice that gateway B has both session descriptions and recognizes how to establish its RTP sessions.
9. The call agent relays the session description to gateway A in a modify connection request (MDCX). This request may contain an encapsulated notify request that describes the relevant signals and events at this stage of the call setup. Now gateway A and gateway B have the required session descriptions to establish the RTP sessions over which the audio travels.
10. At the conclusion of the call, one of the endpoints recognizes an on-hook transition. In the example, the user on gateway A hangs up. Because the call agent requested the gateways to notify in such an event, gateway A notifies the call agent.
11. The call agent sends a delete connection (DLCX) request to each gateway.
12. The gateways delete the connections and respond.

Survivability Strategies

Maintaining high availability in an MGCP environment requires a design that accommodates the failure of a call agent. This topic describes two strategies for managing the loss of a call agent.



In the MGCP environment, the call agent controls all call setup processing on the IP and the telephony sides of a gateway. Because a gateway is associated with only one call agent at a time, if that call agent fails or is inaccessible for any reason, the gateway and its endpoints are left uncontrolled and, for all practical purposes, useless. Cisco Systems has developed two methods to handle lost communication between a call agent and its gateways: MGCP switchover and switchback and MGCP gateway fallback. These features operate in the following manner:

- **MGCP switchover and switchback:** MGCP switchover permits the use of redundant MGCP call agents. This feature requires two or more Cisco CallManager servers to operate as MGCP call agents. One Cisco CallManager server becomes the primary server and functions as the MGCP call agent. The other Cisco CallManager servers remain available as backup servers.

  The MGCP gateway monitors MGCP messages sent by the Cisco CallManager server. If traffic is undetected, the gateway transmits keepalive packets to which the Cisco CallManager server responds. If the gateway does not detect packets from the Cisco CallManager for a specified period, it tries to establish a new connection with a backup Cisco CallManager server.

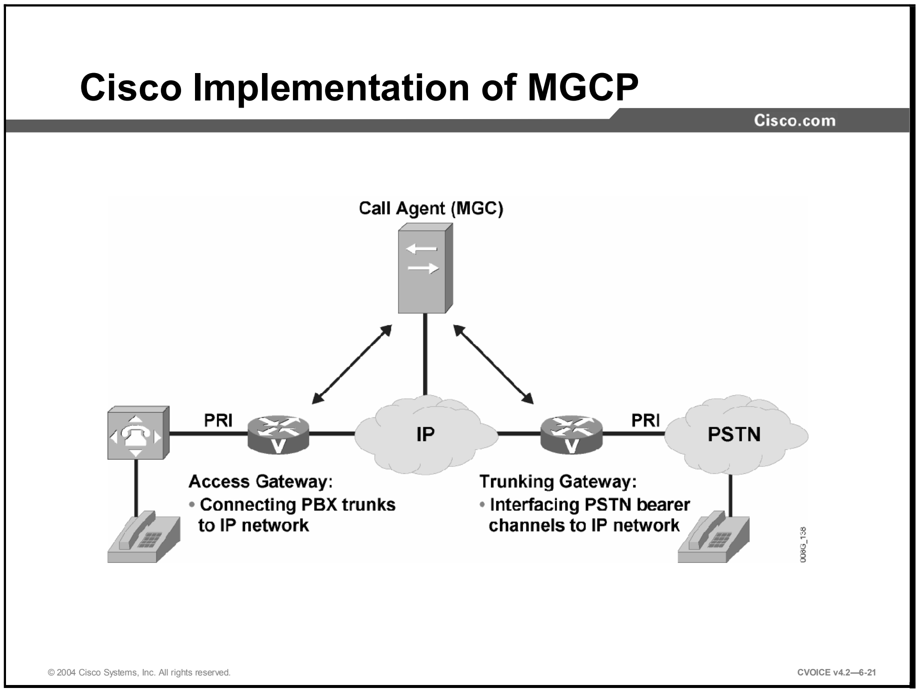
  You can configure a Cisco voice gateway to reestablish connection with the primary Cisco CallManager server when it becomes available again. This is the switchback function.

- **MGCP gateway fallback:** MGCP gateway fallback is a feature that improves the reliability of MGCP branch networks. A WAN link connects the MGCP gateway at the remote site to the Cisco CallManager at the central sites (the MGCP call agent). If the WAN link fails, the fallback feature keeps the gateway working as an H.323 gateway.

  MGCP gateway fallback works in conjunction with the Survivable Remote Site Telephony (SRST) feature. SRST allows Cisco gateways and routers to manage connections temporarily for Cisco IP Phones when a connection to a Cisco CallManager is unavailable.

Cisco Implementation of MGCP

This topic describes how Cisco implements MGCP.



Cisco provides support for MGCP gateways and the call agent in the following way:

- **Gateways:** Cisco implements MGCP trunk gateway and residential gateway support in the following devices:
  - Cisco voice-enabled routers (first available in Cisco IOS Release 12.1)
  - Cisco PGW 2200 PSTN gateways
  - Cisco Voice Gateway 224 (VG224)
  - Voice-enabled AS5*xx*0 access servers
  - BTS 10200 Softswitch

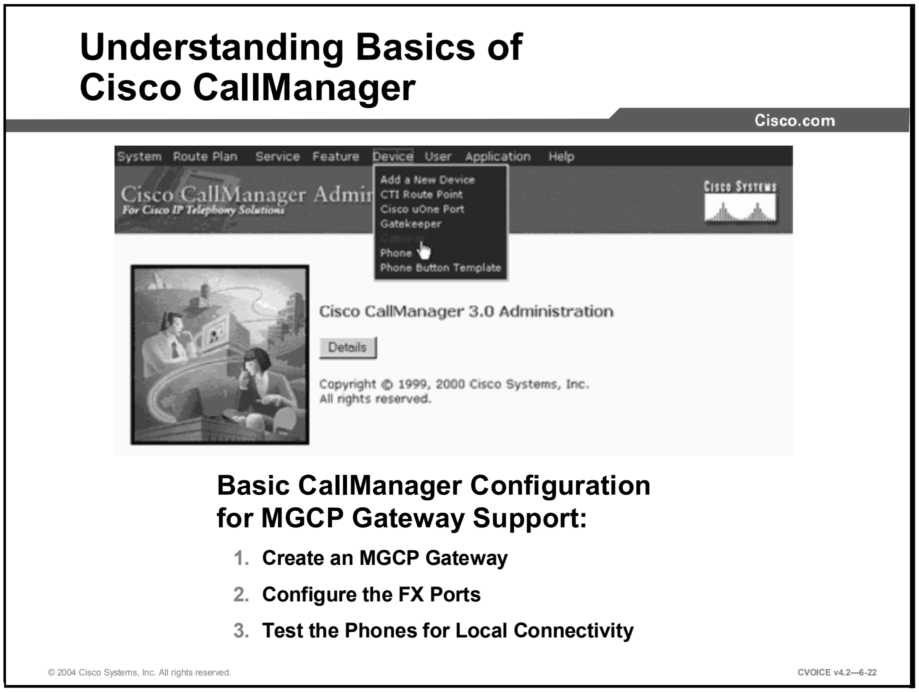
Note Cisco CallManager interworking requires Cisco IOS Release 12.2.

- **Call agent:** Cisco implements call agent support in the following applications:
  - Cisco CallManager
  - BTS 10200 Softswitch

Note Residential gateway and trunk gateway support does *not* include all analog and digital signaling types on the telephone interfaces. Check Cisco.com for an up-to-date list.

Understanding Basics of Cisco CallManager

This topic describes basic configuration of Cisco CallManager to support MGCP gateways. For detailed information regarding configuration of CallManager, refer to the *Cisco IP Telephony* (CIPT) course.



There are three basic tasks for configuring a Cisco CallManager for MGCP. The basic steps for each task are as follows:

- Task 1: Create an MGCP gateway

1. Use the Device Wizard to create an MGCP gateway. Select **Device** > **Gateway**.
2. Click **Add New Gateway**.
3. Select the appropriate gateway type (such as 26*xx*, 36*xx*, or 37*xx*).
4. Click **Next**.
5. For the **MGCP Domain Name**, use the actual host name you have assigned to the gateway. Specify the carrier module that the gateway has installed.
6. Click **Insert**.

- Task 2: Configure the FX ports

1. Identify the voice interface card (VIC) modules installed in the gateway.
2. Click **Update** to activate the changes.
3. The FXO and FXS ports appear at the bottom right of the next screen. These are also referred to as EndPoint Identifiers.

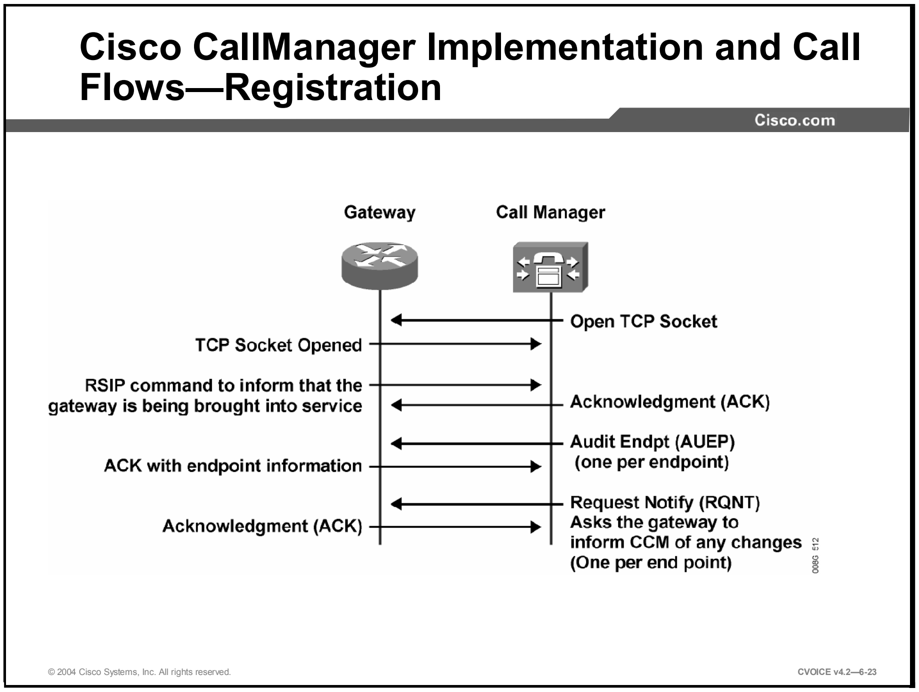
4. Select one of the ports.
5. Select the correct signaling type, **loop** or **ground start**.
6. Configure the parameters of the MGCP Member Configuration screen as required. For example, if this was an FXS port, you would be required to select loop or ground start signaling.
7. Click **Insert**.
8. For FXS ports only, click the **Add DN** text. Add the directory number (phone number) as appropriate, then click **Insert and Close**.
9. Select the **Back To MGCP Configuration** option.
10. Click **Reset Gateway**.

■ Task 3: Test the phones for local connectivity

You should now have dial tone on the analog phones connected to the FXS ports. Try dialing from one FXS port to another. You should be able to make and receive calls between these ports.

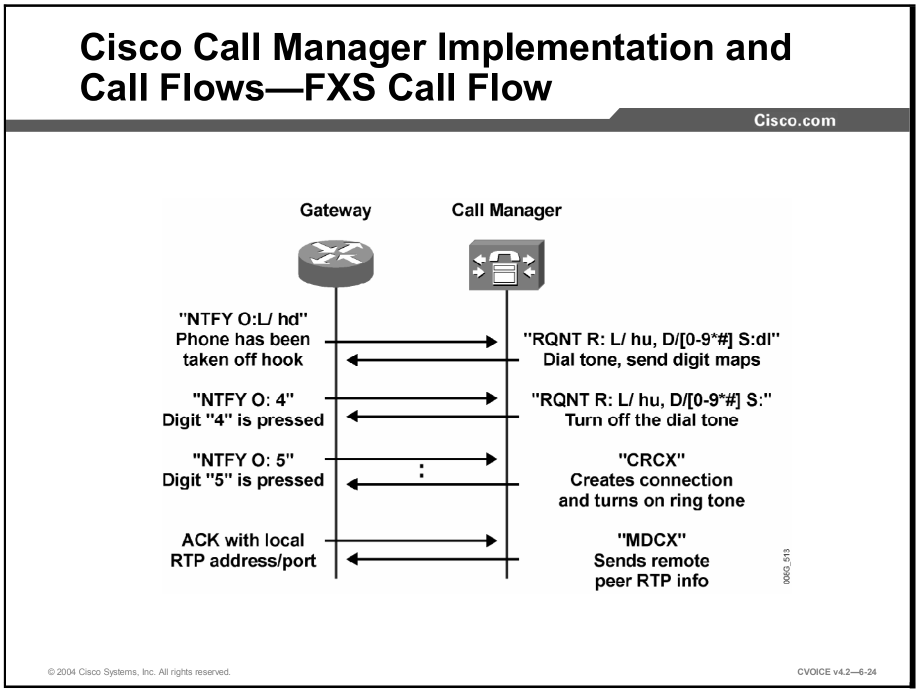
Registration and FXS Call Flow

Cisco CallManager implementation of MGCP uses specific command sequences to perform a variety of tasks. The figures are examples of how calls are made and how the gateways are registered.



Example: Registration

The figure describes how Cisco CallManager registers voice gateways in its database using MGCP. The ACK commands are standard TCP acknowledgments of the received command.

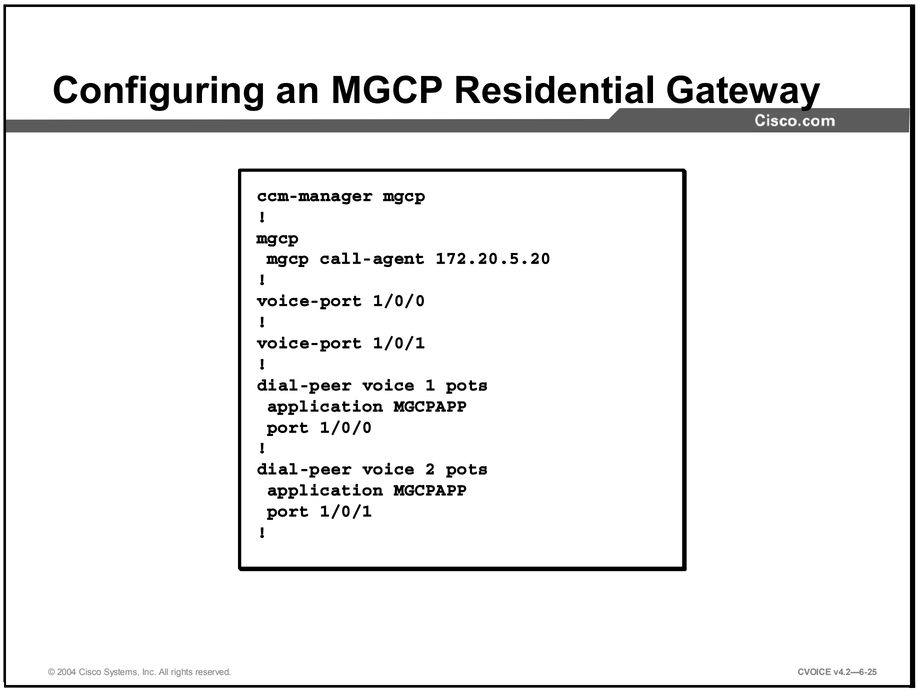


Example: FXS Call Flow

The figure shows a sample FXS call flow (dialing and connection).

Configuring MGCP

This topic illustrates the configuration commands that are required to implement MGCP residential gateway and trunk gateway capabilities on a Cisco router.



The figure highlights the commands required to configure an MGCP residential gateway.

MGCP is invoked with the **mgcp** command. If the call agent expects the gateway to use the default port (UDP 2427), the **mgcp** command is used without any parameters. If the call agent requires a different port, then the port must be configured as a parameter in the **mgcp** command; for example, **mgcp 5036** would tell the gateway to use port 5036 instead of the default port.

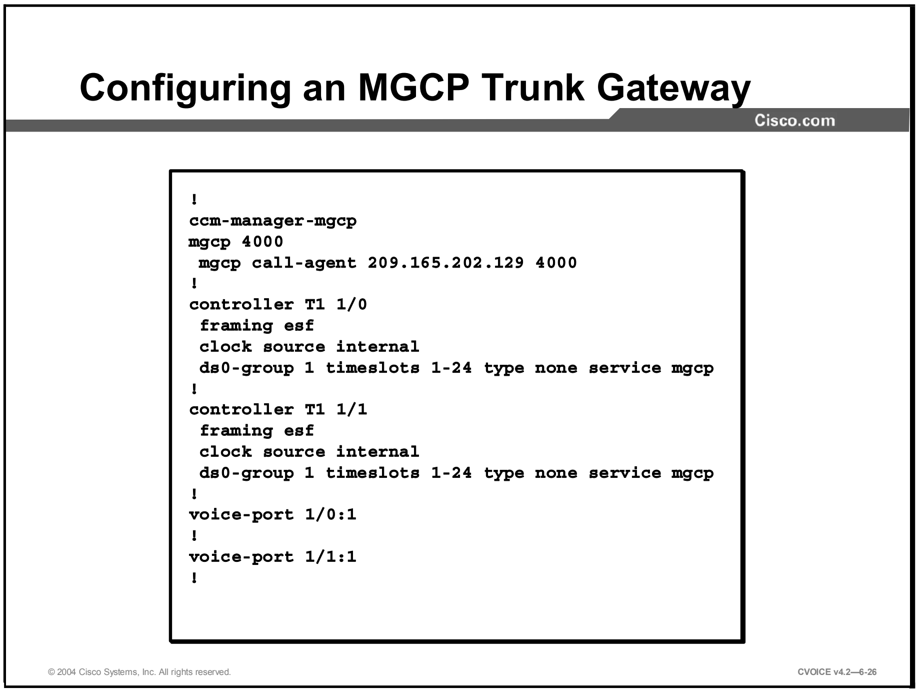
At least one **mgcp call-agent** command is required below the **mgcp** command. This command indicates the location of the call agent. The command identifies the call agent by an IP address or a host name. Using a host name adds a measure of fault tolerance in a network that has multiple call agents. When the gateway asks the DNS for the IP address of the call agent, the DNS may provide more than one address, in which case the gateway can use either one. If multiple instances of the **mgcp call-agent** command are configured, the gateway uses the first call agent to respond.

Other **mgcp** subcommands are optional.

Example: MGCP Residential Gateway Configuration

In the example, the configuration identifies the packages that the gateway expects the call agent to use when it communicates with the gateway. The last **mgcp** command specifies the default the gateway uses if the call agent does not share the capabilities. In this example, the command is redundant because the line package is the default for a residential gateway.

When the parameters of the MGCP gateway are configured, the active voice ports (endpoints) are associated with the MGCP. Dial peer 1 illustrates an **application mgcpapp** subcommand. This command binds the voice port (**1/0/0** in this case) to the MGCP. Also, notice that the dial peer does not have a destination pattern. A destination pattern is not used because the relationship between the dial number and the port is maintained by the call agent.



Note The **ccm-manager-mgcp** command is required only if the call agent is a Cisco CallManager.

The second example illustrates the configuration of a trunk gateway.

Configuring trunk gateways requires the address or the name of the call agent, which is a requirement common to a residential gateway (RGW). The trunk package is the default for a trunk gateway and does not need to be configured. Again, other parameters are optional.

Example: Configuring MGCP Trunk Gateway

The figure illustrates commands for configuring a trunk gateway. Instead of using the **application mgcpapp** command in a dial peer, a trunk endpoint identifies its association with MGCP using the **service mgcp** parameter in the **ds0-group** controller subcommand. As always in MGCP, the call agent maintains the relationship between the endpoint (in this case a digital trunk) and its address.

Monitoring and Troubleshooting MGCP

Several **show** and **debug** commands provide support for monitoring and troubleshooting MGCP. This topic lists many useful **show** and **debug** commands.

Example: show Command

Cisco.com

```
Router# show mgcp statistics

UDP pkts rx 8, tx 9
Unrecognized rx pkts 0, MGCP message parsing errors 0
Duplicate MGCP ack tx 0, Invalid versions count 0
CreateConn rx 4, successful 0, failed 0
DeleteConn rx 2, successful 2, failed 0
ModifyConn rx 4, successful 4, failed 0
DeleteConn tx 0, successful 0, failed 0
NotifyRequest rx 0, successful 4, failed 0
AuditConnection rx 0, successful 0, failed 0
AuditEndpoint rx 0, successful 0, failed 0
RestartInProgress tx 1, successful 1, failed 0
Notify tx 0, successful 0, failed 0
ACK tx 8, NACK tx 0
ACK rx 0, NACK rx 0
IP address based Call Agents statistics:
IP address 10.24.167.3, Total msg rx 8, successful 8,
failed 0
```

 CVOICE v4.2—6-27

The figure illustrates the output of one of the **show** commands. The **show** and **debug** commands are valuable for examining the current status of the MGCP components and for troubleshooting. You should be familiar with the information provided from each command and how this information can help you.

The following **show** commands are useful for monitoring and troubleshooting MGCP:

- **show call active voice [brief]:** Displays the status, statistics, and parameters for all active voice calls. When the call is disconnected, this information is transferred to the history records.
- **show call history voice [last *n* | record | brief]:** Displays call records from the history buffer.
- **show mgcp:** Displays basic configuration information about the gateway.
- **show mgcp connection:** Displays details of the current connections.
- **show mgcp endpoint:** Displays a list of the voice ports that are configured for MGCP.
- **show mgcp statistics:** Displays a count of the successful and unsuccessful control commands (shown in the figure). You should investigate a high unsuccessful count.

The following **debug** commands are useful for monitoring and troubleshooting MGCP:

- **debug voip ccapi inout:** Shows every interaction with the call control API on the telephone interface and the VoIP side. Watching the output allows users to follow the progress of a call from the inbound interface or VoIP peer to the outbound side of the call. This debug is very active; you must use it sparingly in a live network.
- **debug mgcp [all | errors | events | packets | parser]:** Reports all **mgcp** command activity. You must use this debug to trace the MGCP request and responses.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

Cisco.com

- MGCP defines an environment for controlling telephony gateways from a centralized call agent.
- MGCP components include endpoints, gateways, and call agents.
- Calls are created by connecting endpoints. Endpoints can be physical or logical.
- MGCP gateways incorporate endpoints, act upon directives issued by a call agent to manage the telephony interface, and translate voice signals.
- The call agent instructs the MGCP gateway to watch for events and provides signaling on its telephony interfaces.
- Calls and connections, events and signals, and packages and digit maps are basic concepts in MGCP.

 CVOICE v4.2—6-28

Summary (Cont.)

Cisco.com

- During call setup, the gateway associated with each endpoint makes a connection with a specific endpoint and returns the session parameters to the call agent. The call agent sends these session parameters to the other gateway to establish the call.
- The call agent uses events and signals to instruct the gateway on all call control and signaling procedures.
- Events and signals relevant to a specific type of endpoint are packaged together in basic, CAS, or business telephone packages.
- A digit map provides the gateway with a representation of the dial plan.
- A call agent and its gateways exchange requests and responses by way of control commands.

 CVOICE v4.2—6-29

Summary (Cont.)

Cisco.com

- MGCP switchover and switchback and MGCP gateway fallback are two strategies for improving availability in an MGCP implementation.
- Cisco implements an MGCP call agent in the Cisco CallManager. Gateways of various types are implemented by routers or specialized gateway products.
- Cisco CallManager can be configured to support MGCP gateways.
- The mgcp command can be used to configure residential and trunk gateways on a Cisco router.
- Several show and debug commands help to monitor and troubleshoot.

 CVOICE v4.2—6-30

References

For additional information, refer to these resources:

- IETF RFC 2705: *Media Gateway Control Protocol (MGCP), version 1.0*
  http://www.faqs.org/rfcs/rfc2705.html
- IETF RFC 2805: *Media Gateway Control Protocol Architecture and Requirements*
  http://www.faqs.org/rfcs/rfc2805.html
- *Cisco IOS Voice, Video, and Fax Configuration Guide*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_configuration_guide_book09186a0080080ada.html
- *Cisco IOS Voice, Video, and Fax Command Reference*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1835/products_command_reference_book09186a0080080c8b.html
- *Configuring the Cisco CallManager Server*
  http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a008017825e.shtml

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 6-3: VoIP with MGCP

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which call control model is used by MGCP?

- A) distributed
- B) centralized
- C) ad hoc
- D) hybrid

Q2) Which protocol is used by MGCP to describe the type of session to initiate?

- A) SIP
- B) CDP
- C) SDP
- D) MGC

Q3) Which MGCP component represents the point of interconnection between the packet network and the traditional telephone network?

- A) endpoint
- B) gate-array
- C) gatekeeper
- D) call agent

Q4) What is the function of an MGCP gateway?

- A) handles the translation of video between the SCN and packet-switched network
- B) handles the translation of audio between the SCN and packet-switched network
- C) controls the operation of the endpoints and the call agent
- D) only allows authenticated traffic into the network

Q5) Which type of MGCP endpoint represents an access type that bridges two connections for interconnecting incompatible gateways?

- A) DS0
- B) analog line
- C) IVR access point
- D) packet relay
- E) wiretap access point

Q6) Which type of MGCP endpoint can have only one connection?

A) ATM trunk side interface

B) wiretap access point

C) analog line

D) DS0

E) packet relay

Q7) How many call agents can an MGCP gateway interact with?

A) one

B) two

C) seven

D) varies

Q8) Match the type of MGCP gateway with its description.

A) trunk gateway ISUP

B) NAS

C) access gateway

D) residential gateway

_____ 1. supports interconnect to endpoints over which data (modem) applications are provided

_____ 2. supports digital circuit endpoints subject to ISDN signaling

_____ 3. supports endpoints connected to traditional analog interfaces

_____ 4. supports analog and digital endpoints connected to a PBX

Q9) Why does the MGCP require the gateway to observe and report events?

A) so that it can recognize each endpoint that it supports

B) so that it can tell the endpoint what type of signal to send to the attached telephone equipment

C) so that it can recognize the signaling characteristics of each physical interface attached to the gateway

D) so that it can recognize the signaling characteristics of each logical interface attached to the gateway

Q10) Which two pieces of information does the MGCP use to determine call routing? (Choose two.)

A) its directory of endpoints

B) the endpoint type

C) the events reports sent by the gateway

D) the relationship of endpoints with the dial plan

E) the signaling characteristics of the gateway interfaces

Q11) Which two concepts does MGCP use to determine the destination of a call? (Choose two.)

A) calls

B) connections

C) digit maps

D) events

E) packages

F) signals

Q12) Which two concepts does MGCP use to allow a call agent to provide instructions to a gateway? (Choose two.)

A) calls

B) connections

C) digit maps

D) events

E) packages

F) signals

Q13) Which MGCP component is responsible for mixing audio signals in a multipoint call?

A) call agent

B) MGC

C) endpoint

D) gateway

Q14) At the conclusion of an MGCP call, which message does the call agent send to each gateway?

A) bye request

B) cancel request

C) disconnect request

D) delete connection request

Q15) Which two examples represent MGCP events? (Choose two.)

A) busy tone

B) continuity test

C) distinctive ringing

D) fax tones

E) hookflash

F) ringing

Q16) Which two examples represent MGCP signals? (Choose two.)

A) confirm tone

B) continuity detection

C) continuity test

D) DTMF digits

E) on-hook transition

Q17) Which two packages are associated with the NAS? (Choose two.)

A) multifrequency

B) DTMF

C) trunk

D) line

E) announcement server

F) RTP

Q18) Which three packages are associated with the residential gateway? (Choose three.)

A) generic media

B) multifrequency

C) DTMF

D) trunk

E) network access server

F) line

Q19) A gateway that is using a digit map notifies the call agent when which two things have occurred? (Choose two.)

A) each digit is collected

B) all the digits have been collected

C) the gateway finds a match for the digits

D) the digits that match the area code have been collected

E) the gateway concludes that the dialed digits cannot be matched

Q20) In the United States, how would the digit map identify digits for the local PSTN?

A) xxxx

B) 91xxxxxxxxxx

C) 9 + 7 or 10 digits

D) 9011 + up to 15 digits

Q21) Match the MGCP control command with its function.

A) EndpointConfiguration

B) NotificationRequest

C) ModifyConnection

D) AuditEndpoint

E) RestartInProgress

_____ 1. requests the status of an endpoint

_____ 2. instructs the gateway on what action to take on the occurrence of an event

_____ 3. identifies the coding characteristics of the endpoint interface on the line side of the gateway

_____ 4. notifies the call agent that the gateway and its endpoints are removed from service

_____ 5. instructs the gateway to update its connection parameters for a previously established connection

Q22) Which two commands are issued by a gateway? (Choose two.)

A) EndpointConfiguration

B) NotificationRequest

C) CreateConnection

D) DeleteConnection

E) RestartInProgress

Q23) Which two types of information are included in the notify (NTFY) sent by a gateway to the call agent? (Choose two.)

A) call destination

B) endpoint identification

C) event identification

D) local gatekeeper

E) signal type

Q24) In MGCP calls, when do the gateways delete a connection?

A) when one user hangs up

B) when one endpoint recognizes an on-hook transition

C) when the gateway notifies the call agent of an on-hook transition event

D) when the call agent instructs the gateways to delete the connection

Q25) What is the MGCP switchback function?

A) The gateway establishes a connection with the backup Cisco CallManager server after failing to get packets from the primary Cisco CallManager.

B) The gateway reestablishes a connection with the primary Cisco CallManager server when it becomes available.

C) When the WAN link between the gateway and Cisco CallManager fails, the gateway continues to function as an H.323 gateway.

D) Cisco gateways manage connections temporarily when a connection to Cisco CallManager goes down.

Q26) When Cisco CallManager is unavailable, which feature works with MGCP gateway fallback to manage connections temporarily for Cisco IP Phones?

A) SRST

B) RSVP

C) Cisco VG200

D) BTS 10200 Softswitch

Q27) Which two Cisco applications support the MGCP call agent? (Choose two.)

A) Cisco voice-enabled routers with Cisco IOS Release 12.1 and later

B) Cisco PGW 2200 PSTN gateways

C) Cisco CallManager

D) Cisco VG200

E) BTS 10200 Softswitch

Q28) Which Cisco application does NOT provide residential gateway support?

A) Cisco CallManager

B) Cisco voice-enabled routers with Cisco IOS Release 12.1 and later

C) Cisco PGW 2200 PSTN gateways

D) BTS 10200 Softswitch

Q29) How many AUEP (Audit Endpoint) messages are sent by CallManager during registration?

A) six

B) one per gateway

C) sixteen

D) one per endpoint

Q30) What does the "NOTIFY O:L/hd" message indicate?

A) dial tone

B) phone has been taken off hook

C) phone has been placed on hook

D) endpoint has been written to hard drive (hd)

Q31) Which configuration command enables MGCP on UDP port 5000?

A) **mgcp 5000** global configuration command

B) **mgcp udp 5000** global configuration command

C) **mgcp 5000** interface configuration subcommand

D) **mgcp 2427** global configuration subcommand

Q32) How do you configure a router to use MGCP on a digital port?

A) Add the **application mgcpapp** subcommand to the dial peer.

B) Add the **service mgcp** subcommand to the dial peer.

C) Add the parameter application mgcpapp to the ds0-group controller subcommand.

D) Add the **service mgcp** parameter to the **ds0-group controller** subcommand.

Q33) Which two commands display the current MGCP calls? (Choose two.)

A) **show call active voice**

B) **show mgcp endpoints**

C) **show mgcp connections**

D) **debug mgcp packets**

E) **show mgcp statistics**

Q34) Which command allows you to view the details of every MGCP packet?

A) **debug voip ccapi inout**

B) **show mgcp packets**

C) **show call active voice**

D) **debug mgcp packets**

Quiz Answer Key

Q1) B

Relates to: MGCP and its Associated Standards

Q2) C

Relates to: MGCP and its Associated Standards

Q3) A

Relates to: Basic MGCP Components

Q4) B

Relates to: Basic MGCP Components

Q5) D

Relates to: MGCP Endpoints

Q6) B

Relates to: MGCP Endpoints

Q7) A

Relates to: MGCP Gateways

Q8) 1-B, 2-A, 3-D, 4-C

Relates to: MGCP Gateways

Q9) B

Relates to: MGCP Call Agents

Q10) A, D

Relates to: MGCP Call Agents

Q11) C, E

Relates to: Basic MGCP Concepts

Q12) D, F

Relates to: Basic MGCP Concepts

Q13) C

Relates to: MGCP Calls and Connections

Q14) D

Relates to: MGCP Calls and Connections

Q15) D, E

Relates to: MGCP Events and Signals

Q16) A, C

Relates to: MGCP Events and Signals

Q17) A, C

Relates to: MGCP Packages

Q18) A, C, F

Relates to: MGCP Packages

Q19) C, E

Relates to: MGCP Digit Maps

Q20) C

Relates to: MGCP Digit Maps

Q21) 1-D, 2-B, 3-A, 4-E, 5-C

Relates to: MGCP Control Commands

Q22) D, E

Relates to: MGCP Control Commands

Q23) B, C

Relates to: Call Flows

Q24) D

Relates to: Call Flows

Q25) B

Relates to: Survivability Strategies

Q26) A

Relates to: Survivability Strategies

Q27) C, E

Relates to: Cisco Implementation of MGCP

Q28) A

Relates to: Cisco Implementation of MGCP

Q29) D

Relates to: Understanding Basics of Cisco CallManager

Q30) B

Relates to: Understanding Basics of Cisco CallManager

Q31) A

Relates to: Configuring MGCP

Q32) D

Relates to: Configuring MGCP

Q33) A, C

Relates to: Monitoring and Troubleshooting MGCP

Q34) D

Relates to: Monitoring and Troubleshooting MGCP

Comparing Call Control Models

Overview

This lesson compares the features and functions of the three call control models: H.323, SIP, and MGCP. This lesson also highlights the environments for which each call control model is best suited.

Relevance

Understanding the capabilities of the H.323, SIP, and MGCP models helps you decide which call control model best meets your requirements.

Objectives

Upon completing this lesson, you will be able to determine the best call control model for your network. This includes being able to meet these objectives:

- Compare the features and benefits of H.323, SIP, and MGCP
- Describe the environments best suited to H.323, SIP, and MGCP

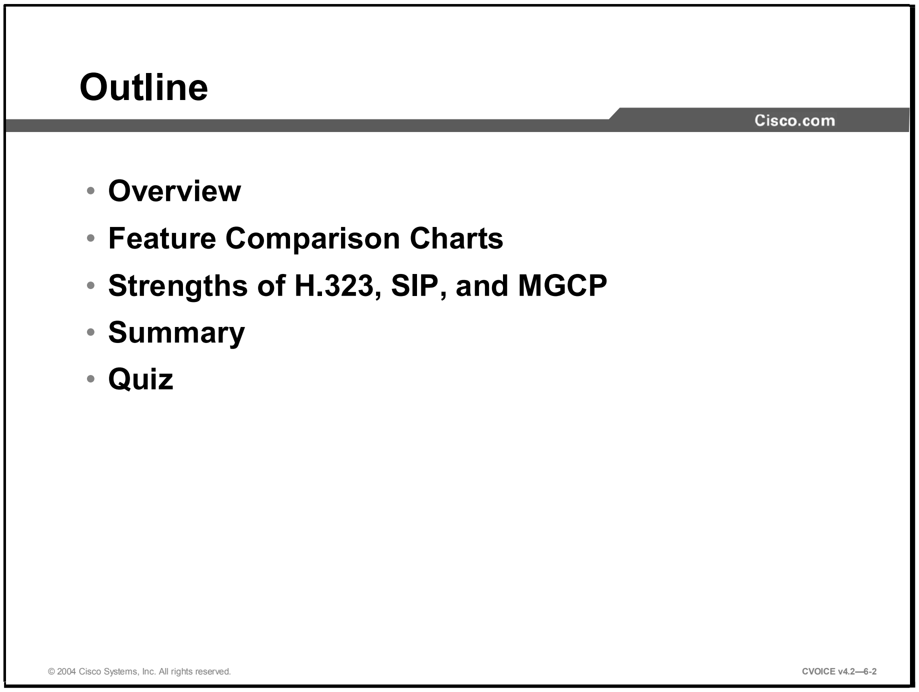
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- An understanding of the objectives and principles of signaling and call control in the context of VoIP
- An understanding of the H.323, SIP, and MGCP call control models

Outline

The outline lists the topics included in this lesson.



Feature Comparison Charts

This topic compares the origins, architectures, characteristics, and capabilities of the H.323, SIP, and MGCP call control models.

Components and Services

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	H.323	SIP	MGCP
Common Control Components	Gatekeeper	Proxy Server, Redirect Server, Location Server, Registrar Server	Call Agent
Endpoints	Gateway, Terminal	Client (IP Telephone, Gateway)	Media Gateway (or Gateway)
Call Administration and Accounting	Gateway, Gatekeeper	Gateway	Call Agent
Call Status	Gateway, Gatekeeper	Gateway	Call Agent
Address Management	Gatekeeper	Location Server, Registrar Server	Call Agent
Admission Control	Gatekeeper	Not Supported	Call Agent

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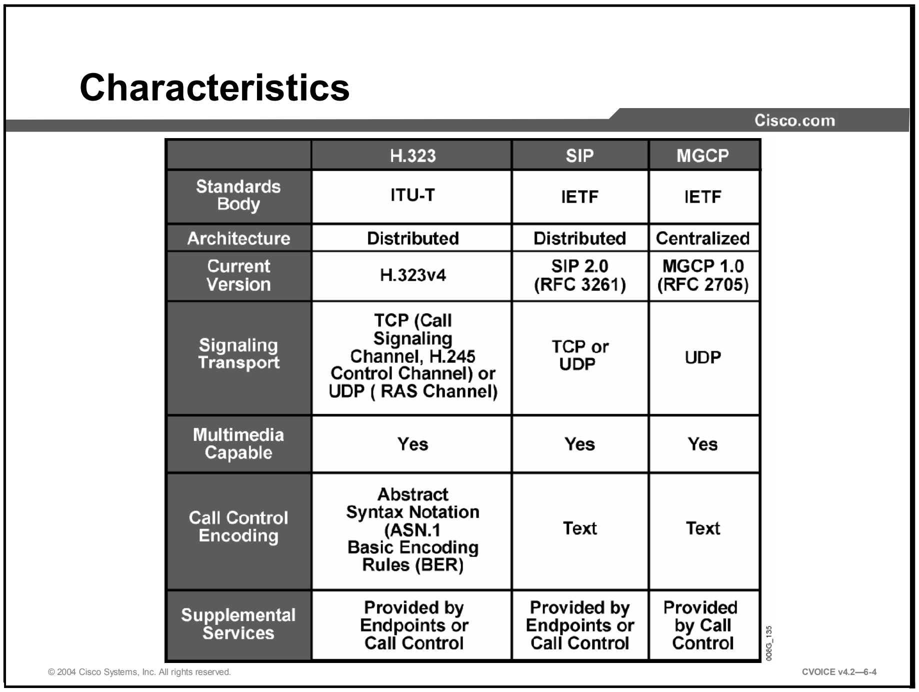
In a generic model, the components of signaling and call control are identified as common control components and endpoints. Common control components provide a set of optional services: call administration and accounting, call status, address management, and admission control. The chart in the figure identifies how the basic components of the generic model are configured in H.323, SIP, and MGCP, and, if applicable, where optional services are provided.



Characteristics

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	H.323	SIP	MGCP
Standards Body	ITU-T	IETF	IETF
Architecture	Distributed	Distributed	Centralized
Current Version	H.323v4	SIP 2.0 (RFC 3261)	MGCP 1.0 (RFC 2705)
Signaling Transport	TCP (Call Signaling Channel, H.245 Control Channel) or UDP (RAS Channel)	TCP or UDP	UDP
Multimedia Capable	Yes	Yes	Yes
Call Control Encoding	Abstract Syntax Notation (ASN.1 Basic Encoding Rules (BER)	Text	Text
Supplemental Services	Provided by Endpoints or Call Control	Provided by Endpoints or Call Control	Provided by Call Control

CVOICE v4.2—6-4

Note	As of version 3 of H.323, the calling signaling channel and the H.245 control channel can be UDP-based.

The chart in the figure compares several factors that can influence your decision to select H.323, SIP, or MGCP.

Standards Bodies (ITU-T vs. IETF)

The two originating authorities for the model may seem to have little relevance. However, the ITU-T and the IETF work under different conditions, a fact which has an impact on the results and the speed of their work.

Although the ITU-T is older than the IETF, it is associated with a publishing cycle and consensus process that is often blamed for delay. However, its rigorous procedures result in mature recommendations with the consistent use of language and terminology. The consensus process requires a high level of agreement and is generally accepted as the preferred way to proceed internationally.

Without being subject to the rigors of the ITU-T procedures and policies, the IETF can respond quickly to user demands, although the solutions can be less mature than those created by the ITU-T.

Knowing which standards body is involved provides a sense of the standards development process, the pace of work, and the quality of results.

Architecture (Centralized vs. Distributed)

The distinction between the centralized architecture and the distributed architecture can influence which model you choose.

Current Version

The current version of a specification or recommendation is an indication of its maturity.

Signaling Transport (TCP vs. UDP)

Understanding the underlying transport of the signaling channels helps to explain the performance and overheads of the relationship between H.323, SIP, or MGCP components. Connectionless, UDP-based relationships must shift reliability and sequencing into the application, making them more complex. Both reliability and sequencing are built into TCP. However, UDP-based applications are designed to respond more quickly than TCP-based applications. This is significant, for example, during call setup.

Multimedia Capability (Yes or No)

The ability to transport information of different types, such as audio, video, and data, can be a determining factor in choosing between H.323, SIP, or MGCP.

Call Control Encoding (ASN.1 vs. Text)

Traditionally, the ITU-T and the IETF have proposed different methods of encoding the information that travels between endpoints.

It is generally accepted that applications using text-based encoding are easier to encode, decode, and troubleshoot, compared to Abstract Syntax Notation One (ASN.1)-based encoding, which is more compact and efficient.

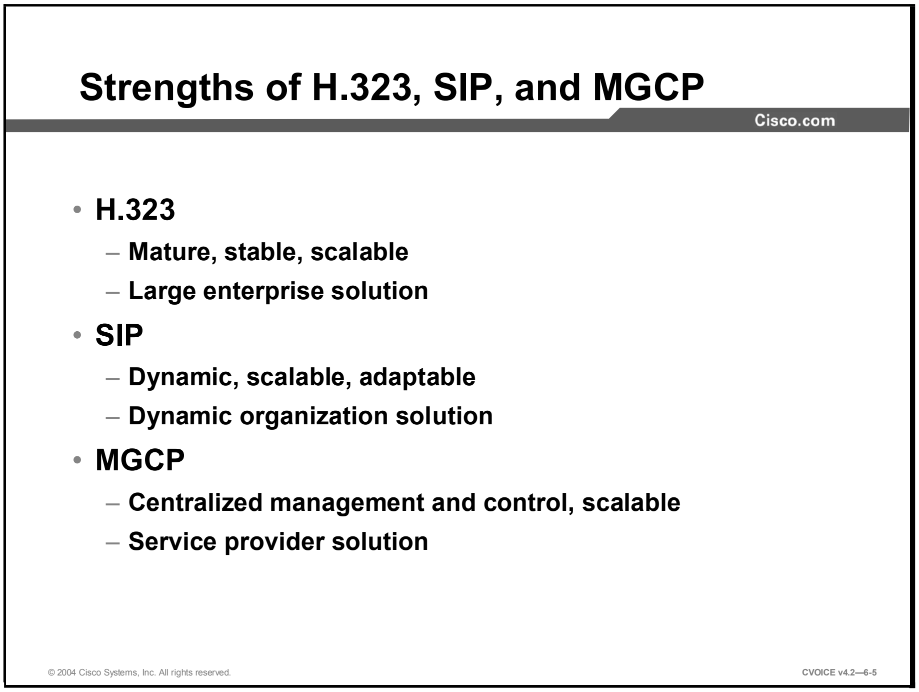
Supplementary Services (Endpoint vs. Call Control)

Where and how you introduce supplementary services can be important considerations in a comparison of H.323, SIP, or MGCP..

Services deployed throughout the network are easily implemented centrally in a call control component. Services with regional relevance can be implemented effectively in the endpoints.

Strengths of H.323, SIP, and MGCP

Because there are several different telecommunication environments, more than one choice for signaling and call control is necessary. This topic looks at the strengths of H.323, SIP, and MGCP, and suggests the type of environment that best suits each call control model.



H.323

H.323, which has been the only viable option in VoIP signaling and call control solutions for a long period of time, is mature and attracts supporters. Consequently, H.323 products are widely available and deployed extensively.

When properly designed, H.323 is both scalable (accommodates the implementation of large distributed networks) and adaptable (allows for the introduction of new features). The H.323 call control model works well for large enterprises because the gatekeeper-centralized call control provides some capability for Operation, Administration, and Maintenance (OA&M).

SIP

SIP is a multimedia protocol that uses the architecture and messages that are found in popular Internet applications. By using a distributed architecture—with URLs for naming and text-based messaging—SIP takes advantage of the Internet model for building VoIP networks and applications.

SIP is a protocol that is used in a distributed architecture and allows companies to build large-scale networks that are scalable, resilient, and redundant. SIP provides mechanisms for interconnecting with other VoIP networks and for adding intelligence and new features on the endpoints, SIP proxy, or redirect servers.

Although the IETF is progressive in defining extensions that allow SIP to work with legacy voice networks, the primary motivation behind SIP is to create an environment that supports next-generation communication models that utilize the Internet and Internet applications. In addition, the lack of centralized management support makes SIP more suitable for growing, dynamic organizations and Internet telephony service providers.

MGCP

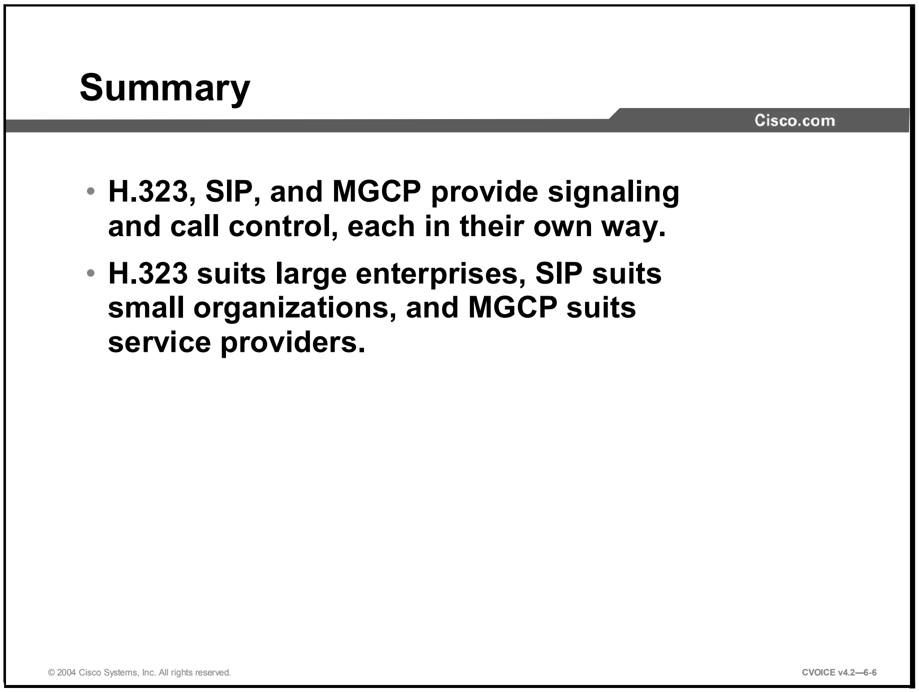
MGCP describes an architecture in which call control and services such as OA&M are centrally added to a VoIP network. As a result, MGCP architecture closely resembles the existing PSTN architecture and services.

In a centralized architecture, MGCP allows companies to build large-scale networks that are scalable, resilient, and redundant. MGCP provides mechanisms for interconnecting with other VoIP networks and adding intelligence and features to the call agent.

MGCP works well for organizations that are comfortable with centralized management and control; for example, service providers are well suited for MGCP.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to these resources:

- ITU-T Recommendation H.323 (Version 4)
  http://www.itu.int/rec/recommendation.asp?type=items&lang=E&parent=T-REC-H.323-200011-S
- IETF RFC 3261: *SIP: Session Initiation Protocol*
  http://www.faqs.org/rfcs/rfc3261.html
- IETF RFC 2705: *Media Gateway Control Protocol (MGCP), version 1.0*
  http://www.faqs.org/rfcs/rfc2705.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which call control model's encoding is more compact but harder to decode and troubleshoot?

A) H.323

B) SGCP

C) SIP

D) MGCP

Q2) Match the type of endpoint to its call control model. A call control model can be used more than once.

A) client

B) terminal

C) gateway

D) media gateway

_____ 1. H.323

_____ 2. SIP

_____ 3. MGCP

Q3) Which call control model most closely resembles the PSTN?

A) H.323

B) SGCP

C) SIP

D) MGCP

Q4) Which call control model is popular in large enterprises because of its maturity and stability?

A) H.323

B) SGCP

C) SIP

D) MGCP

Q5) Match the common control component with the call control model. A common control component may be used more than once.

A) call agent

B) gatekeeper

C) proxy server

D) redirect server

E) location server

F) registrar server

_____ 1. MGCP

_____ 2. SIP

_____ 3. H.323

Q6) Match the signaling transport method with the call control model.

A) UDP

B) TCP

C) UDP and TCP

D) UDP or TCP

_____ 1. H.323

_____ 2. SIP

_____ 3. MGCP

Q7) Which two components are responsible for address management in the SIP call control model? (Choose two.)

A) proxy server

B) location server

C) registrar server

D) redirect server

Q8) Match the method of supplemental service control with the call control model.

A) provided by end points or call control

B) provided by end points

C) provided by call control

_____ 1. H.323

_____ 2. SIP

_____ 3. MGCP

Quiz Answer Key

Q1) A

Relates to: Feature Comparison Charts

Q2) 1-C or B, 2-A, 3-C or D

Relates to: Feature Comparison Charts

Q3) D

Relates to: Strengths of H.323, SIP, and MGCP

Q4) A

Relates to: Strengths of H.323, SIP, and MGCP

Q5) 1-A, 2-C, D, E, or F, 3-A and B

Relates to: Feature Comparison Charts

Q6) 1-C, 2-D, 3-A

Relates to: Feature Comparison Charts

Q7) B and C

Relates to: Feature Comparison Charts

Q8) 1-A, 2-A, 3-C

Relates to: Feature Comparison Charts

Module 7

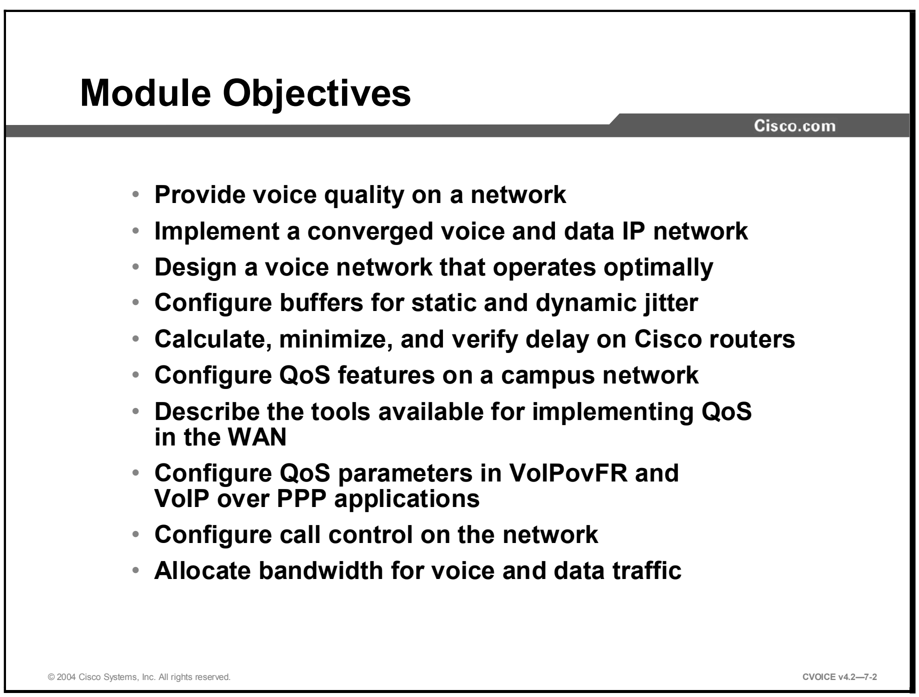
Improving and Maintaining Voice Quality

Overview

When human speech is converted to analog electrical signals and then digitized and compressed, some of the qualitative components are lost. This module explores the components of voice quality that you must maintain, the methods that you can use to measure voice quality, and the effective quality of service (QoS) tools that you can implement in a network to improve voice quality.

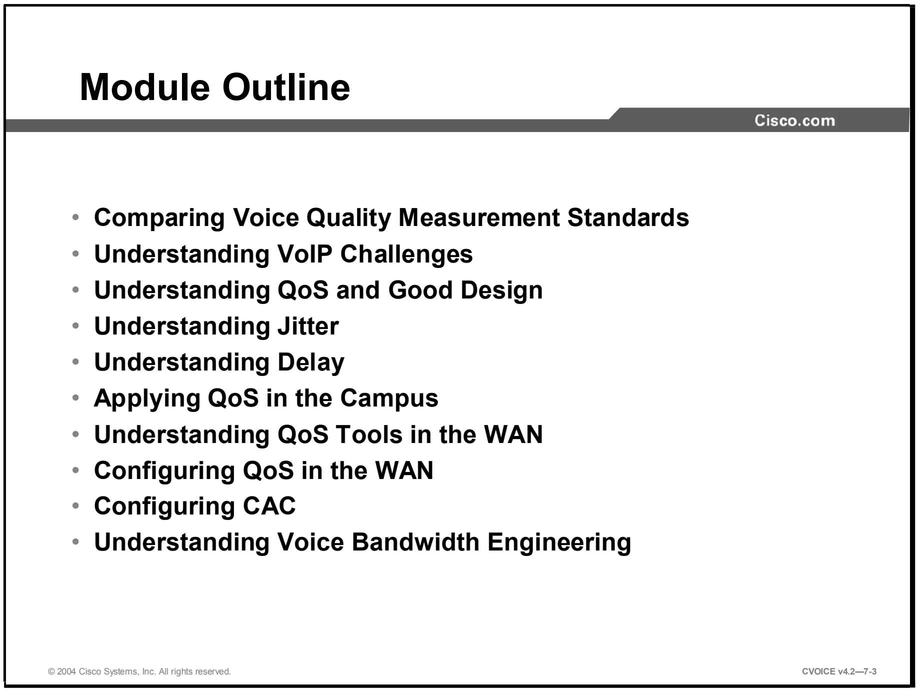
Module Objectives

Upon completing this module, you will be able to describe specific voice quality issues and the QoS solutions used to solve them.



Module Outline

The outline lists the components of this module.



Comparing Voice Quality Measurement Standards

Overview

As technology invents and reinvents new ways to electronically represent the human voice, a method to compare the quality of each representation is necessary. Three quality measurement techniques are MOS, PESQ, and PSQM. In this lesson, you will learn the details of these three measurement techniques and the scores attained with different compression methods.

Relevance

To understand more about the tools that you must use to improve voice QoS, you must be knowledgeable about how voice quality is measured and how it is affected.

Objectives

Upon completing this lesson, you will be able to provide voice quality on a network. This includes being able to meet these objectives:

- List the attributes that affect audio clarity
- Describe the psychological comfort factors that affect voice quality
- State the purpose of MOS and PSQM and the methods used to calculate them
- State the purpose of PESQ and the method used to calculate it
- Describe how the codec voice quality scores differ and how they are measured

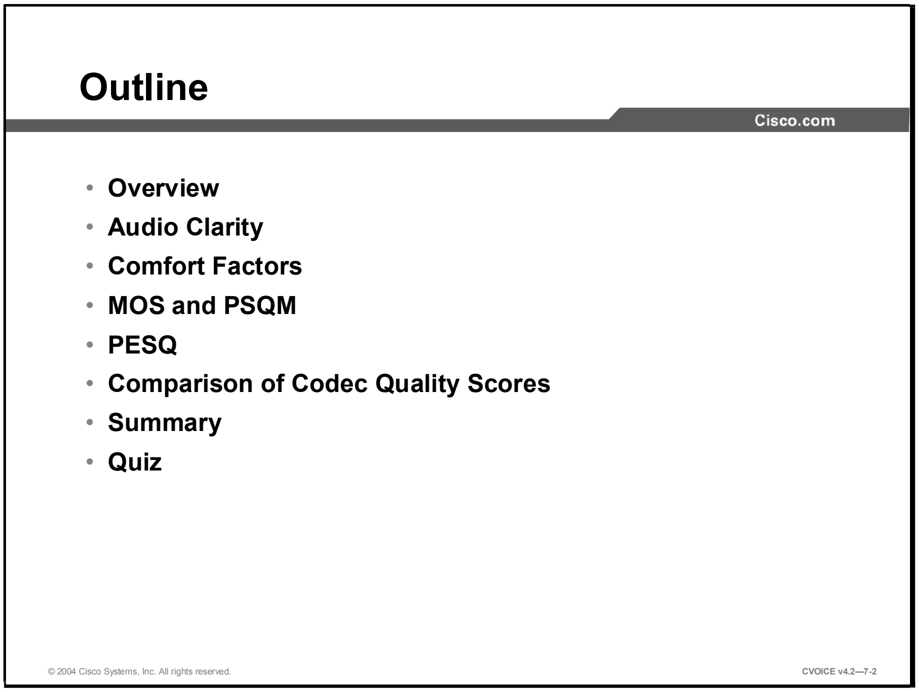
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of voice telephony systems, including traditional and Voice over IP (VoIP) systems
- Knowledge of Cisco IOS software

Outline

The outline lists the topics included in this lesson.



Audio Clarity

The effectiveness of a telephone conversation depends on its clarity. If the conversation does not sound good, the listener is annoyed and the speaker is unable to express the message. The clarity of the conversation must be maintained end-to-end, from the speaker to the listener. This topic lists the factors affecting audio clarity.

Factors Affecting Audio Clarity

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- **Fidelity (transmission bandwidth versus original)**
- **Echo**
- **Delay**
- **Delay variation (jitter)**

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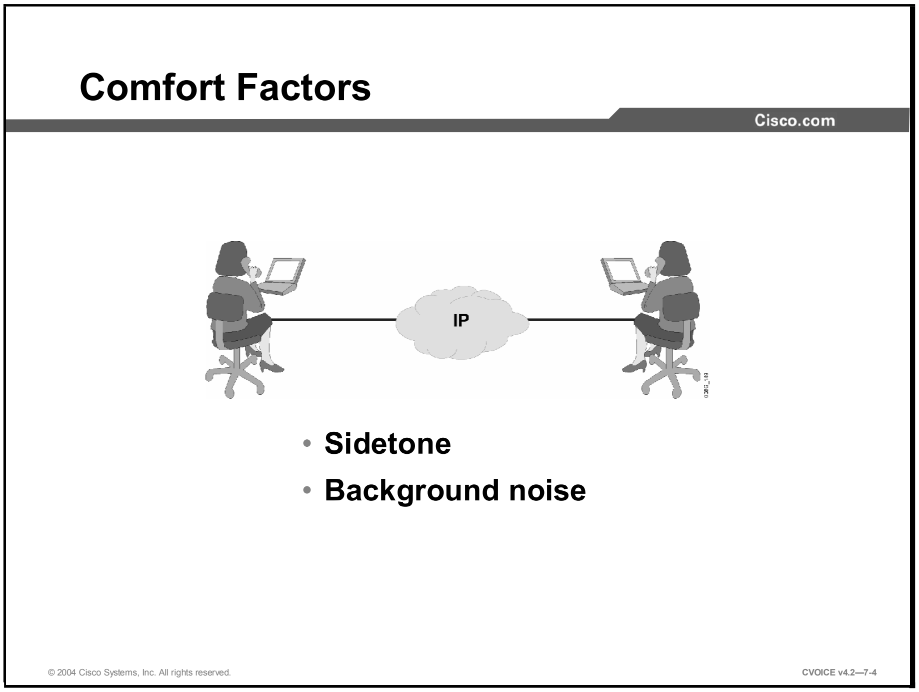
The clarity, or cleanliness and crispness, of the audio signal is of utmost importance. The listener should recognize the identity of the speaker and sense the mood. Factors that can affect clarity include:

- **Fidelity:** Consistency of transmission bandwidth to the original bandwidth. The bandwidth of the transmission medium almost always limits the total bandwidth of the spoken voice. Human speech typically requires a bandwidth from 100 to 10,000 Hz, although 90 percent of speech intelligence is contained between 100 and 3000 Hz.
- **Echo:** A result of electrical impedance mismatches in the transmission path. Echo is always present. The two components that affect echo are amplitude (loudness of the echo) and delay (the time between the spoken voice and the echoed sound). You can control echo using suppressors or cancellers.
- **Delay:** The time between the spoken voice and the arrival of the electronically delivered voice at the far end. Delay is affected by a number of factors, including distance (propagation delay), coding, compression, serialization, and buffers.
- **Delay variation:** Because of the nature of an IP delivery network, the arrival of coded speech at the far end of a Voice over IP (VoIP) network can vary. The varying arrival time of the packets can cause gaps in the re-creation and playback of the voice signal. These gaps are undesirable and cause the listener great annoyance. Delay is induced in the network by variation in the routes of individual packets, contention, or congestion. You can solve variable delay by using dejitter buffers.

Comfort Factors

In addition to audio clarity, a number of psychological comfort factors affect the perceived quality of voice. Together these factors can cause the listener to rate the overall quality as good or poor. This topic describes these psychological comfort factors.

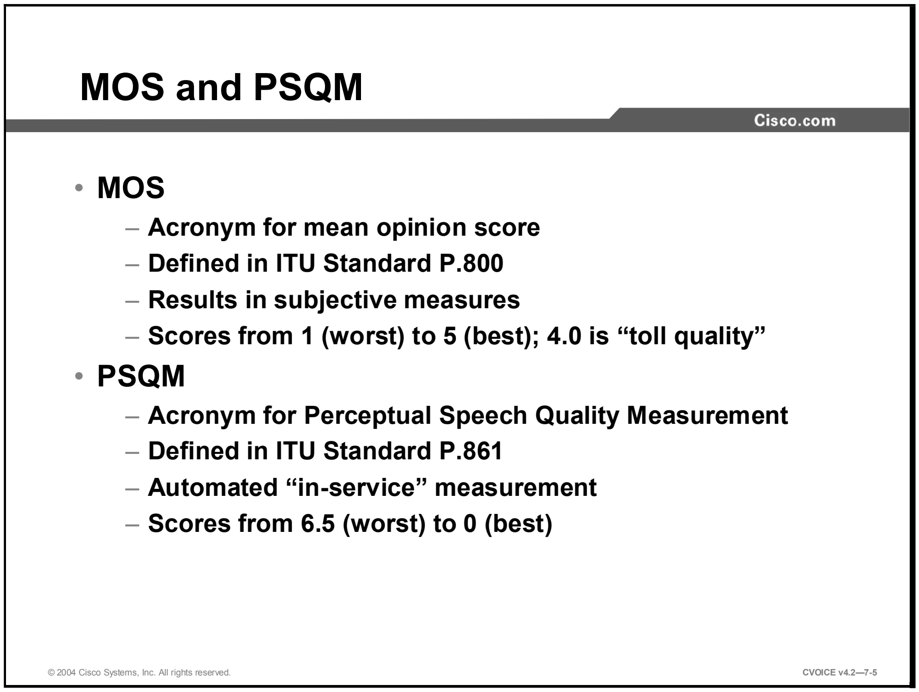


When transporting VoIP, the perceived quality of the received voice must mimic the traditional public switched telephone network (PSTN). There should be no perceived differences. The following two comfort factors affect quality:

- **Sidetone:** The purposeful design of the telephone that allows the speaker to hear the spoken audio in the earpiece. Without sidetone, the speaker is left with the impression that the telephone instrument is not working.
- **Background noise:** The low-volume audio that is heard from the far-end connection. Certain bandwidth-saving technologies can eliminate background noise altogether, such as voice activity detection (VAD). When this technology is implemented, the speaker audio path is open to the listener, while the listener audio path is closed to the speaker. The effect of VAD is often that the speaker thinks that the connection is broken because nothing is heard in the earpiece from the far end.

MOS and PSQM

This topic explains calculating mean opinion score (MOS) and using it to compare voice quality. It also explains the calculation of Perceptual Speech Quality Measurement (PSQM) scores.



MOS is a scoring system for voice quality. An MOS score is generated when listeners evaluate prerecorded sentences that are subject to varying conditions, such as compression algorithms. Listeners then assign the sentences values, based on a scale from 1 to 5, where 1 is the worst and 5 is the best. The sentence used for English-language MOS testing is, "Nowadays, a chicken leg is a rare dish." This sentence is used because it contains a wide range of sounds found in human speech, such as long vowels, short vowels, hard sounds, and soft sounds.

The test scores are then averaged to a composite score. The test results are subjective, because they are based on the opinions of the listeners. The tests are also relative, because a score of 3.8 from one test cannot be directly compared to a score of 3.8 from another test. Therefore, you must establish a baseline for all tests, such as using G.711 as a baseline, so that the scores can be normalized and compared directly.

PSQM is an automated method of measuring speech quality "in-service," or as it happens. PSQM software usually resides with IP call management systems, which are sometimes integrated into Simple Network Management Protocol (SNMP) systems. This topic describes PSQM calculations.

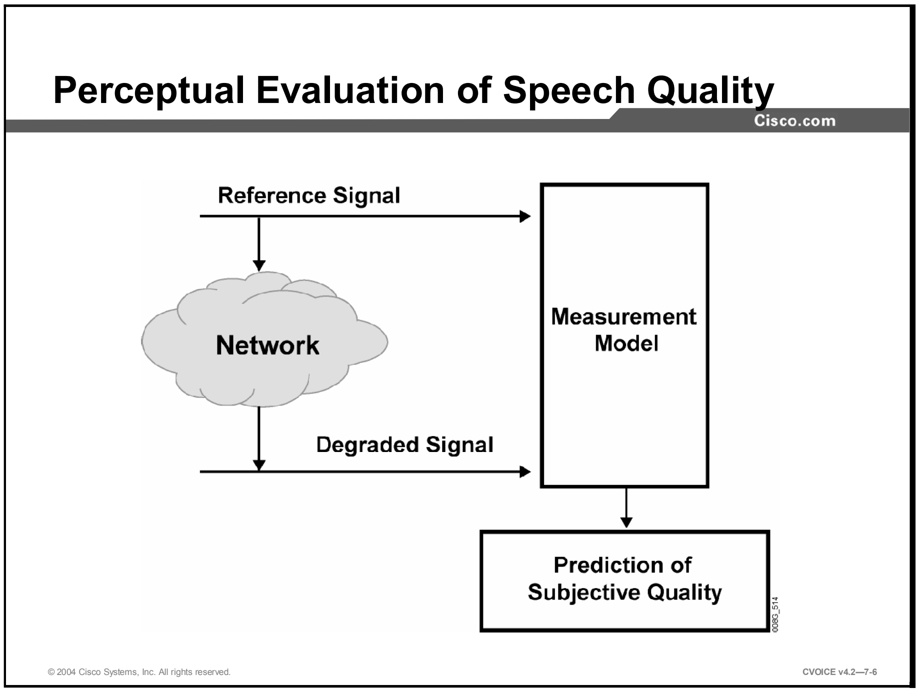
Equipment and software that can measure PSQM is available through third-party vendors; it is not implemented in Cisco Systems equipment. The measurement is made by comparing the original transmitted speech to the resulting speech at the far end of the transmission channel. PSQM systems are deployed as in-service. The PSQM measurements are made during real conversation on the network. This automated testing algorithm has over 90 percent accuracy compared to the actual listening tests, such as MOS. Scoring is based on a scale from 0 to 6.5, where 0 is the best and 6.5 is the worst. Originally designed for circuit-switched voice, PSQM does not take into account jitter or delay problems, which are experienced most in packet-switched voice systems.

Example: MOS and PSQM in VoIP Networks

MOS and PSQM are not recommended for today's VoIP networks. Each was originally designed before VoIP, and the typical problems, such as jitter and delay, associated with VoIP are not measured. For example, it is possible to obtain an MOS score of 3.8 on a VoIP network, when the one-way delay exceeds 500 milliseconds. This is because the MOS evaluator has no concept of a two-way conversation and only listens to audio quality. The one-way delay is not evaluated.

PESQ

This topic describes the Perceptual Evaluation of Speech Quality (PESQ) voice quality measurement calculation.



PESQ was originally developed by British Telecom, Psytechnics, and KPN Research of the Netherlands. It has earned ITU Standard P.862 and is considered the current standard for voice quality measurement. PESQ can take into account coder-decoder (codec) errors, filtering errors, jitter problems, and delay problems typical in VoIP. It combines the best of the PSQM method along with a method called Perceptual Analysis Measurement System (PAMS). PESQ scores range from 1 (worst) to 4.5 (best), with 3.8 considered toll quality. It should be stated here that PESQ is meant to measure only one aspect of the voice quality. The effects of two-way communication, such as loudness loss, delay, echo, and sidetone, are not reflected in PESQ scores.

Example: PESQ Applied

Many equipment vendors offer PESQ measurement systems. Such systems are either stand-alone or plug in to existing network management systems. PESQ was designed to mirror the MOS measurement system, so if a score of 3.2 was measured by PESQ, a score of 3.2 should be achieved using MOS methods.

Comparison of Codec Quality Scores

This topic provides a relative comparison of scores from various coding and compression schemes.

MOS Rating of Digital Voice

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Codec	Compress Technique	Bit Rate (kbps)	MIPS	Compress Delay	Framing Size	Mean Opinion Score
G.711	PCM	64	0.34	0.75	0.125	4.1
G.726	ADPCM	32	13	1	0.125	3.85
G.728	LDCELP	16	33	3.5	0.625	3.61
G.729	CS-ACELP	8	20	10	10	3.92
G.729A	CS-ACELP	8	10.5	10	10	3.9
G.723.1	MPMLQ	6.3	16	30	30	3.9
G.723.1	ACELP	5.3	16	30	30	3.8

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The International Telecommunication Union (ITU) conducted MOS testing to develop formulas for rating standards G.711 through G.729. The results are summarized in the table in the figure.

MOS Under Varying Conditions

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Example: G.729	MOS Rating
Average speech level	3.92
Low input level	3.54
Two tandem codings	3.46
Three tandem codings	2.68
5% bit error rate	3.24
5% frame error rate	3.02

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The table in the figure demonstrates how the MOS rating for the codec quality score G.729 is affected by various network conditions. With an average speech level, the MOS score will be high. When a user speaks too softly, the score is lower, due to mismatches in the G.729 code book. With tandem coding, the score gets progressively worse. Tandem coding is conversion from analog to digital, back to analog, then to digital again, and back to analog once more. The result is analogous to making a copy of a copy of a videotape—the quality gets worse and worse. Tandem codings should be avoided in VoIP networks. Finally, with bit and frame errors, the MOS suffers due to incorrect or missing packets.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- The factors that affect audio clarity are fidelity, echo, delay, and jitter.
- Comfort factors, such as sidetone and background noise, must be maintained when transporting VoIP.
- MOS is a scoring system for voice quality that produces subjective and relative test results.
- PSQM is an automated method used to measure in-service speech quality.
- ITU developed formulas to rate codec standards; these ratings can be affected by various network conditions.

CVOICE v4.2—7-9

References

For additional information, refer to these resources:

- PESQ information
  http://www.pesq.org
- Cisco PESQ article
  Robison, Helen M. *Designing Voice Quality in Hybrid TDM-IP Networks*
  http://www.cisco.com/en/US/about/ac123/ac114/ac173/ac225/about_cisco_packet_technology0900aecd800c9140.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which two factors contribute to delay variation in voice transmission? (Choose two.)

A) congestion on the network

B) electrical impedance mismatches

C) compression of packets

D) inconsistency in bandwidth

E) varying routes of packets

Q2) Which two factors affect voice clarity? (Choose two.)

A) fidelity

B) echo

C) sidetone

D) background noise

E) distance

Q3) Which two psychological comfort factors contribute to the quality of voice being perceived as good? (Choose two.)

A) silence suppression

B) buffering

C) sidetone

D) background noise

E) ring cadence

Q4) What is the disadvantage of using VAD on a call?

A) The cost of the call may increase.

B) The listener may hear unwanted noise.

C) The bandwidth requirement may increase.

D) The speaker may think that the connection is broken.

Q5) Which voice quality scoring system is based on subjective evaluation?

A) MOS

B) PCM

C) PSQM

D) YBTM

Q6) What is the accuracy of PSQM compared to other listening tests?

A) over 50 percent

B) over 65 percent

C) over 90 percent

D) over 99.9 percent

Q7) In PESQ measurements, which score is considered toll quality?

A) 3.1

B) 3.8

C) 4.0

D) 6.5

Q8) PESQ is defined as which ITU standard?

A) P.800

B) P.861

C) P.862

D) G.711

Q9) Which codec scores highest in MOS testing?

A) G.711

B) G.723.1

C) G.726

D) G.728

E) G.729

Q10) According to MOS tests, which network condition distorts voice the least?

A) 5 percent bit error rate

B) 5 percent frame error rate

C) two tandem codings

D) three tandem codings

E) low input level

Quiz Answer Key

Q1) A, E

Relates to: Audio Clarity

Q2) A, B

Relates to: Audio Clarity

Q3) C, D

Relates to: Comfort Factors

Q4) D

Relates to: Comfort Factors

Q5) A

Relates to: MOS and PSQM

Q6) C

Relates to: MOS and PSQM

Q7) B

Relates to: PESQ

Q8) C

Relates to: PESQ

Q9) A

Relates to: Comparison of Codec Quality Scores

Q10) E

Relates to: Comparison of Codec Quality Scores

Understanding VoIP Challenges

Overview

Because of the inherent characteristics of a converged voice and data IP network, administrators face certain challenges in delivering voice traffic correctly. This lesson describes these challenges and offers solutions for avoiding and overcoming them.

Relevance

Administrators of converged networks must consider and overcome technical challenges that hinder the proper delivery of voice packets. Network administrators must complete and understand this lesson before they can successfully implement a converged voice and data IP network.

Objectives

Upon completing this lesson, you will be able to implement a converged voice and data IP network. This includes being able to meet these objectives:

- Name some of the inherent problems that occur when delivering voice in IP networks
- Describe the cause of jitter and the solution used to eliminate it from an IP network
- List the two types of packet delay and the solution that is used to eliminate packet delay from an IP network
- State the cause of packet loss and its effect on voice quality
- Describe the contention issues associated with transmitting voice, data, and video on the same outbound interface
- Describe the method that is used to sequence voice packets
- Describe the effect of circuit reliability and availability on voice quality

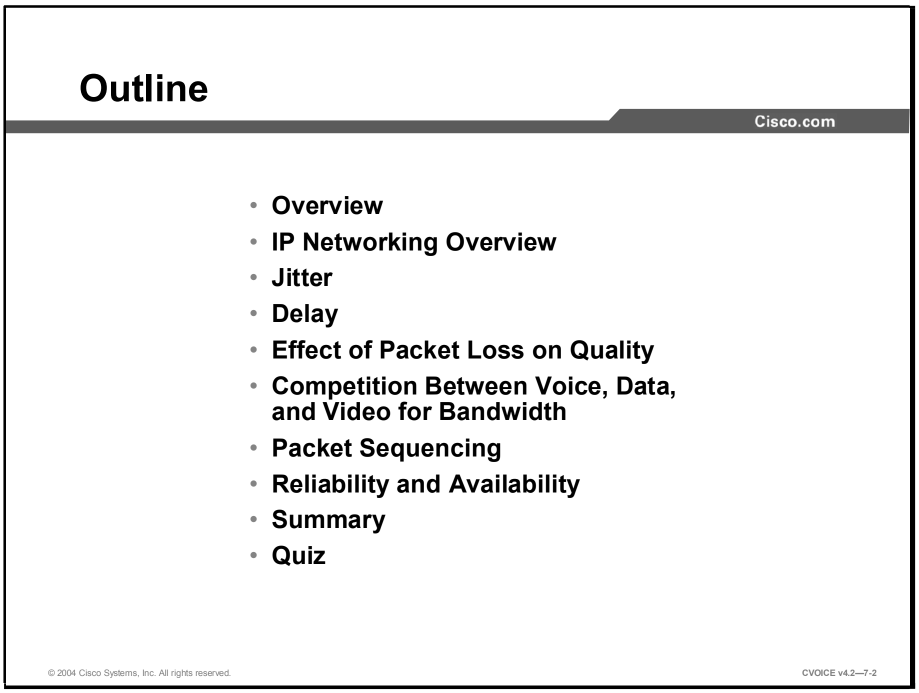
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of VoIP basics
- Knowledge of TCP/IP networks, including routing, User Datagram Protocol (UDP), and best-effort concepts
- Knowledge of traditional telephony networks

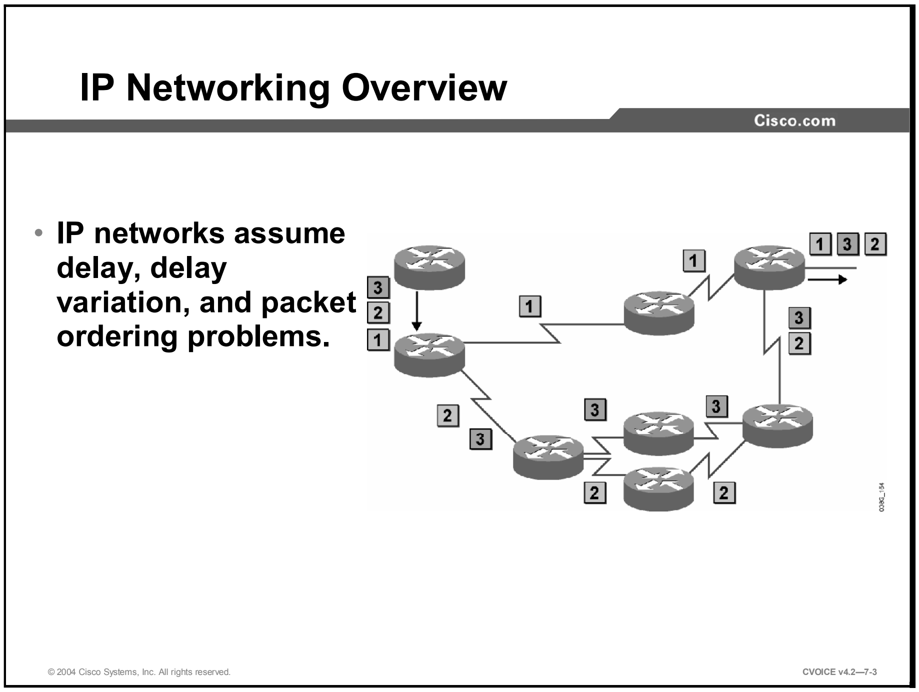
Outline

The outline lists the topics included in this lesson.



IP Networking Overview

This topic provides an overview of IP networking and some of the inherent challenges when conveying voice over an IP network.



IP is a connectionless network protocol. Connectionless networks generally do not participate in signaling. The concept of session establishment exists between end systems, although the connectionless network remains unaware of the virtual circuit (VC).

IP resides at the network layer of the Open System Interconnection (OSI) protocol stack. Therefore, it can transport IP packets over deterministic and nondeterministic Layer 2 protocols, such as Frame Relay or ATM. IP can be used to communicate across any set of interconnected networks and is equally suited to both LAN and WAN communication.

IP information is transferred in a sequence of datagrams. A message is sent as a series of datagrams that are reassembled into the completed message at the receiving location. Because a voice conversation that is transported in IP can be considered a continuous audio file, all packets must be received in sequence immediately and without interpacket variable delay.

Traditionally, IP traffic transmits on a FIFO basis. Different packet types vary in size, allowing large file transfers to take advantage of the efficiency that is associated with larger packet sizes. FIFO queuing affects the way that voice packets transmit, causing delay and delay variation at the receiving end.

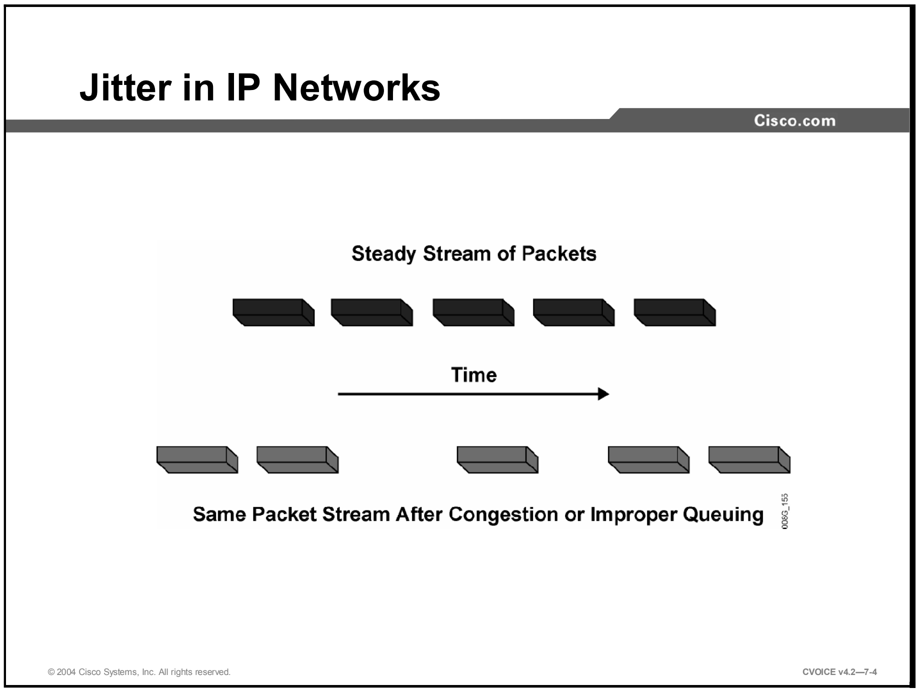
UDP is the connectionless transport layer protocol used for VoIP. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery. UDP requires that other protocols handle error processing and retransmission. The figure shows how packets may be received out of sequence or become completely lost at the receiving end.

Example: IP Networking

Due to the very nature of IP networking, voice packets sent across IP will be subject to certain transmission problems. These problems include jitter, delay, and packet ordering. In the figure, packets sent from the originating router on the left are in sequence and sent with predictable transmission intervals. As they traverse the IP network, the routing protocol may send some of the packets through one path, while other packets traverse a different path. As the packets arrive at the destination router on the right, they arrive with varying delays and out of sequence. These problems must be addressed with QoS mechanisms explained further in this lesson.

Jitter

This topic describes the occurrence of jitter in IP networks and the Cisco Systems solution to this problem.



Jitter is defined as a variation in the delay of received packets. On the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Because of network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant, as displayed in the figure.

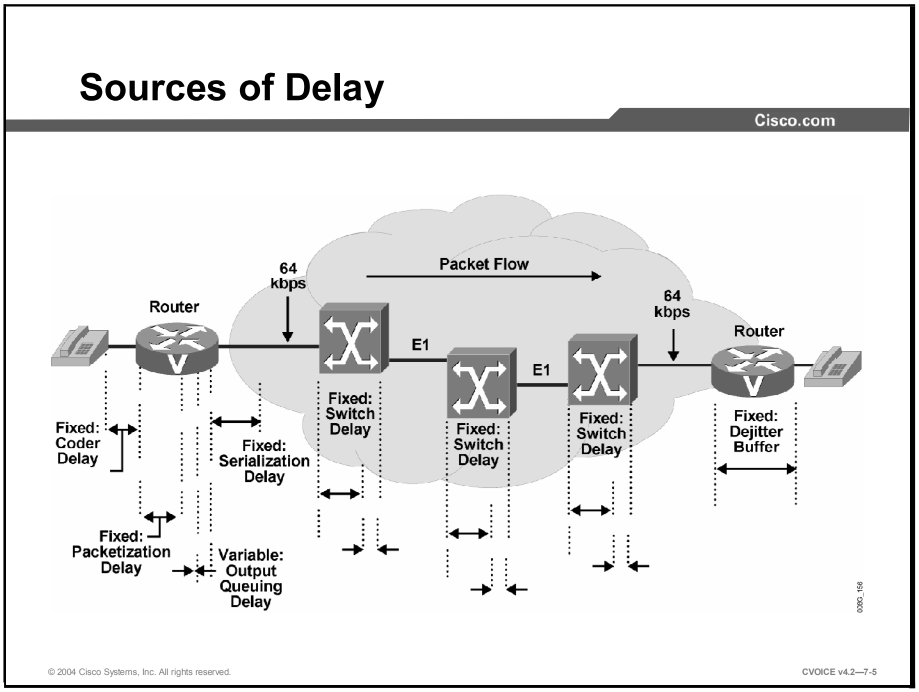
When a router receives an audio stream for VoIP, it must compensate for the jitter that is encountered. The mechanism that handles this function is the playout delay buffer, or dejitter buffer. The playout delay buffer must buffer these packets and then play them out in a steady stream to the digital signal processors (DSPs) to be converted back to an analog audio stream. The playout delay buffer, however, affects overall absolute delay.

Example: Jitter in Voice Networks

When a conversation is subjected to jitter, the results can be clearly heard. If the talker says, “Watson, come here. I want you,” the listener might hear, “Wat….s…on…….come here, I……wa……nt……..y……ou.” The variable arrival of the packets at the receiving end causes the speech to be delayed and garbled.

Delay

Overall or absolute delay can affect VoIP. You might have experienced delay in a telephone conversation with someone on a different continent. The delays can be very frustrating, causing words in the conversation to be cut off. This topic describes the causes of packet delay and the Cisco solution to this problem.



When you design a network that transports voice over packet, frame, or cell infrastructures, it is important to understand and account for the delay components in the network. You must also correctly account for all potential delays to ensure that overall network performance is acceptable. Overall voice quality is a function of many factors, including the compression algorithm, errors and frame loss, echo cancellation, and delay.

There are two distinct types of delay:

- Fixed-delay components add directly to the overall delay on the connection.
- Variable delays arise from queuing delays in the egress trunk buffers that are located on the serial port that is connected to the WAN. These buffers create variable delays, called jitter, across the network.

Acceptable Delay: G.114

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Range in Milliseconds	Description
0 to 150	Acceptable for most user applications
150 to 400	Acceptable, provided that administrators are aware of the transmission time and its impact on the transmission quality of user applications
Above 400	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded

 CVOICE v4.2—7-6

The ITU considers network delay for voice applications in Recommendation G.114. This recommendation defines three bands of one-way delay, as shown in the table in the figure.

Note This recommendation is for connections with echo that are adequately controlled, implying that echo cancellers are used. Echo cancellers are required when one-way delay exceeds 25 ms (G.131).

This recommendation is oriented toward national telecommunications administrations, and therefore is more stringent than recommendations that would normally be applied in private voice networks. When the location and business needs of end users are well known to a network designer, more delay may prove acceptable. For private networks, a 200-ms delay is a reasonable goal and a 250-ms delay is a limit. This goal is what Cisco Systems proposes as reasonable as long as jitter does not impact voice quality. However, all networks must be engineered so that the maximum expected voice connection delay is known and minimized.

Example: Acceptable Delay

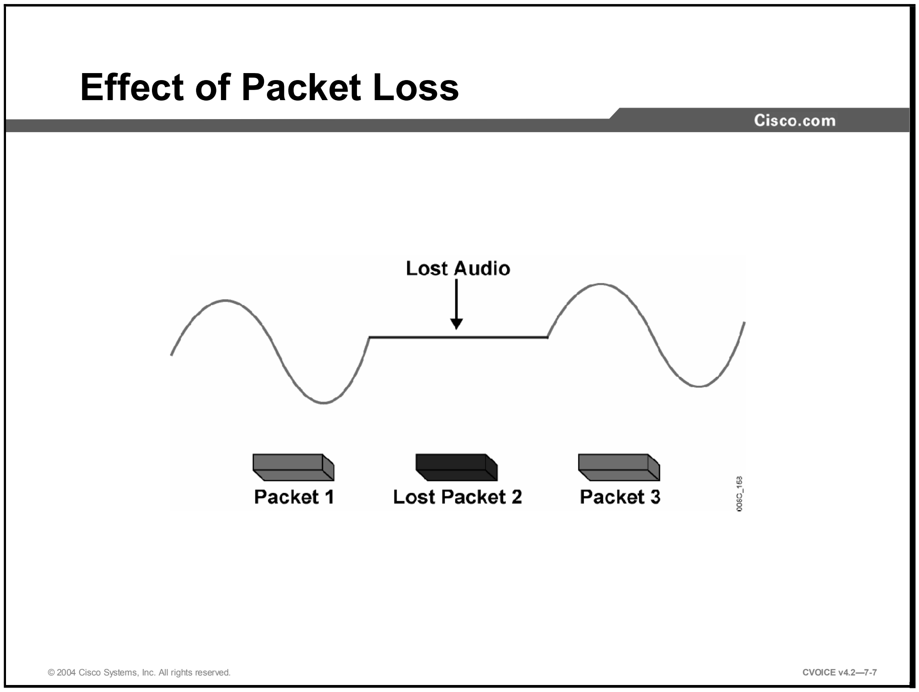
The G.114 recommendation is for one-way delay only and does not account for round-trip delay. Network design engineers must consider all delays, variable and fixed. Variable delays include queuing and network delays, while fixed delays include coder, packetization, serialization, and dejitter buffer delays. The table is an example of calculating delay budget.

Calculating Delay Budget

Delay Type	Fixed (ms)	Variable (ms)
Coder delay	18	
Packetization delay	30	
Queuing and buffering		8
Serialization (64 kbps)	5	
Network delay (public frame)	40	25
Dejitter buffer	45	
Totals	138	33

Effect of Packet Loss on Quality

Lost data packets are recoverable if the endpoints can request retransmission. Lost voice packets are *not* recoverable, because the audio must be played out in real time and retransmission is not an option. This topic describes the causes and effects of lost voice packets.



Voice packets might be dropped under the following conditions:

- The network quality is poor
- The network is congested
- There is too much variable delay in the network

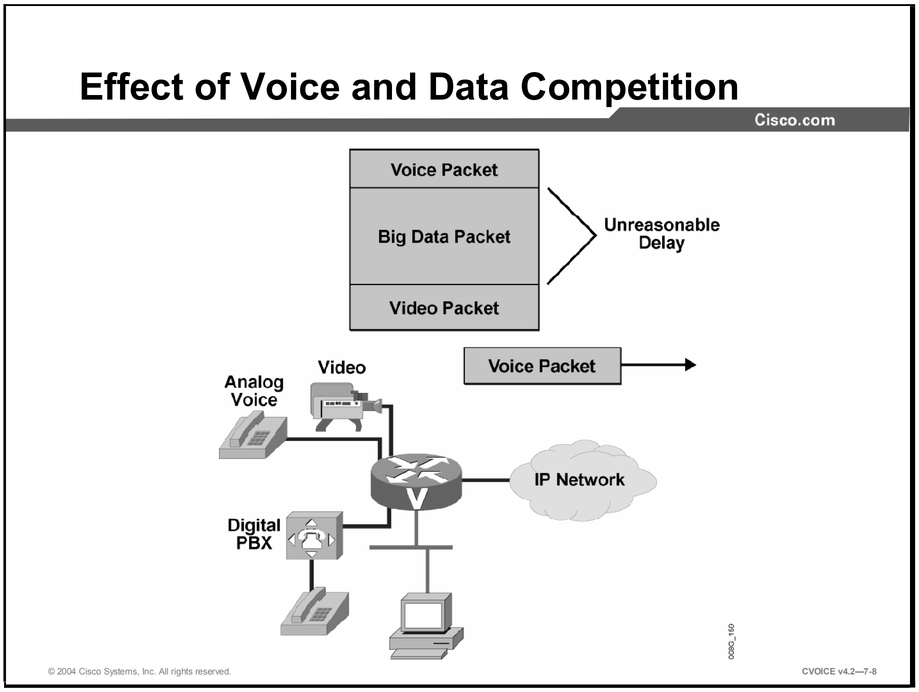
Packet loss causes voice clipping and skips. As a result, the listener hears gaps in the conversation, as shown in the figure. The industry standard codec algorithms that are used in Cisco DSPs will correct for 20–50 ms of lost voice. Cisco VoIP technology uses 20-ms samples of voice payload per VoIP packet by default. Effective codec correction algorithms require that only a single packet can be lost at any given time. If more packets are lost, the listener experiences gaps.

Example: Packet Loss in Voice Networks

If a conversation experiences packet loss, the effect is immediately heard. If the talker says, “Watson, come here. I want you,” the listener might hear, “Wat----, come here, ------you.”

Competition Between Voice, Data, and Video for Bandwidth

In a converged network, voice and data packets compete with each other for transmission on the outbound interface. This topic describes the effects of this competition and offers solutions.



When data, voice, and video are placed in the same output queue, voice can suffer delay and delay variation. The time it takes to transmit large data packets can cause voice problems that result in communication gaps. Also, the demands of video on the available bandwidth can easily limit the number of voice packets sent.

Serialization Delays

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Frame Size

Link Speed	1 Byte	64 Byte	128 Byte	256 Byte	512 Byte	1024 Byte	1500 Byte
56 kbps	143 µs	9 ms	18 ms	36 ms	72 ms	144 ms	214 ms
64 kbps	125 µs	8 ms	16 ms	32 ms	64 ms	128 ms	187 ms
128 kbps	62.5 µs	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256 kbps	31 µs	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512 kbps	15 µs	1 ms	2 ms	4 ms	8 ms	16 ms	23 ms
768 kbps	10 µs	640 µs	1.28 ms	2.56 ms	5.1 ms	10.2 ms	15 ms
1536 kbps	5 µs	320 µs	640 µs	1.28 ms	2.56 ms	5.12 ms	7.5 ms

µs: microseconds
ms: milliseconds

CVOICE v4.2—7-9

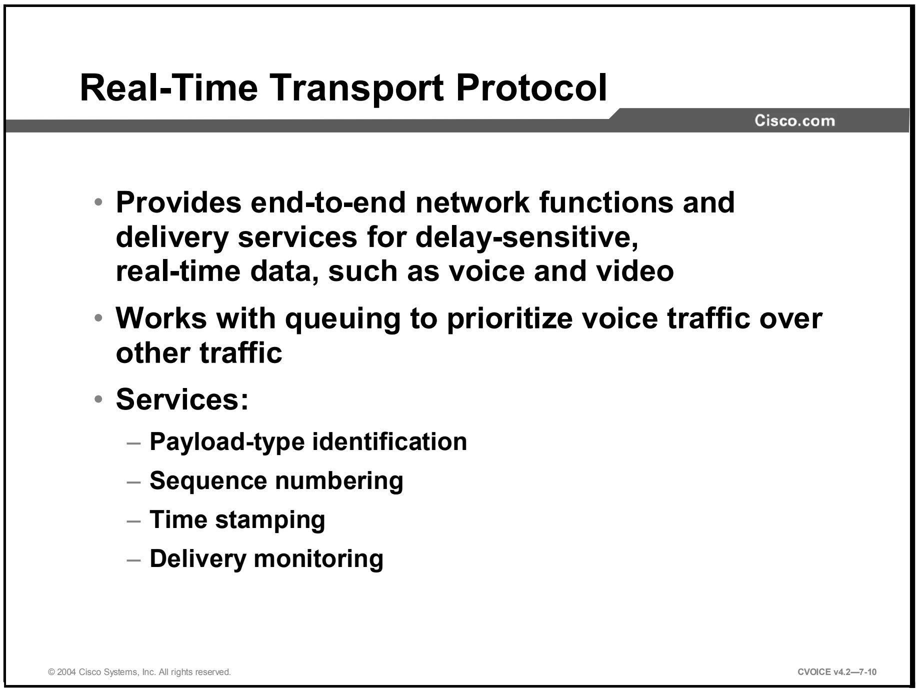
Serialization delay is defined as the amount of time it takes to deposit a packet on to a serial interface. This delay is directly related to the clock speed of the interface and the size of the packet. The table in the figure shows the delay problems that occur when dealing with large packets and slow links. When you are calculating acceptable delay, slower links may require that you fragment larger packets and prioritize voice packets. Look at the time it takes to send a 1500-byte packet across a 56-kbps link. If the goal for a private network is to stay around 200 ms for one-way delay, it has already been exceeded on just the WAN link alone. The only way to resolve this issue is to upgrade the WAN link, which is costly, or to fragment all of the packets so they are the same size as voice packets.

Example: Serialization Delays on Voice Networks

Serialization delays contribute to jitter and one-way delay on voice networks. If the talker says, "Watson, come here. I want you," the listener might hear, "Wat…..son, come…….here, I…….wa………..nt you." The time between the spoken voice and the speech that is heard might also experience a long delay.

Packet Sequencing

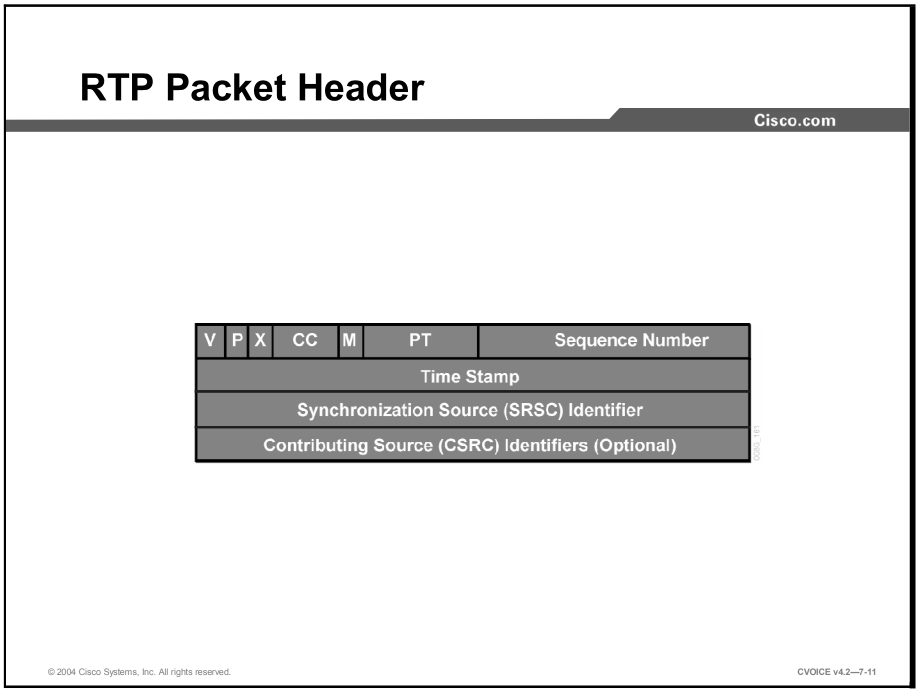
Because voice packets are carried using UDP over IP, packets often arrive out of sequence at the receiving end. This topic describes the method that you must use to sequence packets.



Real-Time Transport Protocol (RTP) provides end-to-end network functions and delivery services for delay-sensitive, real-time data, such as voice and video.

To support VoIP, RTP works with queuing to prioritize voice traffic over other traffic. RTP services include the following:

- **Payload-type identification:** A field in the RTP header that specifies that this packet carries real-time audio.
- **Sequence numbering:** A field in the RTP header that sequences this packet in relation to all others.
- **Time stamping:** A field in the RTP header that aids in measuring the time it takes this packet to reach its destination. The time stamp is relative to a synchronized clock maintained by Real-Time Transport Control Protocol (RTCP).
- **Delivery monitoring:** Since each packet is sequenced, RTP aids in detection of missing packets, and can therefore inform the DSP as to when prediction is necessary.



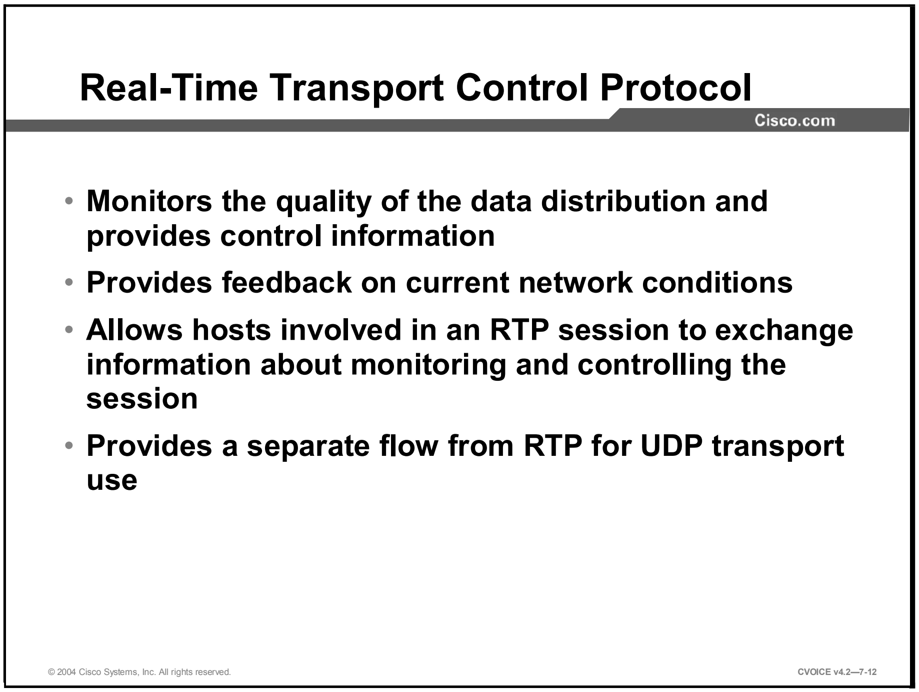
The RTP header consists of nine fields and is a minimum of 12 bytes. If multiple sources are included in the stream, the optional contributing source (CSRC) identifier field contents are appended to the end of the RTP header.

The table shows the RTP fields, their lengths, and their uses.

RTP

Field	Length	Use
Version (V)	2 bits	RTP version; current version is 2
Padding (P)	1 bit	Indicates if the header has padding bits at the end
Extension (X)	1 bit	If set, indicates that exactly one header extension is included
Contributing source count (CC)	4 bits	Number of CSRC identifiers in the CSRC field following the initial RTP header
Marker (M)	1 bit	Interpretation defined by a profile
Payload type (PT)	7 bits	Payload type, frequently the codec type
Sequence number	16 bits	Sequential number, beginning with a random number, that the receiver may use to indicate packet loss
Time stamp	32 bits	Time stamp of the first octet of data in the RTP data packet; the initial value is a random value
Synchronization source (SRSC)	32 bits	Randomly chosen value used within a data stream to permit multiplexing multiple streams
Contributing source (CSRC)	0 to 15 items, 32 bits each	If multiple streams are mixed into a single stream, the list of source streams is included here; for example, in a conference in which multiple streams are sent, each participant is marked as a separate contributing source

Note For more information on RTP, refer to RFC 1889.



RTCP monitors the quality of the data distribution and provides control information. It provides the following feedback on current network conditions:

- A mechanism for hosts involved in an RTP session to exchange information about monitoring and controlling the session. RTCP monitors quality for such elements as packet counts, packets lost, and interarrival jitter. RTCP generally allocates a fixed 5 percent of total session bandwidth (for its own use) with a minimum interval of 5 seconds between RTCP packets. RTCP imposes these limits on itself so that it does not interfere with the packets that carry the digitized voice.
- A separate flow from RTP for UDP transport use.

Both RTP and RTCP are detailed in the H.323 specification. After the H.323 call setup and control process is complete, UDP sends audio and video packets. You must remember that UDP cannot guarantee packet delivery and ordering. To assist with streaming audio and video, the H.323 specification calls for an RTP header.

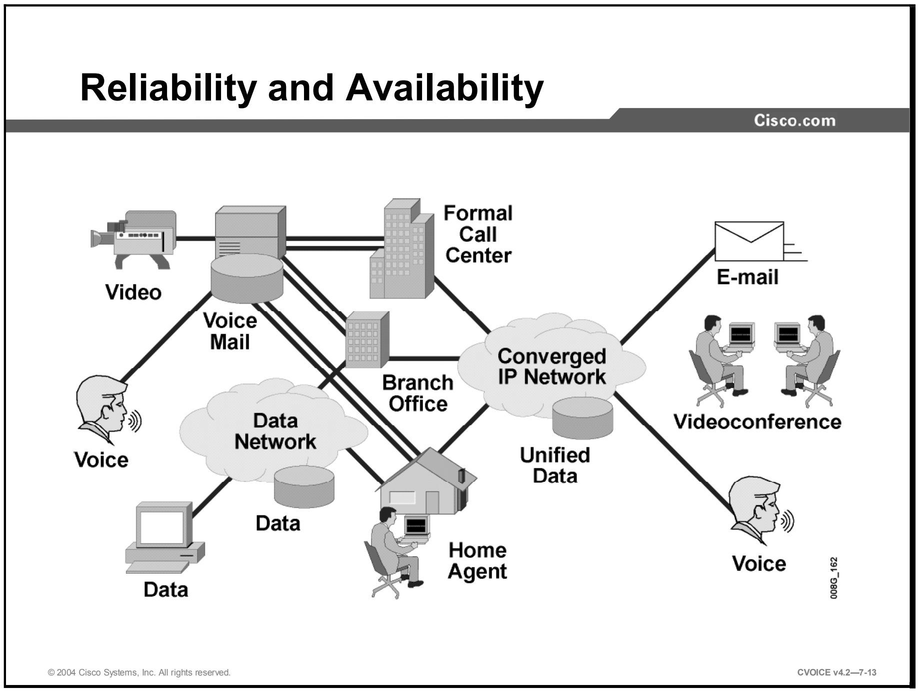
An RTP header contains a time stamp and sequence number that enables the receiving device to buffer as much as necessary to remove jitter and latency by synchronizing the packets to play back a continuous stream of sound.

Example: RTP and RTCP Applied

When RTP and RTCP are applied in a voice network, together they measure the average jitter experienced. The receiving voice equipment can then set up a buffer to be used to alleviate the interarrival delay. RTCP synchronizes a clock between the end stations, allowing the RTP time stamp to aid in determining the size of the dejitter buffer. For example, if packets continually arrive with delays near 200 ms, the dejitter buffer will be set up for an average size of 8 to 10 packets.

Reliability and Availability

This topic discusses the importance of reliability and availability of the converged network.



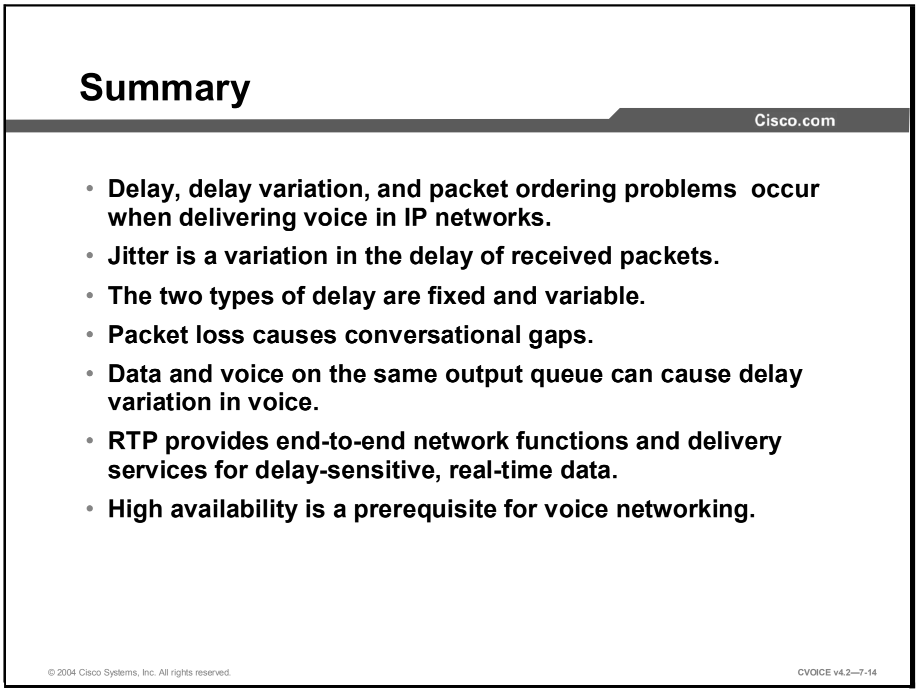
A prerequisite for voice networking is high availability as shown in the figure. Voice networks are often considered more critical and available than their data-centric counterparts. This situation exists even though e-mail is considered the primary form of business communications media and is more popular than voice. There is little impact to the user if an e-mail is a few seconds late, but if the packets carrying a voice call are late by just a few milliseconds, then the quality of the call is degraded. When the voice and data networks are converged, the availability and reliability of the data network now becomes every bit as important as the voice network. When you do the engineering for a new network to support converged services, or the reengineering of an existing network to support converged services, you must consider how and where high availability and reliability can be applied to the network, and the costs associated with these services. Cisco Architecture for Voice, Video and Integrated Data (AVVID) is a distributed architecture that is inherently available and scalable. The ability to seamlessly provide additional capacity for infrastructure, services, and applications is a unique benefit of the architecture.

Example: Reliability and Availability

Users of the PSTN have a certain perception relating to its performance. When someone goes off hook on their home telephone, they expect dial tone. When they dial a destination, they expect the network to complete the call. As voice and data networks are converged, this perception of availability and performance translates to the VoIP network. Network administrators must ensure that tools for high reliability and availability are in place. This includes redundancy of switching systems, routers, switches, and uninterruptible power supply (UPS) systems.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to these resources:

- *Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)*
  http://www.cisco.com/warp/public/788/voice-qos/jitter_packet_voice.html
- *Understanding Delay in Packet Voice Networks*
  http://www.cisco.com/warp/public/788/voip/delay-details.html
- *QoS Documentation*
  http://www.cisco.com/pcgi-bin/Support/PSP/psp_view.pl?p=Internetworking:Voice:QoS

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which feature of an IP network causes delay and delay variation in voice traffic?

A) connectionless transmission

B) fragmentation and reassembly

C) FIFO queuing

D) no error correction or acknowledgements

Q2) Which Layer 4 protocol is used for transmitting packets in VoIP networks?

A) IP

B) TCP

C) UDP

D) Frame Relay

Q3) How can you reduce jitter in a VoIP network?

A) by using dejitter buffers

B) by using FIFO queuing

C) by using header compression

D) by using guaranteed delay

Q4) Variable delay is also known as _____.

A) nondeterministic

B) jitter

C) playout delay

D) FIFO

Q5) According to ITU-T Recommendation G.114 for national telecommunications administrations, how much delay is acceptable for most user applications?

A) 0 to 150 ms

B) 200 to 250 ms

C) 150 to 400 ms

D) above 400 ms

Q6) According to Cisco, what is the acceptable level of one-way delay for private networks?

A) 0 to 150 ms

B) 200 to 250 ms

C) 150 to 400 ms

D) above 400 ms

Q7) In Cisco VoIP networks using default payload sizes, how many packets can be lost without the listener experiencing gaps in conversation?

A) none

B) one

C) two

D) no more than five

Q8) Which two conditions can result in voice packets being dropped? (Choose two.)

A) slow links

B) too much noise

C) too much jitter

D) too much congestion

Q9) What is the major cause of delay when voice and data packets are placed in the same output queue?

A) large packets

B) low bandwidth

C) firewalls

D) retransmission of lost packets

Q10) Which two techniques can be used on slower links in a voice and data network to reduce delay in voice transmission? (Choose two.)

A) FIFO queuing

B) buffering of voice packets

C) fragmentation of large packets

D) prioritization of voice packets

E) compression of data packets

Q11) What is the size of the RTP header?

A) 3 bytes

B) 5 bytes

C) 12 bytes

D) 40 bytes

Q12) Which RTP header field is used to allow multiplexing of multiple data streams?

A) contributing source count

B) marker

C) payload type

D) sequence

E) time stamp

F) synchronization source

G) contributing source

Q13) What is one of the main benefits of using Cisco AVVID for voice and data networks?

A) seamless scalability of infrastructure and applications

B) ability to provide equal quality for voice and data

C) low cost of implementation

D) integration with legacy networks

Q14) Which factor is a prerequisite for voice networking?

A) TCP sessions

B) FIFO queuing

C) high availability

D) low cost

Quiz Answer Key

Q1) C

Relates to: IP Networking Overview

Q2) C

Relates to: IP Networking Overview

Q3) A

Relates to: Jitter

Q4) B

Relates to: Jitter

Q5) A

Relates to: Delay

Q6) B

Relates to: Delay

Q7) B

Relates to: Effect of Packet Loss on Quality

Q8) C, D

Relates to: Effect of Packet Loss on Quality

Q9) A

Relates to: Competition Between Voice, Data, and Video for Bandwidth

Q10) C, D

Relates to: Competition Between Voice, Data, and Video for Bandwidth

Q11) C

Relates to: Packet Sequencing

Q12) F

Relates to: Packet Sequencing

Q13) A

Relates to: Reliability and Availability

Q14) C

Relates to: Reliability and Availability

Understanding QoS and Good Design

Overview

This lesson introduces the concept of QoS as it applies to a converged network and guides the student through resources to learn more about QoS.

Relevance

Successful implementation of VoIP requires an overall understanding of QoS and knowledge of QoS resources.

Objectives

Upon completing this lesson, you will be able to design a voice network that operates optimally. This includes being able to meet these objectives:

- Describe the problems that are associated with transmitting voice in converged networks
- List five ways that QoS improves voice quality
- Identify the features that are offered by Cisco IOS QoS for voice at different points in the network
- Describe the certification course that is offered by Cisco as a resource for network administrators
- Find information about QoS in the appropriate location on the Cisco.com website
- List three Cisco Press publications that can be used as resources for network administrators
- Identify four characteristics of a poorly designed VoIP network
- List the Cisco educational resources that are available for QoS implementation methods

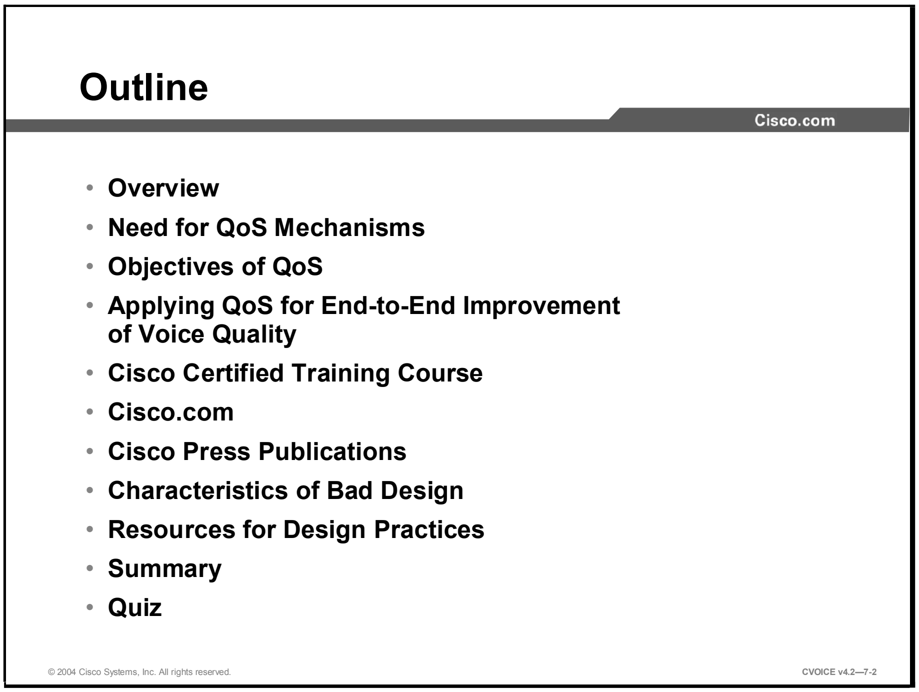
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of VoIP networks
- Knowledge of traditional telephony networks
- Knowledge of TCP/IP networks

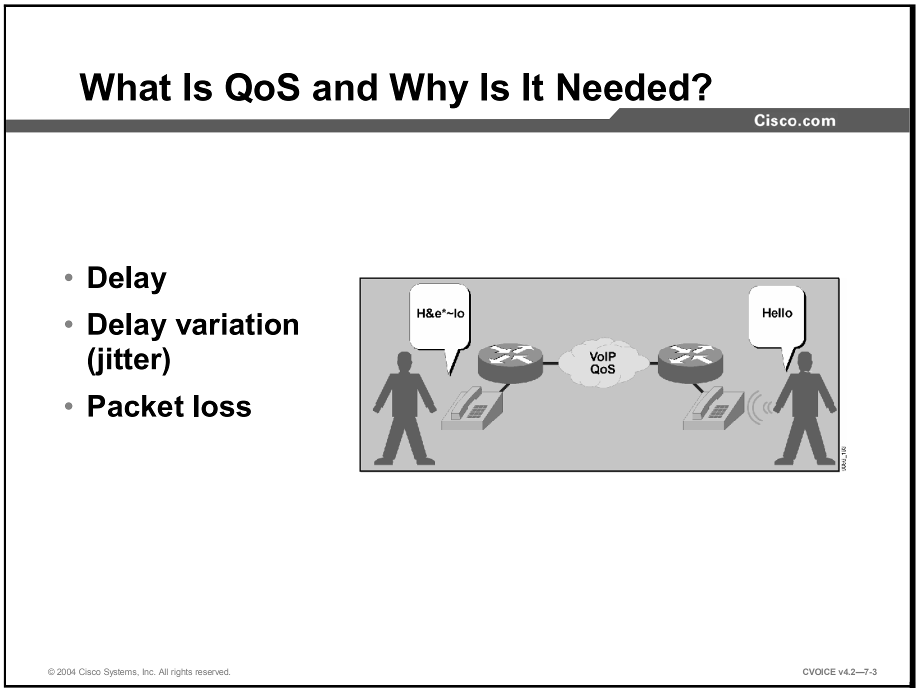
Outline

The outline lists the topics included in this lesson.



Need for QoS Mechanisms

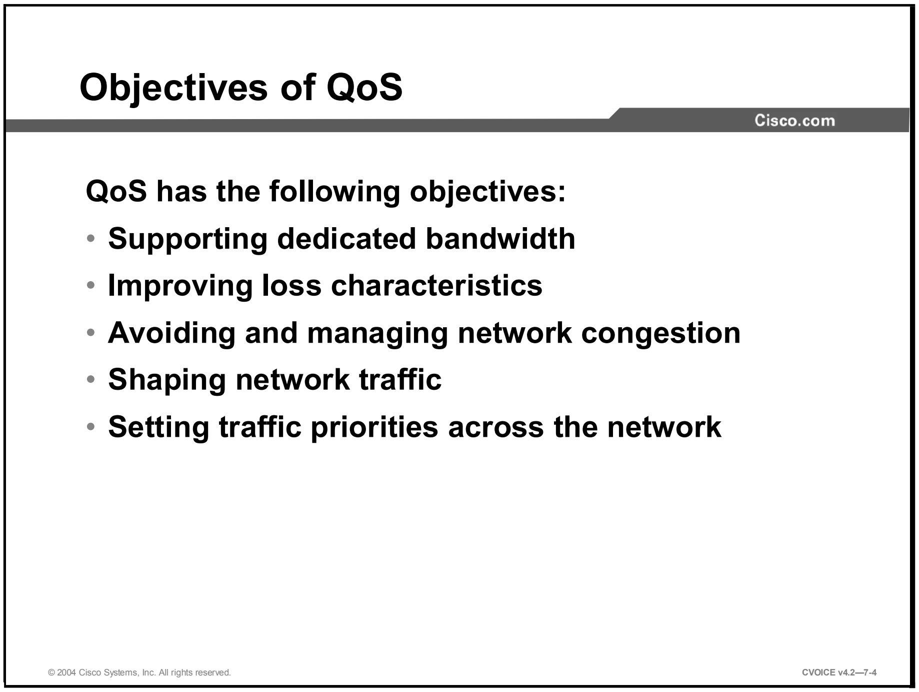
This topic describes problems associated with transmitting voice over a data network and the need for QoS in such a network.



Real-time applications, such as voice applications, have different characteristics and requirements than traditional data applications. Voice applications tolerate little variation in the amount of delay. This delay variation affects delivery of voice packets. Packet loss and jitter degrade the quality of the voice transmission that is delivered to the recipient. The figure shows how these problems can affect a voice message.

Objectives of QoS

To ensure that VoIP is a realistic replacement for standard PSTN telephony services, customers must receive the same consistently high quality of voice transmission that they receive with basic telephone services. This topic discusses how QoS can help you achieve this objective.



Like other real-time applications, VoIP is extremely sensitive to issues related to bandwidth and delay. To ensure that VoIP transmissions are intelligible to the receiver, voice packets cannot be dropped, excessively delayed, or subject to variations in delay, or jitter.

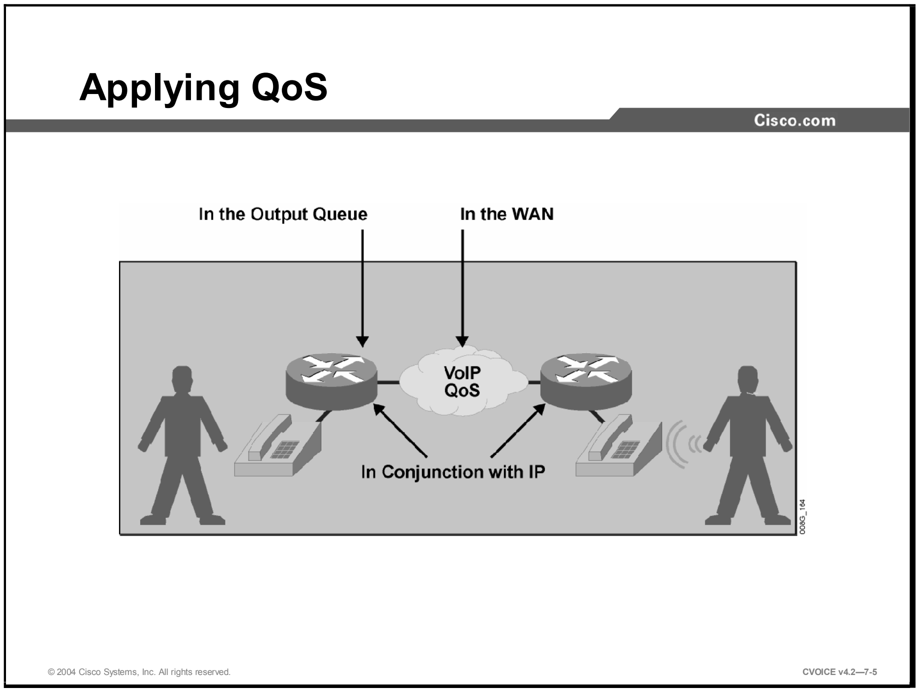
Example: QoS Objectives

VoIP guarantees high-quality voice transmission only if the signaling and audio channel packets have priority over other kinds of network traffic. To deploy VoIP, you must provide an acceptable level of voice quality by meeting VoIP traffic requirements for issues related to bandwidth, latency, and jitter. QoS provides better, more predictable network service by performing the following:

- **Supporting dedicated bandwidth:** Designing the network such that speeds and feeds can support the desired voice and data traffic
- **Improving loss characteristics:** Designing the Frame Relay network such that discard eligibility is not a factor, keeping voice below committed information rate (CIR)
- **Avoiding and managing network congestion:** Ensuring that the LAN and WAN infrastructure can support the volume of data traffic and voice calls
- **Shaping network traffic:** Using Cisco traffic-shaping tools to ensure smooth and consistent delivery of frames to the WAN
- **Setting traffic priorities across the network:** Marking the voice traffic as priority and queuing it first

Applying QoS for End-to-End Improvement of Voice Quality

Voice features for Cisco IOS QoS are deployed at different points in the network and designed for use with other QoS features to achieve specific goals, such as control over jitter and delay. This topic lists the network areas in which Cisco IOS QoS is implemented.



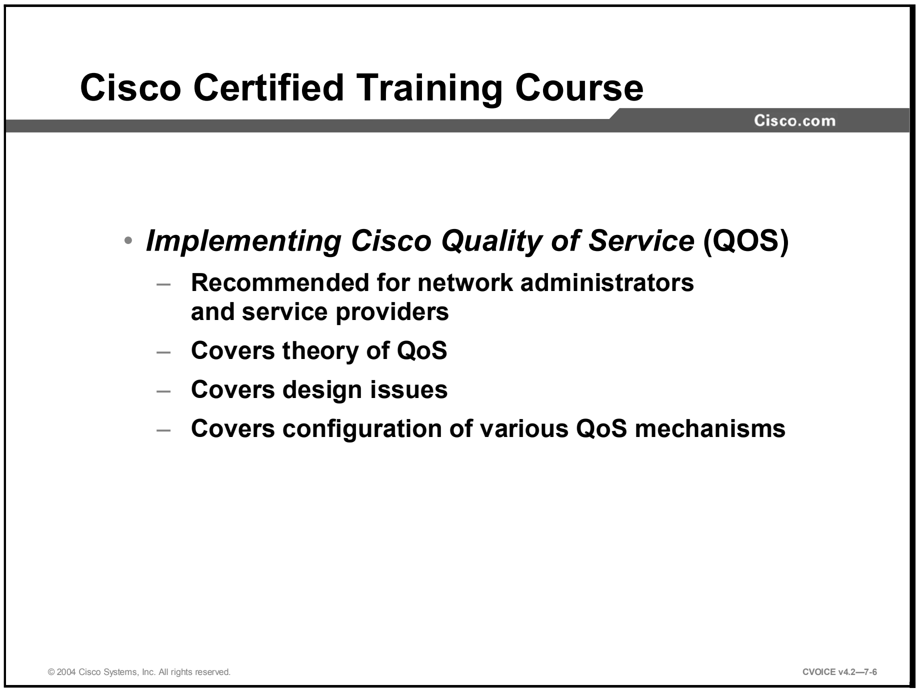
Cisco IOS software includes a complete set of features for delivering QoS throughout the network. Following are Cisco IOS features that address the voice packet delivery requirements of end-to-end QoS and service differentiation:

- In the output queue of the router:
  - **Class-based weighted fair queuing (CBWFQ):** Extends the standard weighted fair queuing (WFQ) functionality by providing support for user-defined traffic classes. You can create a specific class for voice traffic by using CBWFQ.
  - **Low Latency Queuing (LLQ):** Provides strict priority queuing on ATM VCs and serial interfaces. LLQ configures the priority status for a class within CBWFQ and is not limited to UDP port numbers (as in IP RTP priority). LLQ is considered a “best practice” by the Cisco Enterprise Solutions Engineering (ESE) group for delivering voice QoS services over a WAN.
  - **WFQ and distributed weighted fair queuing (DWFQ):** Segregates traffic into flows and then schedules traffic onto the outputs to meet specified bandwidth allocation or delay bounds.
  - **Weighted random early detection (WRED) and distributed weighted random early detection (DWRED):** Provides differentiated performance characteristics for different classes of service. This classification allows preferential handling of voice traffic under congestion conditions without worsening the congestion.

- In the WAN or WAN protocol:
  - **Committed access rate (CAR):** Provides a rate-limiting feature for allocating bandwidth commitments and bandwidth limitations to traffic sources and destinations. At the same time, it specifies policies for handling the traffic that may exceed bandwidth allocation.
  - **Frame Relay traffic shaping (FRTS):** Delays excess traffic by using a buffer or queuing mechanism to hold packets and shape the flow when the data rate of the source is higher than expected.
  - **Frame Relay Forum Standard 12 (FRF.12):** Ensures predictability for voice traffic by providing better throughput on low-speed Frame Relay links. FRF.12 interleaves delay-sensitive voice traffic on one VC with fragments of a long frame on another VC that is using the same interface.
  - **IP to ATM class of service (CoS):** Includes a feature suite that maps CoS characteristics between the IP and ATM. It also offers differential service classes across the entire WAN—not just the routed portion—and gives mission-critical applications exceptional service during periods of high network usage and congestion.
  - **Multilink PPP (MLP) with link fragmentation and interleaving (LFI):** Allows large packets to be multilink-encapsulated and fragmented so that they are small enough to satisfy the delay requirements of real-time traffic. LFI also provides a special transmit queue for smaller, delay-sensitive packets, enabling them to be sent earlier than other flows.
- In conjunction with the IP operation:
  - **Compressed Real-Time Transport Protocol (CRTP):** Compresses the extensive RTP header when used in conjunction with RTP. The result is decreased consumption of available bandwidth for voice traffic and a corresponding reduction in delay.
  - **Resource Reservation Protocol (RSVP):** Supports the reservation of resources across an IP network, allowing end systems to request QoS guarantees from the network. For networks that support VoIP, RSVP—in conjunction with features that provide queuing, traffic shaping, and voice call signaling—provides Call Admission Control (CAC) for voice traffic.
  - **QoS policy propagation on Border Gateway Protocol (BGP):** Steadies BGP to distribute QoS policy to remote routers in a network. It allows classification of packets and then uses other QoS features, such as CAR and WRED, to specify and enforce business policies to fit a business model.

Cisco Certified Training Course

A number of resources are available to help network administrators understand and implement QoS. This topic describes the certification course that Cisco offers.



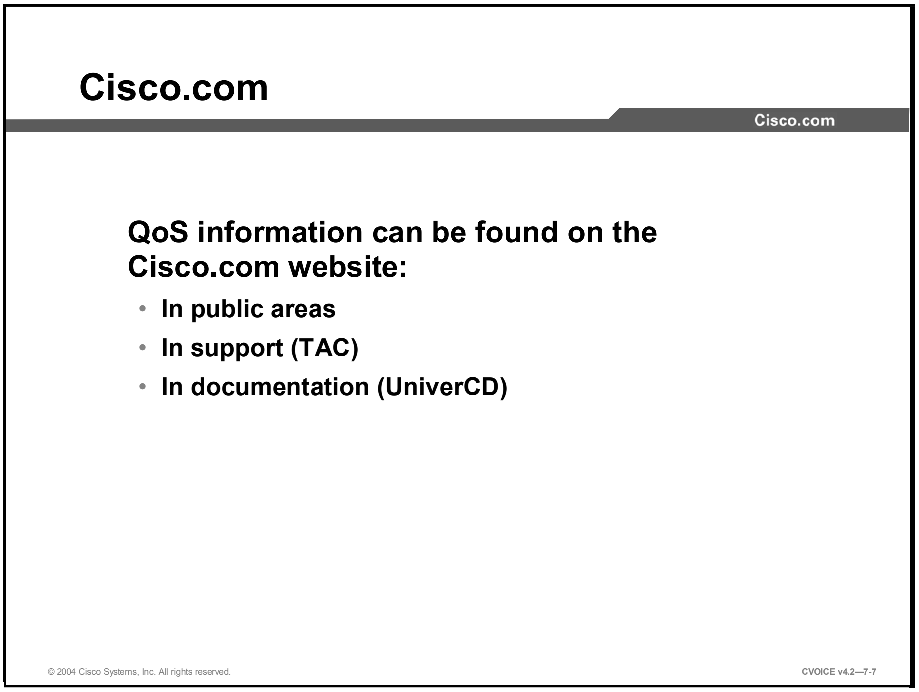
Cisco Systems offers the certification course, *Implementing Cisco Quality of Service* (QOS). This course provides students with in-depth knowledge of IP QoS requirements, conceptual models such as Best Effort, Integrated Services (IntServ), Differentiated Services (DiffServ), and the implementation of IP QoS on Cisco IOS platforms.

The curriculum covers the theory of IP QoS, design issues, and configuration of various QoS mechanisms to facilitate the creation of effective administrative policies providing QoS. Case studies and lab exercises included in the course help students apply concepts mastered in individual modules to real-life scenarios.

The course also gives students design and usage rules for various advanced IP QoS features and the integration of IP QoS with underlying Layer 2 QoS mechanisms, allowing them to design and implement efficient, optimal, and trouble-free multiservice networks.

Cisco.com

The Cisco.com website is a resource that provides network administrators with additional information on QoS. This topic lists the QoS resources.

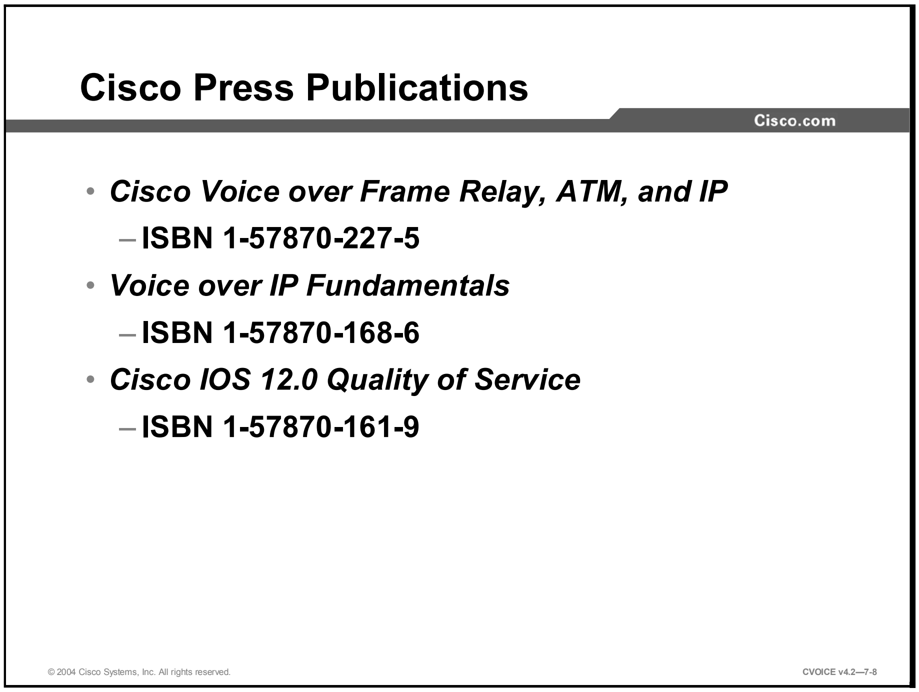


Information about QoS on the Cisco.com website is found as follows:

- **In public areas:**
  - *Voice Quality*
    http://www.cisco.com/pcgi-bin/Support/browse/index.pl?i=Technologies&f=775
  - *Understanding Delay in Packet Voice Networks*
    http://www.cisco.com/warp/public/788/voip/delay-details.html
  - *Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)*
    http://www.cisco.com/warp/public/788/voice-qos/jitter_packet_voice.html
- **In support (Technical Assistance Center [TAC]):**
  - *Technical Support: Voice Quality*
    http://www.cisco.com/pcgi-bin/Support/PSP/psp_view.pl?p=Internetworking:Voice:QoS
- **In documentation (UniverCD):**
  - *Congestion Management Overview*
    http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt2/qcfconmg.htm#48685

Cisco Press Publications

This topic lists Cisco Press publications that can be helpful to network administrators in understanding and implementing QoS.

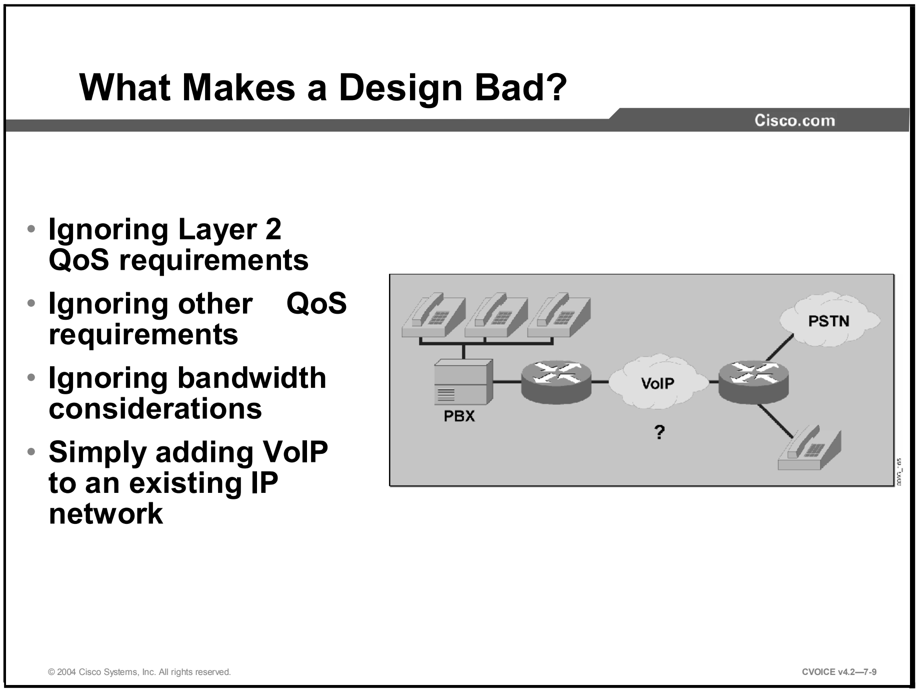


Cisco Press publications are also excellent sources of information. The following titles may be helpful:

- *Cisco Voice over Frame Relay, ATM, and IP*
  - ISBN 1-57870-227-5
- *Voice over IP Fundamentals*
  - ISBN 1-57870-168-6
- *Cisco IOS 12.0 Quality of Service*
  - ISBN 1-57870-161-9

Characteristics of Bad Design

A network that is well engineered from end to end is necessary for running delay-sensitive applications such as VoIP. Fine-tuning the network to adequately support VoIP involves a series of protocols and features geared toward improving QoS. This topic identifies characteristics of a poorly designed network.



QoS is the ability of a network to provide DiffServ to selected network traffic over various underlying technologies. QoS is not inherent in a network infrastructure. Instead, QoS is implemented by strategically enabling appropriate QoS features throughout the network.

Poor design characteristics include:

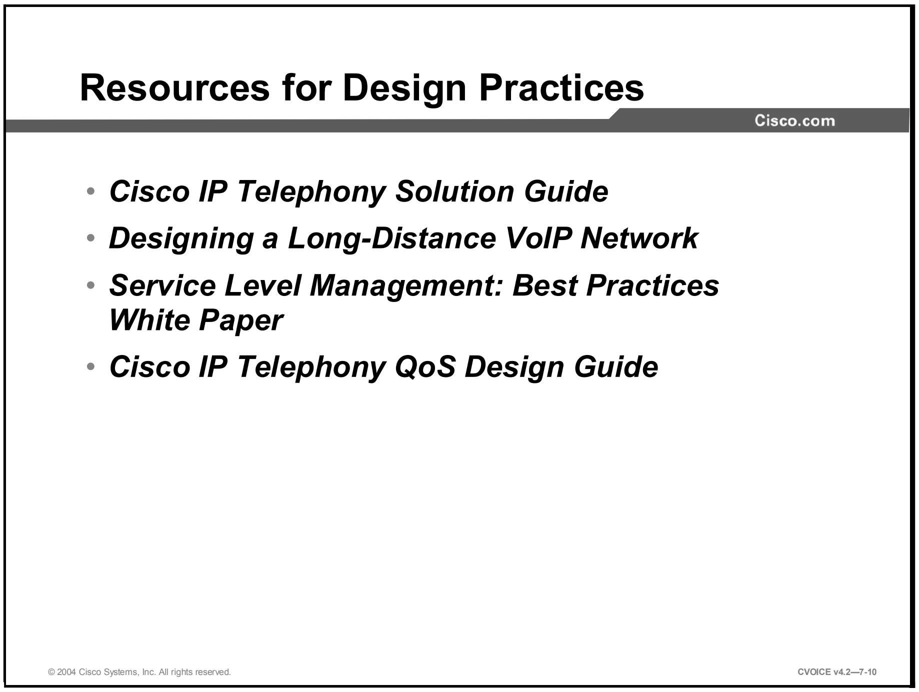
- **Ignoring Layer 2 QoS requirements:** Layer 2 QoS includes FRF.12, LFI, and traffic shaping.
- **Ignoring other QoS requirements:** Services such as LLQ, RTP, congestion management, and congestion avoidance must be enabled.
- **Ignoring bandwidth considerations:** Planning for the total number of calls and their effect on data bandwidth is critical to all users of the network.
- **Simply adding VoIP to an existing IP network:** When considering VoIP, network administrators must insist on a complete network redesign for a comprehensive end-to-end solution.

Example: Deploying QoS

Many people think the fastest way to fix network performance is to add a lot of bandwidth. That may work well in certain situations like campus networks, in which upgrading from 10 Mbps to 100 Mbps or 1 GB connections may be possible. What happens in the WAN? The cost to upgrade a WAN circuit from 56 kbps to T1 may be quite high, or not available for certain locations of your network. What is the answer? You must configure QoS throughout the network—not simply on the Cisco devices that are running VoIP—to improve voice network performance. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in a network do not necessarily perform the same operations; the QoS tasks they perform may differ as well. To configure an IP network for real-time voice traffic, you must consider the functions of both edge and backbone routers, and then select the appropriate QoS tools.

Resources for Design Practices

Well-designed networks are usually created as a result of trial and error. This topic lists various resources that Cisco offers to help you design a voice network that operates optimally.



Many resources are available through the Cisco.com website. You can access these resources by using the search tool on the main page. Here are several examples:

- To help organizations prepare for Cisco IP telephony, Cisco offers the *Cisco IP Telephony Solution Guide*, and the web-based *Cisco IP Telephony Readiness Assessment*. These resources help organizations implement both the network design needed to support IP telephony and the Cisco IP telephony solution itself.
  - Cisco IP Telephony Solution Guide (A user ID and password are required to access this site.) http://www.cisco.com/tac/iptelsolguide
  - *Cisco IP Telephony Readiness Assessment* (A user ID and password are required to access this site.) http://tools.cisco.com/Assessments/jsp/welcome.jsp?asmt=VOIP
- The long-distance VoIP network design solution includes multiple components in various combinations from both Cisco and third-party vendors. Voice points of presence (POPs) connected to other service providers are a central component in the delivery of wholesale voice services. The types of interconnections, or call topologies, that service providers support will determine the specific components and design methods that Cisco recommends.
  - *Designing a Long-Distance VoIP Network* http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/longd.htm

- A white paper that describes service-level management (SLM) and service level agreements (SLAs) can be found on the Cisco website:
  - *Service Level Management: Best Practices White Paper* http://www.cisco.com/warp/public/126/sla.htm
- To help organizations plan for and implement QoS, *Cisco IP Telephony QoS Design Guide* suggests strategies for both campus and WAN environments. This information can be found on the Cisco website:
  - Cisco IP Telephony Network Design Guides http://www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/iptlink.htm

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- QoS mitigates delay, jitter, and packet loss in converged voice and data networks.
- QoS supports dedicated bandwidth, improves loss characteristics, avoids and manages network congestion, shapes network traffic, and sets traffic priorities across the network.
- Cisco IOS QoS for voice features can be implemented across the entire network.
- *Implementing Cisco Quality of Service* (QOS) is the certification course offered by Cisco to help network administrators understand and implement QoS.

 CVOICE v4.2—7-11

Summary (Cont.)

Cisco.com

- Information about QoS on Cisco.com is found in public areas, in support, and in documentation.
- *Cisco Voice over Frame Relay, ATM , and IP*; *Voice over IP Fundamentals;* and *Cisco IOS 12.0 Quality of Service* are Cisco Press publications that are valuable resources for network administrators.
- Ignoring Layer 2 and other QoS requirements and bandwidth considerations, and simply adding VoIP to an existing IP network are poor network design elements that contribute to poor QoS.
- Cisco offers resources on the Cisco.com website to help network administrators learn more about network design.

 CVOICE v4.2—7-12

References

For additional information, refer to these resources:

- *Voice Quality*
  http://www.cisco.com/pcgi-bin/Support/PSP/psp_view.pl?p=Internetworking:Voice:QoS
- *Voice Quality (Quality of Service)*
  http://www.cisco.com/pcgi-bin/Support/browse/index.pl?i=Technologies&f=775

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which two factors have a minimal effect on data transmissions but negatively impact voice transmissions? (Choose two.)

A) high bandwidth

B) T1 links

C) packet loss

D) jitter

E) Layer 2 protocol

Q2) What are the two requirements of effective voice transmission over an IP network? (Choose two.)

A) low latency

B) G.711 codecs

C) reliable delivery of packets

D) use of compression algorithms

E) high-speed links

Q3) Why is VoIP highly sensitive to bandwidth and delay problems?

A) because it transmits both voice and data

B) because it is a real-time application

C) because it is an older technology

D) because of its default settings

Q4) Which is NOT an objective of QoS?

A) improving loss characteristics

B) providing error correction

C) shaping network traffic

D) setting traffic priorities across the network

Q5) Which two Cisco IOS QoS features are employed in the output queue of the router? (Choose two.)

A) FRF.12

B) IP to ATM CoS

C) CBWFQ

D) CRTP

E) RSVP

F) WRED

Q6) Which two Cisco QoS features are deployed in the WAN? (Choose two.)

A) CAR

B) DWFQ

C) MLP with LFI

D) QoS policy propagation via BGP

E) CRTP

Q7) If you want to learn about IP QoS features provided by Cisco IOS, which certification course should you take?

A) Deploying QoS for Enterprise Networks

B) Implementing Cisco Quality of Service

C) Cisco IOS 12.0 Quality of Service

D) Cisco IP Telephony Network Design

Q8) Which Cisco course describes the Integrated Services and the Differentiated Services QoS models?

A) Service Level Management

B) Cisco IP Telephony Network Design

C) Deploying QoS for Enterprise Networks

D) Implementing Cisco Quality of Service

Q9) Which three areas of the Cisco.com website contain information about QoS? (Choose three.)

A) documentation

B) private

C) products

D) public

E) publications

F) support

Q10) In which area of Cisco.com would you find the document *Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)*?

A) public

B) support

C) documentation

Q11) Which title is a Cisco Press publication that may be helpful for understanding QoS implementation?

A) *Cisco IP Telephony Solution Guide*

B) *Designing a Long Distance VoIP Network*

C) *Cisco IOS 12.0 Quality of Service*

D) *Graphical Navigation Guide for VoIP Issues*

Q12) Which title is published by Cisco Press to help network administrators understand and implement QoS?

A) *Cisco IP Telephony Network Design Guide*

B) *IP Telephony Readiness Assessment*

C) *Voice over IP Fundamentals*

D) *Voice Quality (Quality of Service)*

Q13) Which practice indicates a bad network design?

A) completely redesigning the network

B) using Low Latency Queuing

C) ignoring Layer 3 QoS requirements

D) ignoring bandwidth requirements

Q14) Different QoS tools must be selected for each device based on _____.

A) the devices that are running VoIP

B) the functions of the device

C) the country in which the device is located

D) the times of highest traffic

Q15) Which resource on Cisco.com provides information on the specific components and design methods recommended by Cisco for long-distance VoIP networks?

A) *Cisco IP Telephony Network Design Guide*

B) *IP Telephony Readiness Assessment*

C) *Voice over IP Fundamentals*

D) *Deploying QoS for Enterprise Networks*

E) *Designing a Long-Distance Telephone Network*

Q16) Which resource on Cisco.com is meant to help organizations plan for and implement QoS?

A) *Implementing Cisco Quality of Service*

B) *Cisco IP Telephony Network Design Guides*

C) *Service Level Management: Best Practices white paper*

D) *Designing a Long-Distance Telephone Network*

E) *Deploying QoS for Enterprise Networks*

Quiz Answer Key

Q1) C, D

Relates to: Need for QoS Mechanisms

Q2) A, C

Relates to: Need for QoS Mechanisms

Q3) B

Relates to: Objectives of QoS

Q4) B

Relates to: Objectives of QoS

Q5) C, F

Relates to: Applying QoS for End-to-End Improvement of Voice Quality

Q6) A, C

Relates to: Applying QoS for End-to-End Improvement of Voice Quality

Q7) B

Relates to: Cisco Certified Training Course

Q8) D

Relates to: Cisco Certified Training Course

Q9) A, D, F

Relates to: Cisco.com

Q10) A

Relates to: Cisco.com

Q11) C

Relates to: Cisco Press Publications

Q12) C

Relates to: Cisco Press Publications

Q13) D

Relates to: Characteristics of Bad Design

Q14) B

Relates to: Characteristics of Bad Design

Q15) E

Relates to: Resources for Design Practices

Q16) B

Relates to: Resources for Design Practices

Understanding Jitter

Overview

This topic describes jitter and the methods that are used to measure and compensate for it.

Relevance

Jitter and its effects can cause significant distortions in VoIP. Network administrators must be aware of the causes of and solutions for jitter in VoIP networks.

Objectives

Upon completing this lesson, you will be able to configure buffers for static and dynamic jitter. This includes being able to meet these objectives:

- Explain the reasons for jitter in a converged network and list two ways that networks can compensate for it
- Describe the methods that are used on Cisco Systems voice routers to overcome the problem of jitter
- Identify three symptoms that require the adjustment of playout delay buffer parameters
- Describe all the fields in the **show call active voice** command output that indicate the size of jitter
- Use the **playout-delay mode adaptive** command to configure a dynamic jitter buffer mode
- Use the **playout-delay mode fixed** command to configure a static jitter buffer mode

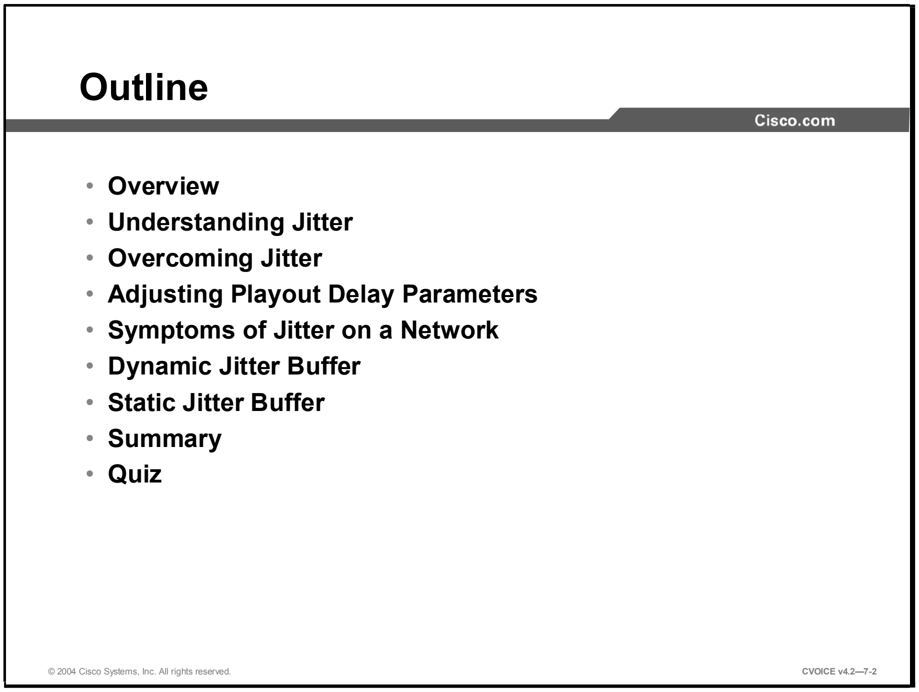
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of traditional telephony
- Knowledge of VoIP basics
- Knowledge of TCP/IP networks

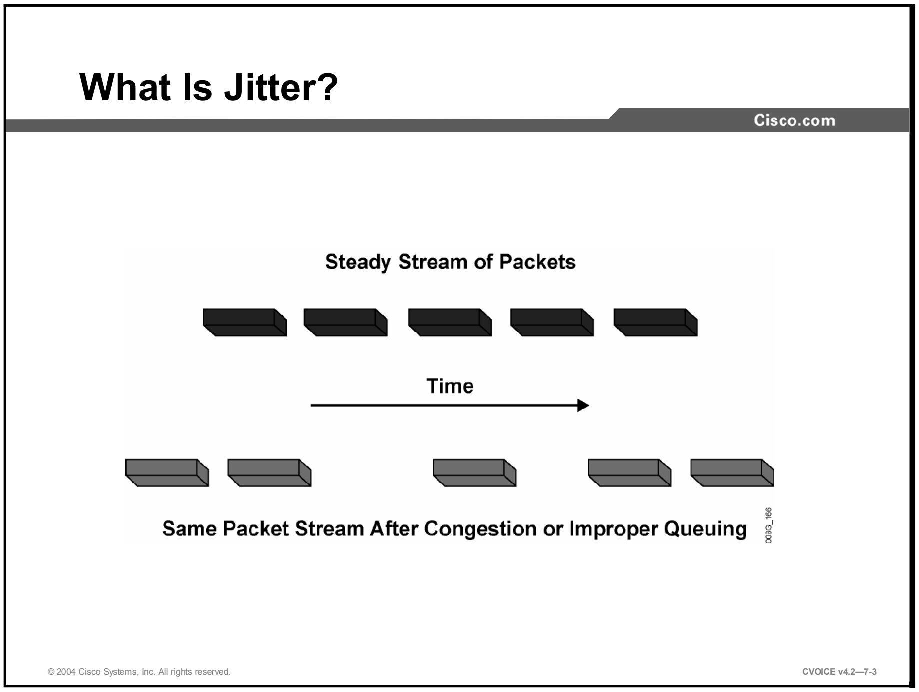
Outline

The outline lists the topics included in this lesson.

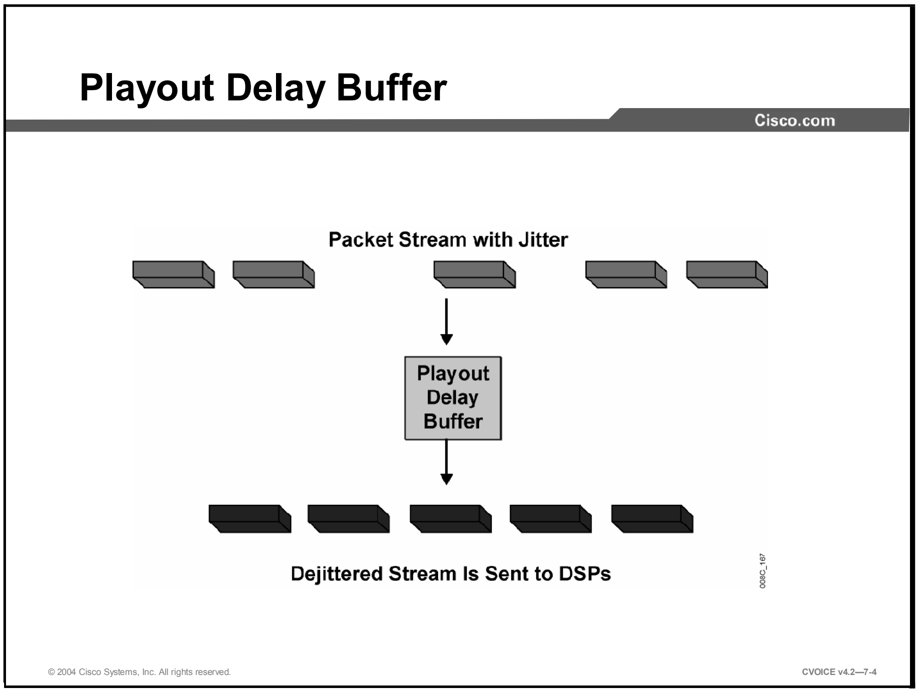


Understanding Jitter

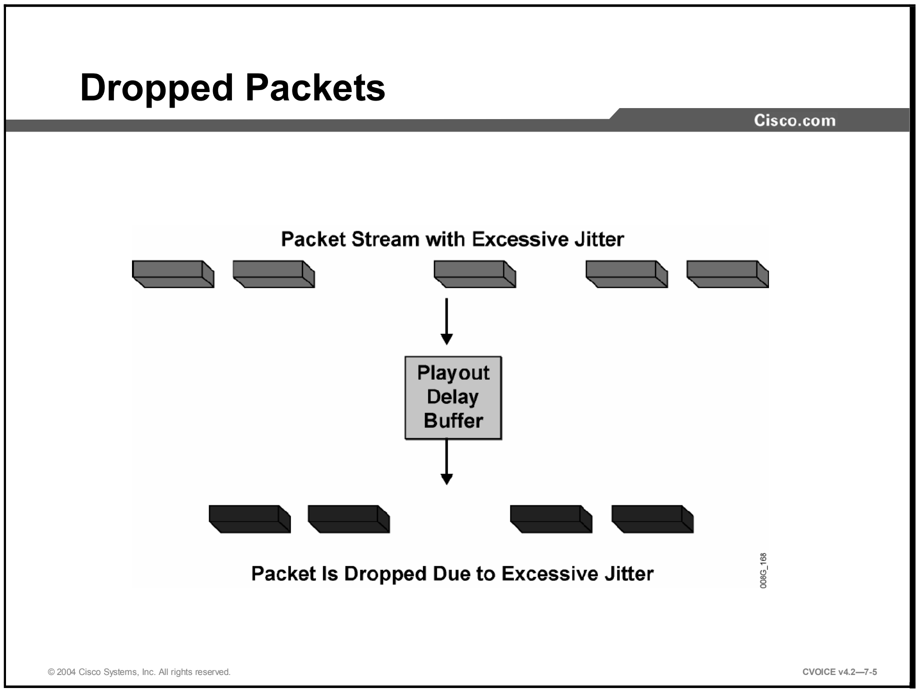
Jitter is an undesirable effect caused by the inherent tendencies of TCP/IP networks and components. This topic describes the cause and effect of jitter.



Jitter is defined as a variation in the delay of received packets. The sending side transmits packets in a continuous stream and spaces them evenly apart. Because of network congestion, improper queuing, or configuration errors, the delay between packets can vary instead of remaining constant, as shown in the figure. This variation causes problems for audio playback at the receiving end. Playback may experience gaps while waiting for the arrival of variable delayed packets.



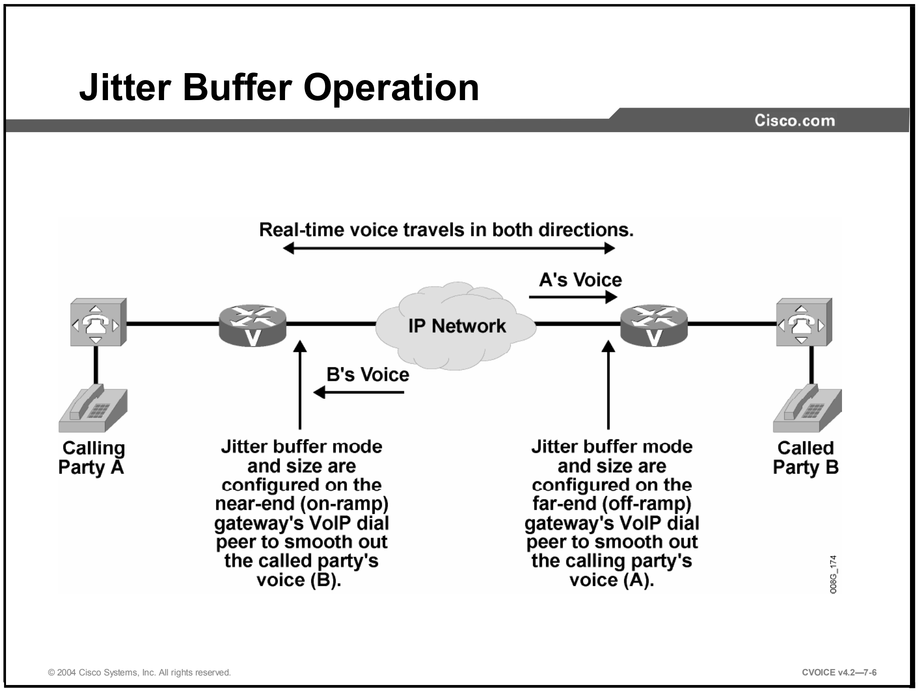
When a router receives an audio stream for VoIP, it must compensate for any jitter that it detects. The playout delay buffer mechanism handles this function. Playout delay is the amount of time that elapses between the time a voice packet is received at the jitter buffer on the DSP and the time a voice packet is played out to the codec. The playout delay buffer must buffer these packets and then play them out in a steady stream to the DSPs. The DSPs then convert the packets back into an analog audio stream. The playout delay buffer is also referred to as the dejitter buffer.



If the magnitude of jitter is so great that packets are received out of range of the playout delay buffer, the out-of-range packets are discarded and dropouts appear in the audio. For losses as small as one packet, the DSP interpolates what it calculates the audio should be, making the problem inaudible through the Cisco IOS Packet Loss Concealment (PLC) service.

Overcoming Jitter

Cisco voice networks compensate for jitter by setting up a buffer, called the "jitter buffer," on the gateway router at the receiving end of the voice transmission. This topic explains how to overcome jitter.



The jitter buffer receives voice packets from the IP network at irregular intervals. Occasionally, the voice packets are out of sequence. The jitter buffer holds the packets briefly, reorders them if necessary, and then plays them out at evenly spaced intervals to the decoder in the DSP on the gateway. Algorithms in the DSP determine the size and behavior of the jitter buffer based on user configuration and current network jitter conditions. The DSP uses this information to maximize the number of correctly delivered packets and minimize the amount of delay.

The size of the jitter buffer and the amount of delay is configurable by the user with the **playout-delay** command. Proper configuration is critical. If voice packets are held for too short a time, variations in delay may cause the buffer to underrun (become empty) and cause gaps in speech. However, packets that arrive at a full buffer are dropped, also causing gaps in speech.

To improve voice quality, the speech gaps are hidden by several different techniques that synthesize packets to replace those that were lost or not received in time. Depending on the contiguous duration of the gaps, the missing voice frames are replaced by prediction from the past frames (usually the last frame), followed by silence if the condition persists (for more than 30 to 50 ms, for example). The **show call active voice** command output gives buffer overflow and concealment statistics, which are a good indication of the network effect on audio quality.

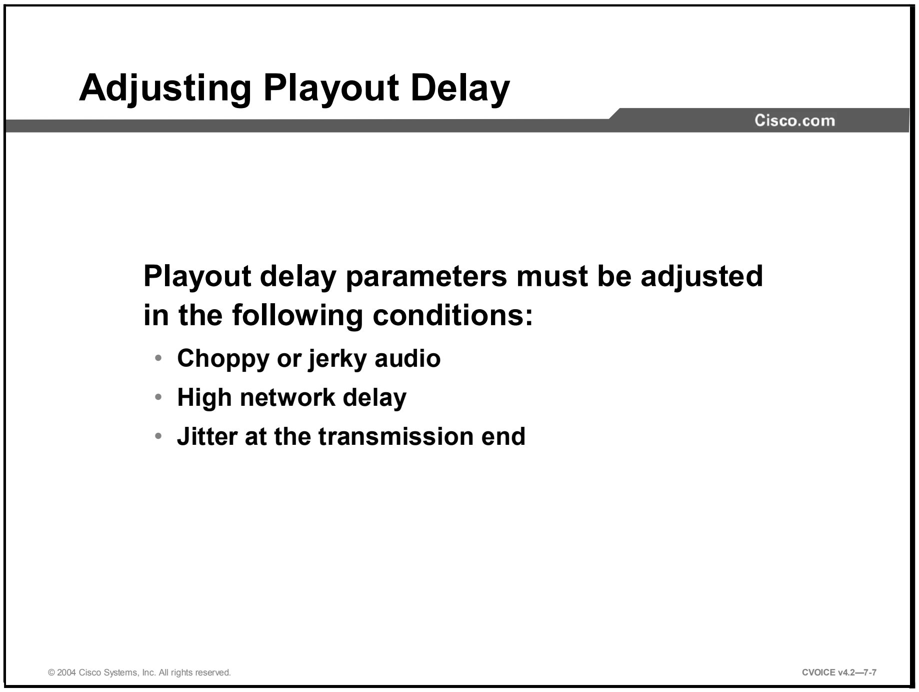
Example: Overcoming Jitter

In an example that demonstrates how packets can be lost, a jitter buffer is configured with a maximum playout delay of 40 ms. On the network, packets are delayed from their source; perhaps a media server stops sending packets for 60 ms, or there is severe network congestion. The jitter buffer empties while waiting for input from the network. Input does not arrive until *after* the maximum playout delay time is reached and there is a noticeable break in voice transmission. Now, the media server sends packets *to* the jitter buffer at a faster rate than the packets *leave* the jitter buffer; this makes the jitter buffer fill up. The jitter buffer discards subsequent packets, resulting in a choppy voice signal.

Even though the size of the jitter buffer is configurable, it is important to note that if the buffer size is too large, the overall delay on the connection may rise to unacceptable levels. You must weigh the benefit of improving jitter conditions against the disadvantage of increasing total end-to-end delay, which can also cause voice quality problems.

Adjusting Playout Delay Parameters

This topic lists the symptoms that lead to adjusting playout delay parameters.



The conditions that require you to adjust playout delay parameters are as follows:

- **Choppy or jerky audio:** Gaps in speech patterns that produce choppy or jerky audio suggest that you should increase the minimum playout delay, increase the maximum playout delay, or both, if you are using adaptive mode. For fixed mode, you must increase the nominal value.
- **High network delay:** High overall network delay suggests that you should reduce the maximum playout delay in adaptive mode, or reduce the nominal delay in fixed mode. You must watch for loss of voice quality. The maximum delay value sets an upper limit on adaptive playout delay, which in many cases is the major contributor to end-to-end delay. In many applications, it may be preferable to have the system or the user terminate the call, rather than allow an arbitrarily large delay. The data received with jitter outside this limit will show up in the late packet count in the **show call active voice** playout statistics.
- **Jitter at the transmission end:** A noisy but well-understood network or interworking with an application that has lots of jitter at the transmission end, from a source such as a unified messaging server or interactive voice response (IVR) application, suggests selection of fixed mode.

Symptoms of Jitter on a Network

This topic provides examples of output for the **show call active voice** command, which can be used to determine the size of jitter problems.

Symptoms of Jitter

Cisco.com

```
Router# show call active voice

<output omitted>

VOIP:
 ConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 IncomingConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 RemoteIPAddress=192.168.100.101
 RemoteUDPPort=18834
 RoundTripDelay=11 ms
 SelectedQoS=best-effort
 tx_DtmfRelay=inband-voice
 FastConnect=TRUE

 Separate H245 Connection=FALSE

 H245 Tunneling=FALSE
```


CVOICE v4.2—7-8

Symptoms of Jitter (Cont.)

Cisco.com

```
SessionProtocol=cisco
 SessionTarget=
 OnTimeRvPlayout=417000
 GapFillWithSilence=850 ms

 GapFillWithPrediction=2590 ms

 GapFillWithInterpolation=0 ms
 GapFillWithRedundancy=0 ms
 HiWaterPlayoutDelay=70 ms
 LoWaterPlayoutDelay=29 ms
 ReceiveDelay=39 ms

 LostPackets=0
 EarlyPackets=0
 LatePackets=86
```


CVOICE v4.2—7-9

The figure shows sample output for the **show call active voice** command. Several fields in the **show call active voice** command output that can help you to determine the actual size of the jitter problems are as follows:

- **ReceiveDelay:** The playout delay for jitter compensation plus the average expected delay after the frame is available for playout to the decoder. The current low-water mark and high-water mark statistics for the receive delay are available in the output.
- **GapFillWith:** These fields refer to the amount of *concealment*—or packet synthesizing—that took place in this call, to replace the voice packets that were lost or not received in time.
- **LostPackets:** The actual number of packets that were lost; that is, the packets *not* received at the egress gateway. This is detected using the **sequence number** field in the RTP packets.
- **EarlyPackets:** The actual number of packets that arrived *earlier* than the current minimum delay packet. They cause the dejitter algorithm to readjust the minimum delay packet used in jitter estimation.
- **LatePackets:** The actual number of packets that arrived later than the current playout delay setting. The information in these packets is discarded.

Average Jitter Statistics

Cisco.com

```
# show call active voice

<output omitted>
.
.
.
VOIP:
 ConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 IncomingConnectionId[0xECDE2E7B 0xF46A003F 0x0 0x47070A4]
 RemoteIPAddress=192.168.100.101
 RemoteUDPPort=18834
 RoundTripDelay=26 ms
 SelectedQoS=best-effort
 tx_DtmfRelay=inband-voice
 FastConnect=TRUE

 Separate H245 Connection=FALSE

 H245 Tunneling=FALSE
```

 CVOICE v4.2—7-10

Average Jitter Statistics (Cont.)

Cisco.com

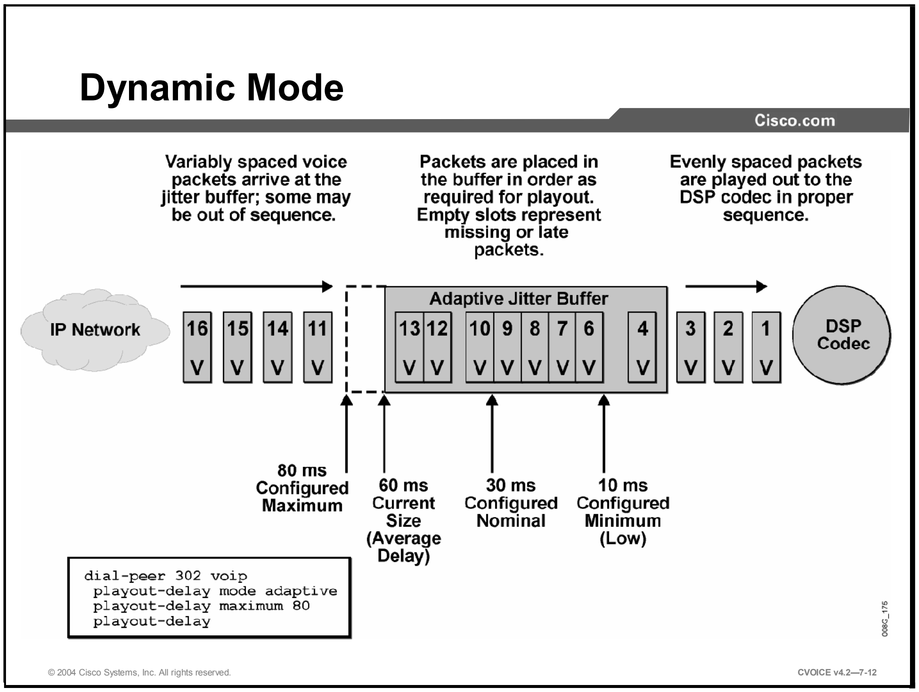
```
 SessionProtocol=cisco
 SessionTarget=
 OnTimeRvPlayout=482350
 GapFillWithSilence=1040 ms     <------------  Increased
 GapFillWithPrediction=3160 ms  <------------  Increased
 GapFillWithInterpolation=0 ms
 GapFillWithRedundancy=0 ms
 HiWaterPlayoutDelay=70 ms
 LoWaterPlayoutDelay=29 ms
 ReceiveDelay=43 ms             <------------  Increased
 LostPackets=0
 EarlyPackets=0
 LatePackets=105                <------------  Increased
```

 CVOICE v4.2—7-11

The sample output in this figure displays average jitter statistics when poor voice quality was perceived on the network. The GapFillWithSilence line indicates that too many consecutive packets were lost or late, and the DSP could not predict and fill in the gaps. The GapFillWithPrediction line indicates that packets were late or lost, and the DSP filled in the lost audio with prediction. The ReceiveDelay line indicates the average one-way delay the packets are experiencing as per the time stamps in the RTP header. The LatePackets line indicates the number of packets that were too late to be processed by the DSP. As these fields increase, the playout delay buffer should be increased in size.

Dynamic Jitter Buffer

This topic describes the dynamic jitter buffer mode.



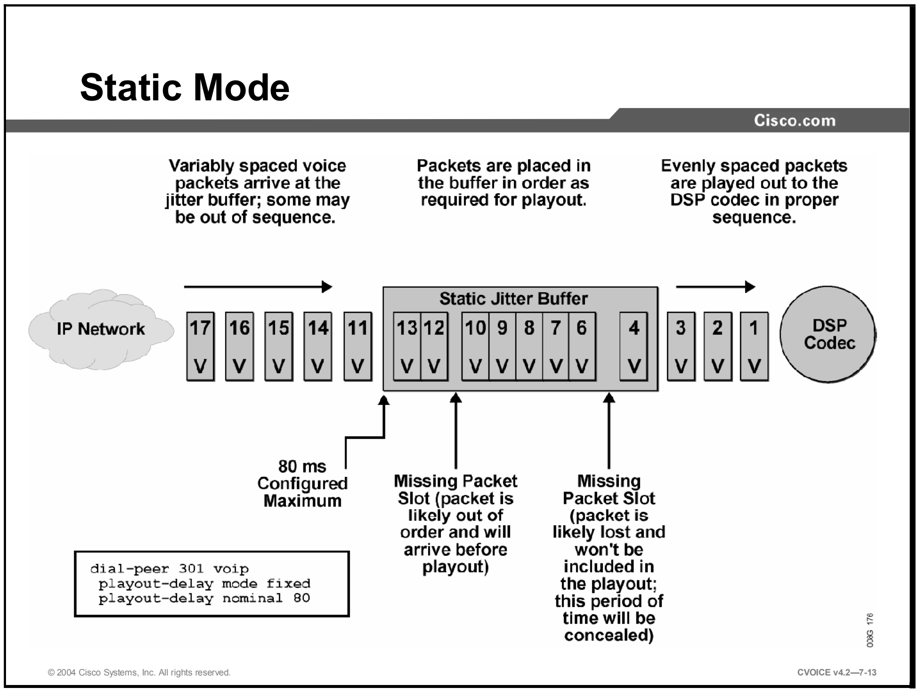
The **playout-delay** command allows you to select a jitter buffer mode (**static** or **dynamic**) and specify certain values that are used by the DSP algorithms to adjust the size of the jitter buffer. During a voice call, the algorithms read time stamps in the RTP headers of sample packets to determine the amount of delay that the jitter buffer will apply to an average packet; that is, as if there is no jitter at all in the network. This is called the average delay.

Example: Dynamic Jitter Buffer

When you configure the **playout-delay mode adaptive** option, the DSP algorithms in the codec take samples throughout the voice call and adjust the value of the average delay as network jitter conditions change. The size of the jitter buffer and the amount of delay applied are adjusted upward or downward, as needed. This adjustment ensures the smooth transmission of voice frames to the codec within the minimum and maximum limits that you configure. The algorithms are designed to *slowly* reduce the amount of delay and *quickly* increase the amount of delay during adjustment. As a result, voice quality is achieved at the risk of longer delay times.

Static Jitter Buffer

This topic describes the static mode buffer.



When you configure the **playout-delay mode fixed** option, you can specify the nominal delay value, which is the amount of playout delay applied at the beginning of a call by the jitter buffer. This is also the maximum size of the jitter buffer throughout the call.

Example: Static Jitter Buffer

Configuring static or "fixed" playout delay limits the size of the buffer. The figure shows an example of adjusting the nominal size of the buffer to 80 ms. Care should be taken when using the jitter buffer in static mode. Variations in arrival times of voice packets may be put at risk when network conditions change. In addition, if the static jitter buffer is configured to be too large, overall one-way delay is increased, exacerbating delay and echo problems.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

Cisco.com

- **The playout delay buffer and Packet Loss Concealment service are ways to compensate for jitter, a variation in the delay of received packets, that causes problems for audio playback at the receiving end.**
- **The playout delay buffer receives and reorders, if necessary, voice packets, plays them out to the decoder in the DSP, and the DSP uses information from the playout to maximize the number of correctly delivered packets and minimize delay.**
- **Symptoms of jitter include gaps in speech, high network delay, and jitter at the transmission end. These symptoms require adjustment of the playout delay parameters.**
- **Fields in the output for the** show call active voice **command help determine the size of the jitter.**
- **You can configure the dynamic jitter buffer mode with the** playout-delay mode adaptive **option.**
- **You can configure the static jitter buffer mode with the** playout-delay mode fixed **option.**

 CVOICE v4.2—7-14

References

For additional information, refer to these resources:

- *Playout Delay Enhancements for Voice Over IP*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121newft/121t/121t5/dt_pod.htm
- *Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)*
  http://www.cisco.com/warp/public/788/voice-qos/jitter_packet_voice.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which device buffers voice packets that arrive with variable delay and plays them out in a steady stream?

A) codec

B) DSP

C) modem

D) playout delay buffer

Q2) Which Cisco IOS service is used to synthesize packets to replace lost packets?

A) DSP

B) IVR

C) PLC

D) switch

Q3) Which device maximizes the number of correctly delivered packets and minimizes delay?

A) codec

B) DSP

C) FXS interface

D) switch

Q4) When looking at the output from a **show call active voice**, what fields refer to the amount of concealment or packet synthesizing that took place during a call?

A) **ReceiveDelay**

B) **GapFillWith**

C) **LostPackets**

D) **LatePackets**

E) **EarlyPackets**

Q5) How should you adjust the **playout-delay** parameters for fixed mode if there are gaps in speech?

A) increase the nominal delay

B) increase the minimum playout delay

C) increase the maximum playout delay

D) increase both the minimum and maximum playout delay

Q6) What change should you make to the **playout-delay** configuration if you have excessive jitter at the transmission end?

A) reduce the maximum playout delay

B) increase the minimum playout delay

C) select fixed mode

D) select adaptive mode

Q7) What happens to packets that arrive at the router later than the playout delay setting?

A) They are buffered and sent out in a steady stream.

B) The information in the packets is discarded.

C) They are placed at the head of the egress queue to reduce delay.

D) The packets are revitalized.

Q8) Which command can be used to determine the size of jitter problems?

A) **show jitter**

B) **show call jitter**

C) **show call active voice**

D) **show active voice call**

Q9) Which command is used to enable the codecs to dynamically adjust the value of the average delay depending on network conditions?

A) **playout-delay mode adaptive**

B) **playout-delay mode adjust**

C) **playout-delay mode auto**

D) **playout-delay mode dynamic**

Q10) What information do the DSP algorithms use to determine the amount of delay the jitter buffer applies to a packet?

A) average expected delay after the frame is available for playout

B) statistics from the **show** command used to determine jitter problems

C) sequence number field in the RTP packet

D) time stamps in the RTP headers of sample packets

Q11) In fixed mode, what does the nominal delay value signify?

A) the minimum amount of delay that is configurable for each voice packet

B) the amount of delay that is applied by the jitter buffer to each packet

C) the amount of playout delay that is applied by the jitter buffer at the beginning of a call

D) the minimum amount of delay that is applied by the jitter buffer for the duration of the call

Q12) In which mode can you specify the nominal delay value?

A) auto mode

B) adaptive mode

C) fixed mode

D) adaptive and fixed mode

Q13) What causes the dejitter buffer algorithm to readjust the minimum delay packet used in jitter estimation?

A) LatePackets

B) DelayedPackets

C) LostPackets

D) EarlyPackets

E) ReceiveDelay

Quiz Answer Key

Q1) D

Relates to: Understanding Jitter

Q2) C

Relates to: Understanding Jitter

Q3) B

Relates to: Overcoming Jitter

Q4) B

Relates to: Overcoming Jitter

Q5) A

Relates to: Adjusting Playout Delay Parameters

Q6) C

Relates to: Adjusting Playout Delay Parameters

Q7) B

Relates to: Symptoms of Jitter on a Network

Q8) C

Relates to: Symptoms of Jitter on a Network

Q9) A

Relates to: Dynamic Jitter Buffer

Q10) D

Relates to: Dynamic Jitter Buffer

Q11) C

Relates to: Static Jitter Buffer

Q12) C

Relates to: Static Jitter Buffer

Q13) D

Relates to: Symptoms of Jitter on a Network

Understanding Delay

Overview

When designing networks that transport voice over packet, frame, or cell infrastructures, it is important to understand and account for the delay components in the network to ensure acceptable network performance. Overall voice quality is a function of many factors, including the compression algorithm, errors and frame loss, echo cancellation, and delay. This lesson explains the sources of delay when using Cisco routers and gateways over packet networks and proposes solutions for this problem.

Relevance

Delay is the worst enemy of VoIP networks. Network administrators must have a solid understanding of the causes of delay and the solutions for this problem to successfully implement a VoIP network.

Objectives

Upon completing this lesson, you will be able to calculate, minimize, and verify delay on Cisco routers. This includes being able to meet these objectives:

- Describe the purpose of a delay budget and how delay is measured across a VoIP network
- Use ITU Recommendation G.114 to identify acceptable and unacceptable delay
- Describe the six major factors that contribute to voice packet delay
- Identify best-case and worst-case delays that are associated with different types of coders
- Calculate serialization delay based on packet size and line speed
- Use the **frame-relay fragment** *fragment_size* command to configure fragment size
- Use the delay budget to calculate one-way delay

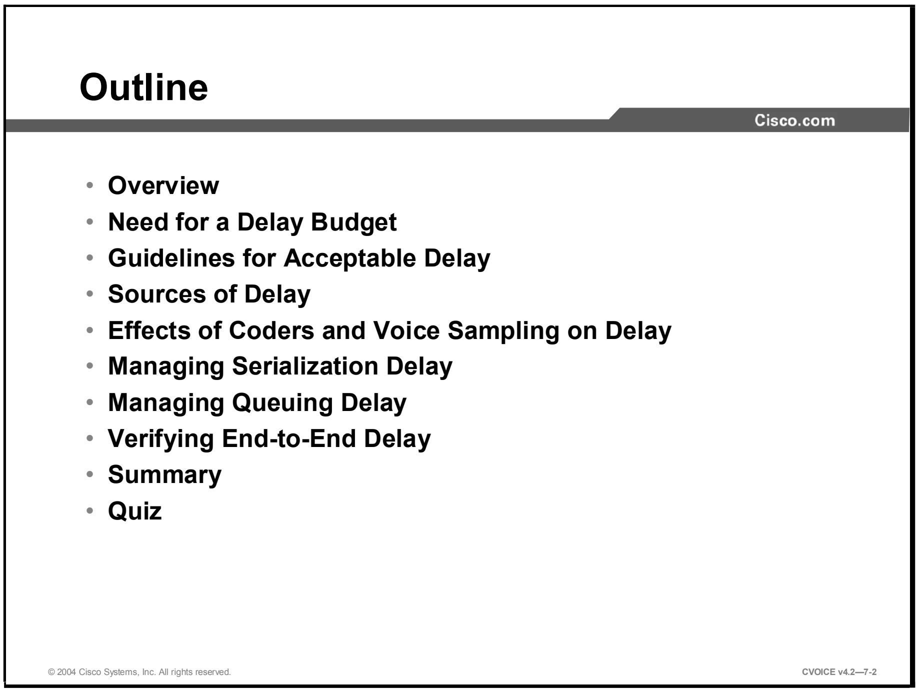
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of traditional telephony networks
- Basic knowledge of VoIP networks
- Knowledge of TCP/IP networks

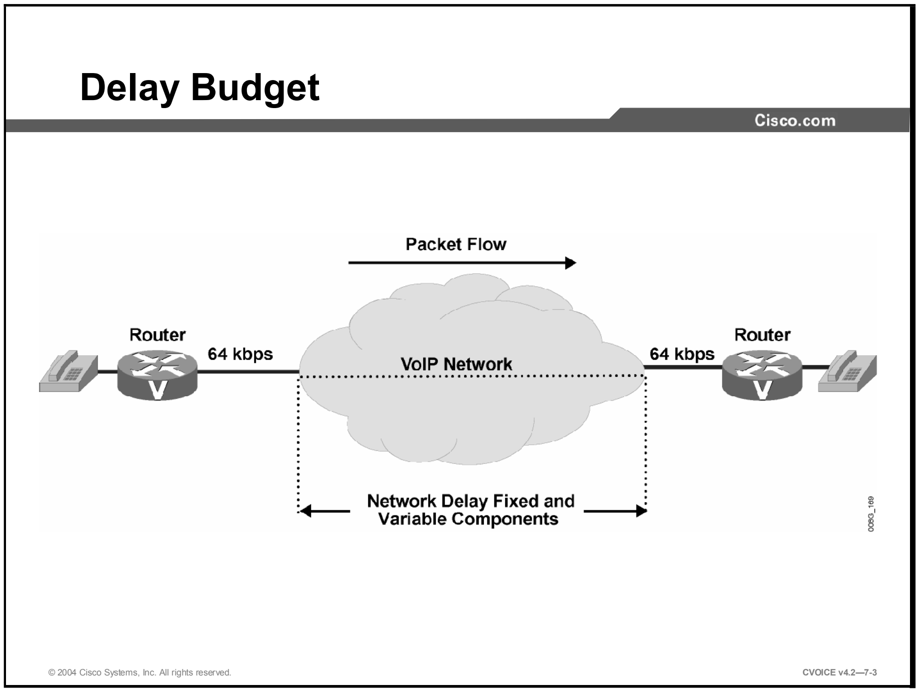
Outline

The outline lists the topics included in this lesson.



Need for a Delay Budget

The end-to-end delay in a VoIP network is known as the delay budget. Network administrators must design a network to operate within an acceptable delay budget. This topic explains the concept of delay budget and how to measure it.



Delay is the accumulated latency of end-to-end voice traffic in a VoIP network. The purpose of a delay budget is to ensure that the voice network does not exceed accepted limits of delay for voice telephony conversation. The delay budget is the sum of all the delays, fixed and variable, that are found in the network along the audio path. You can measure the delay budget by adding up all of the individual contributing components, as shown in the figure. The delay budget is measured in each direction individually, not round-trip.

Network administrators must be aware that delay exists, and then design their network to bring end-to-end delay within acceptable limits.

Example: Need For a Delay Budget

As delay increases, talkers and listeners become unsynchronized and often find themselves speaking at the same time or both waiting for the other to speak. This condition is commonly called *talker overlap*. While the overall voice quality may be acceptable, users may find the stilted nature of the conversation unacceptably annoying. Talker overlap may be observed on international telephone calls that travel over satellite connections. Satellite delay is about 500 ms: 250 ms up and 250 ms down.

Guidelines for Acceptable Delay

International telephony communications must adhere to a delay standard. This topic defines the standard and its limits.

Acceptable Delay: G.114

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Range in Milliseconds	Description
0 to 150	Acceptable for most user applications
150 to 400	Acceptable, provided that administrators are aware of the transmission time and its impact on the transmission quality of user applications
Above 400	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded

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CVOICE v4.2—7-4

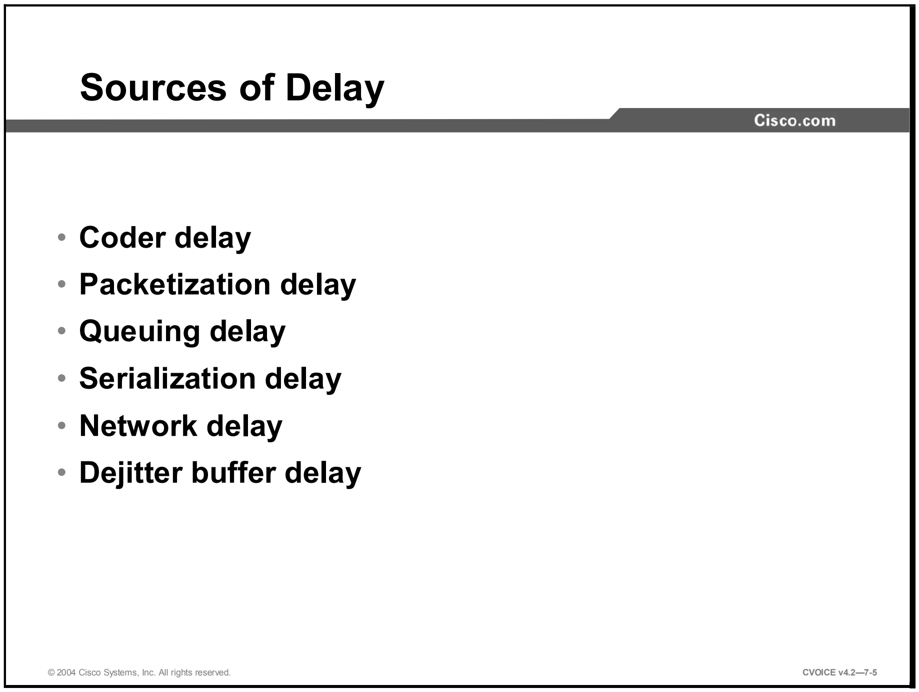
The ITU addresses network delay for voice applications in Recommendation G.114. This recommendation is oriented to national telecommunications administrations and is more stringent than what is normally applied in private voice networks. When the location and business needs of end users are well known to the network designer, more delay may prove acceptable.

Example: Acceptable Delay

As shown in the figure, acceptable delay time is from 0 to 400 ms for private networks. Delay times that exceed 400 ms are unacceptable for general network planning. However, all networks should be engineered to recognize and minimize the voice-connection delay. For example, suppose an enterprise is doing the planning for a new IP telephony roll-out. They plan and design to meet the 150-ms one-way delay, but have some locations in which 250-ms one-way delay is the best they can do. Since most of these calls will be on-net calls, the organization has to decide if the lower quality of the calls is acceptable for these locations.

Sources of Delay

Many factors add to overall delay. This topic lists six factors that are sources of delay.



Following is an explanation of six major factors that contribute to overall fixed and variable delay:

- **Coder delay:** Also called processing delay, coder delay is the time taken by the DSP to compress a block of pulse code modulation (PCM) samples. Because different coders work in different ways, this delay varies with the voice coder that is used and the processor speed.
- **Packetization delay:** Packetization delay is the time it takes to fill a packet payload with encoded or compressed speech. This delay is a function of the sample block size that is required by the vocoder and the number of blocks placed in a single frame. Packetization delay is also called accumulation delay because the voice samples accumulate in a buffer before being released. With typical payload sizes used on Cisco routers, packetization delay for G.711, G.726, and G.729 does not exceed 30 ms.
- **Queuing delay:** After the network builds a compressed voice payload, it adds a header and queues for transmission on the network connection. Because voice should have absolute priority in the router or gateway, a voice frame must wait only for a data frame already playing out or for other voice frames ahead of it. Essentially, the voice frame waits for the serialization delay of any preceding frames that are in the output queue. Queuing delay is a variable delay and is dependent on the trunk speed and the state of the queue.
- **Serialization delay:** Serialization delay is the fixed delay that is required to clock a voice or data frame onto the network interface; it is directly related to the clock rate on the trunk.

- **Network delay:** The public Frame Relay or ATM network that interconnects the endpoint locations is the source of the longest voice-connection delays. These delays are also the most difficult to quantify. If a private enterprise builds its own internal Frame Relay network for the purpose of wide-area connectivity, it is possible to identify the individual components of delay. In general, the fixed components are from propagation delays on the trunks within the network; variable delays are the result of queuing delays that clock frames into and out of intermediate switches. To estimate propagation delay, a popular estimate of 10 microseconds/mile or 6 microseconds/km (G.114) is widely used, although intermediate multiplexing equipment, backhauling, microwave links, and other features of carrier networks create many exceptions. Typical carrier delays for U.S. Frame Relay connections are 40 ms fixed, and 25 ms variable, for a total worst-case delay of 65 ms.
- **Dejitter buffer delay:** Because speech is a constant bit-rate service, the jitter from all the variable delays must be removed before the signal leaves the network. In Cisco routers and gateways, this is accomplished with a dejitter buffer at the far-end (receiving) router or gateway. The dejitter buffer transforms the variable delay into a fixed delay by holding the first sample that is received for a period of time *before* playing it out. This holding period is known as the initial playout delay. The actual contribution of the dejitter buffer to delay is the initial playout delay of the dejitter buffer *plus* the actual amount of delay of the first packet that was buffered in the network. The worst case would be twice the dejitter buffer initial delay (assuming the first packet through the network experienced only minimal buffering delay).

Effects of Coders and Voice Sampling on Delay

The process of encoding an analog voice sample into a compressed digitized bit stream contributes to delay. This topic describes the effect of coder delay. Best-case and worst-case coder delays are shown in the figure.

Coder Delay

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Coder	Rate	Required Sample Block	Best-Case Coder Delay	Worst-Case Coder Delay
ADPCM, G.726	32 kbps	10 ms	2.5 ms	10 ms
CS-ACELP, G.729A	8.0 kbps	10 ms	2.5 ms	10 ms
MPMLQ, G.723.1	6.3 kbps	30 ms	5 ms	20 ms
MP-ACELP, G.723.1	5.3 kbps	30 ms	5 ms	20 ms

CVOICE v4.2—7-6

The compression time for a Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP) process ranges from 2.5 to 10 ms, depending on the loading of the DSP. If the DSP is fully loaded with four voice channels, the coder delay will be 10 ms. If the DSP is loaded with one voice channel only, the coder delay will be 2.5 ms. For design purposes, use the worst-case time of 10 ms.

Decompression time is roughly 10 percent of the compression time for each block. However, because there may be multiple samples in each frame, the decompression time is proportional to the number of samples per frame. Consequently, the worst-case decompression time for a frame with three samples is 3 x 1 ms, or 3 ms. Generally, two or three blocks of compressed G.729 output are put in one frame, while only one sample of compressed G.723.1 output is sent in each frame.

Example: Coder Delay

The figure shows examples of some coder delays. CS-ACELP, for example, lists a best-case delay of 2.5 ms with a worst-case delay of 10 ms. In all calculations, the worst-case number should be used.

Managing Serialization Delay

An important part of delay is the serialization delay on the interface. This topic describes serialization delay and its management.

Serialization Delay

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Frame Size (bytes)	Line Speed (kbps)										
	19.2	56	64	128	256	384	512	768	1024	1544	2048
38	15.83	5.43	4.75	2.38	1.19	0.79	0.59	0.40	0.30	0.20	0.15
48	20.00	6.86	6.00	3.00	1.50	1.00	0.75	0.50	0.38	0.25	0.19
64	26.67	9.14	8.00	4.00	2.00	1.33	1.00	0.67	0.50	0.33	0.25
128	53.33	18.29	16.00	8.00	4.00	2.67	2.00	1.33	1.00	0.66	0.50
256	106.67	36.57	32.00	16.00	8.00	5.33	4.00	2.67	2.00	1.33	1.00
512	213.33	73.14	64.00	32.00	16.00	10.67	8.00	5.33	4.00	2.65	2.00
1024	426.67	146.29	128.00	64.00	32.00	21.33	16.00	10.67	8.00	5.31	4.00
1500	625.00	214.29	187.50	93.75	46.88	31.25	23.44	15.63	11.72	7.77	5.86
2048	853.33	292.57	256.00	128.00	64.00	42.67	32.00	21.33	16.00	10.61	8.00

Serialization Delay in ms

CVOICE v4.2—7-7

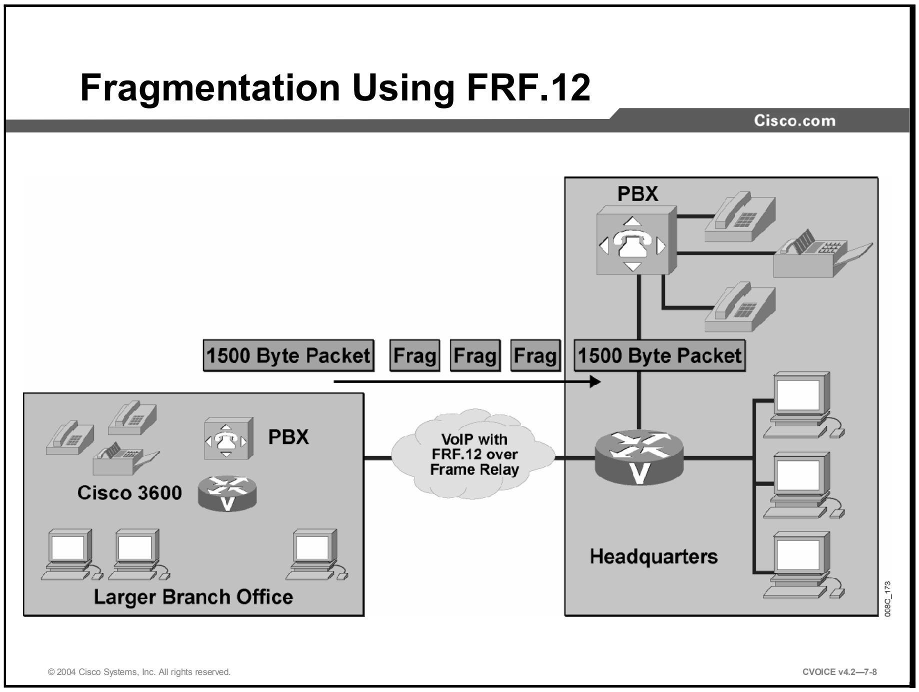
The figure shows the serialization delay required for different frame sizes at various line speeds. This table uses total frame size, not payload size, for computation. As an example, reading from the graphic, on a 64-kbps line, a CS-ACELP voice frame with a length of 38 bytes (37 bytes + 1-byte flag) has a serialization delay of 4.75 ms. If the line speed is increased to 1.544 Mbps, the serialization delay goes down to 0.2 ms. Cisco recommends a 10-ms serialization delay, not to exceed 20 ms.

You can effectively change the serialization delay of a voice packet by performing the following:

- **Increasing the link speed:** Solves the problem, but is expensive
- **Decreasing the packet size:** May not be possible for all codec types, and also increases bandwidth overhead

Managing Queuing Delay

Queuing delay contributes to overall delay. This topic explains queuing delay and offers a standard solution.



Queuing delay occurs when other elements in the outbound queue (voice or data packets) cause voice packets to be delayed. For example, the serialization delay for a 1500-byte packet is 214 ms to leave the router over a 56-kbps link. If a 1500-byte data packet that is not real-time is being sent, real-time (voice) packets are queued until the large data packet is transmitted. This delay is unacceptable for voice traffic. If data packets that are not real-time packets are fragmented into smaller frames, they are interleaved with real-time (voice) frames. In this way, both voice and data frames can be carried together on low-speed links without causing excessive delay to the real-time voice traffic. One way to implement this fragmentation is to use FRF.12 on VoIP over Frame Relay networks. FRF.12 serves to fragment Frame Relay frames into smaller frames, even from different permanent virtual circuits (PVCs).

Example: Fragment Size Configuration

You can configure fragment size using the **frame-relay fragment** *fragment_size* command in a Frame Relay map class. The *fragment_size* argument defines the payload size of a fragment and excludes the Frame Relay headers and any Frame Relay fragmentation header. The valid range is from 16 bytes to 1600 bytes; the default is 53 bytes.

The *fragment_size* argument should be set so that the serialization delay is close to 10 ms; for example, if using a 384-kbps link, the fragmentation size should be set at 512 kbps.

Set the fragmentation size so that the largest data packet is not larger than the voice packets.

Verifying End-to-End Delay

End-to-end delay is calculated and compared to the G.114 recommendation. This topic illustrates the process.

Verifying End-to-End Delay

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Delay Type	Fixed (ms)	Variable (ms)
Coder delay	18	
Packetization delay	30	
Queuing/buffering		8
Serialization delay (64 kbps)	5	
Network delay (public frame)	40	25
Dejitter buffer delay	45	
Totals	138	33

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CVOICE v4.2—7-9

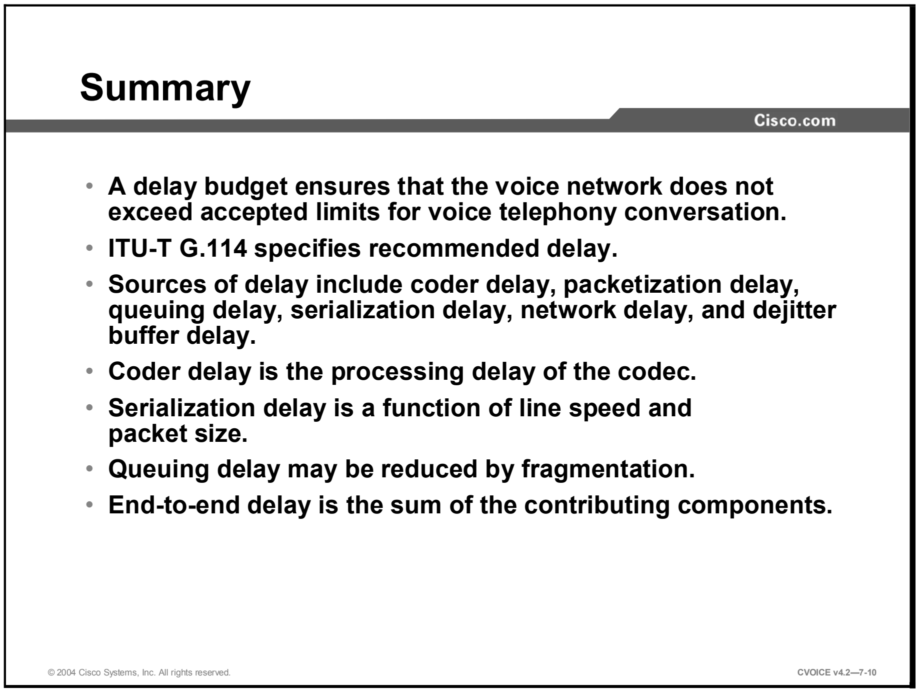
A typical one-hop connection over a public Frame Relay connection may have the delay budget that is shown in the figure. To calculate one-way delay, simply add all of the contributing components together. The goal is to allow a one-way delay as recommended by G.114.

Example: Verifying End-To-End Delay

The figure shows an acceptable one-way delay of 138 ms, plus 33 ms, for a total of 171 ms.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to this resource:

- *Understanding Delay in Packet Voice Networks*
  http://www.cisco.com/warp/public/788/voip/delay-details.html#60

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) What is the purpose of a delay budget?

- A) to synchronize voice traffic
- B) to provide guaranteed delay for voice packets
- C) to ensure that voice traffic does not exceed accepted limits for delay
- D) to provide QoS services

Q2) What is the delay budget for a network?

- A) the sum of fixed and variable delay from end to end for each device
- B) the sum of the one-way fixed and variable delay
- C) the sum of the round-trip fixed and variable delay calculated between each device
- D) the sum of the fixed and variable delay between the egress and ingress router in both directions

Q3) According to ITU Recommendation G.114, what is the acceptable delay for national telecommunications administrations?

- A) 0 to 150 ms
- B) 0 to 400 ms
- C) 150 to 400 ms
- D) above 400 ms

Q4) According to ITU Recommendation G.114, what is the acceptable delay for private networks?

- A) 0 to 150 ms
- B) 0 to 400 ms
- C) 150 to 400 ms
- D) above 400 ms

Q5) Match the factor that contributes to delay with its description.

A) coder delay

B) packetization delay

C) serialization delay

D) dejitter buffer delay

_____ 1. time taken to fill a packet payload with encoded speech

_____ 2. time required to clock a frame onto the network interface

_____ 3. time taken by the DSP to compress a block of PCM samples

_____ 4. time a sample is held before being played out

Q6) Queuing delay is dependent on which two factors? (Choose two.)

A) dejitter buffer delay

B) serialization delay

C) packetization delay

D) trunk speed

E) state of the queue

Q7) Which type of delay is the most difficult to quantify?

A) dejitter buffer delay

B) queuing delay

C) serialization delay

D) propagation delay

E) packetization delay

F) coder delay

Q8) What will be the coder delay if a G.726 is loaded with four channels?

A) 2.5 ms

B) 5 ms

C) 10 ms

D) 20 ms

Q9) What is the decompression time for each sample block of voice traffic?

A) roughly twice the compression time for each block

B) roughly three times the compression time for each block

C) roughly 3 percent of the compression time for each block

D) roughly 10 percent of the compression time for each block

Q10) How much serialization delay is recommended by Cisco?

A) 0.2 ms

B) 0.5 ms

C) 10 ms

D) 20 ms

Q11) Identify two ways to reduce serialization delay. (Choose two.)

A) by increasing packet compression

B) by reducing the processing delay

C) by increasing the link speed

D) by prioritizing the packets

E) by decreasing the packet size

F) by reducing the queuing delay

Q12) Which three factors should you consider when setting the fragment size in the **frame-relay fragment** command? (Choose three.)

A) The payload size is between 16 and 1600 bytes.

B) The serialization delay is not more than 10 ms.

C) The largest data packet is not larger than the voice packets.

D) The Layer 3 access method is being used.

E) The delay associated with the link type is being used.

Q13) Which type of delay directly affects queuing delay?

A) coder delay

B) propagation delay

C) serialization delay

D) packetization delay

E) dejitter buffer delay

Q14) Which two factors contribute to fixed delay only? (Choose two.)

A) coder delay

B) network delay

C) queuing delay

D) packetization delay

Q15) Which ITU-T recommendation is considered to be a measure of acceptable end-to-end delay?

A) G.114

B) G.115

C) G.711

D) G.411

Quiz Answer Key

Q1) C

Relates to: Need for a Delay Budget

Q2) B

Relates to: Need for a Delay Budget

Q3) A

Relates to: Guidelines for Acceptable Delay

Q4) B

Relates to: Guidelines for Acceptable Delay

Q5) 1-B, 2-C, 3-A, 4-D

Relates to: Sources of Delay

Q6) D, E

Relates to: Sources of Delay

Q7) D

Relates to: Sources of Delay

Q8) C

Relates to: Effects of Coders and Voice Sampling on Delay

Q9) D

Relates to: Effects of Coders and Voice Sampling on Delay

Q10) C

Relates to: Managing Serialization Delay

Q11) C, E

Relates to: Managing Serialization Delay

Q12) A, B, C

Relates to: Managing Queuing Delay

Q13) C

Relates to: Managing Queuing Delay

Q14) A, D

Relates to: Verifying End-to-End Delay

Q15) A

Relates to: Verifying End-to-End Delay

Applying QoS in the Campus

Overview

This lesson covers introductory design considerations and recommendations for implementing QoS in a campus environment. For in-depth training on QoS in a campus environment, students are referred to Cisco *Quality of Service* courses.

Relevance

Until recently, many network experts believed that QoS would never be an issue in the campus because of bandwidth availability, buffer overflow, and the bursty nature of network traffic. Gradually, network administrators have realized that buffering, not bandwidth, is the dominant issue in the campus. For this reason, QoS tools are required to manage these buffers to minimize loss, delay, and delay variation.

Objectives

Upon completing this lesson, you will be able to configure QoS features on a campus network. This includes being able to meet these objectives:

- Identify the areas of a campus network that require QoS services
- Describe separation of queues as an approach to QoS management in a campus network
- Describe queue scheduling as an approach to QoS management in a campus network
- Describe marking control and management traffic as an approach to QoS management in a campus network
- Provide an example of a QoS configuration on a campus network
- Describe three queuing and scheduling methods
- Configure switchwide queuing and mark control and management traffic in a campus network

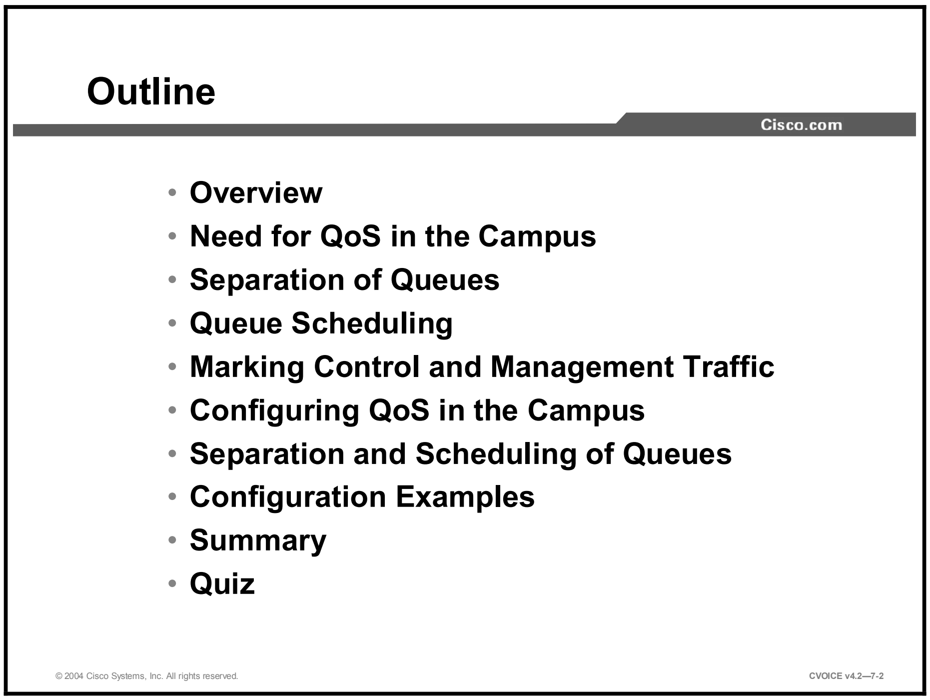
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Basic knowledge of VoIP concepts
- Knowledge of Cisco IOS software
- Knowledge of Cisco Catalyst switches
- Knowledge of campus networks

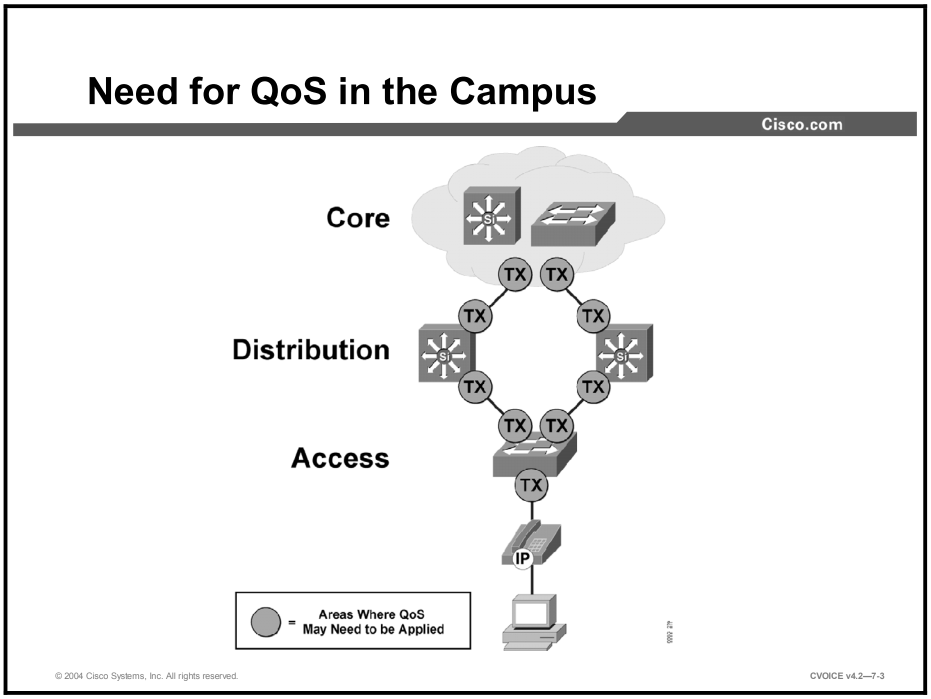
Outline

The outline lists the topics included in this lesson.



Need for QoS in the Campus

This topic describes the need for QoS measures in a campus network.



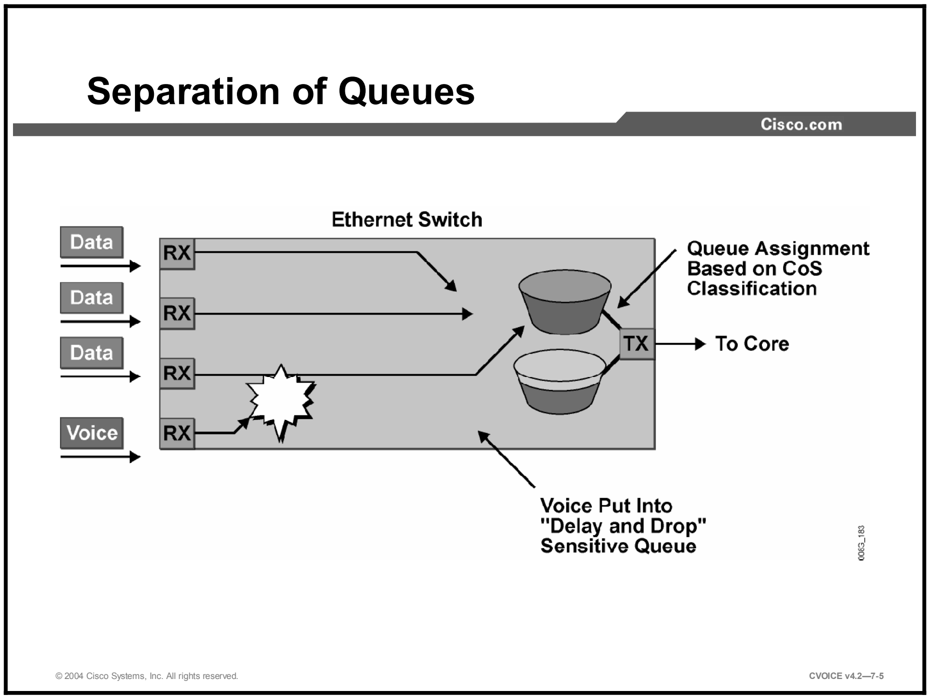
The bursty nature of data networks, combined with the high volume of smaller TCP packets, causes transmit buffers to fill to capacity in high-speed campus networks. If an output buffer fills, ingress interfaces are unable to place new flow traffic into the output buffer. When the ingress buffer is filled, which can happen quickly, packet drops occur. Typically, these drops are more than a single packet in any given flow. TCP packet loss causes retransmissions to occur, further aggravating the problem. Voice packet loss results in voice clipping and skips.

The figure illustrates the areas of a campus network where QoS is a concern. These areas are as follows:

- **Access ports:** Access-level ports that connect to IP Phones or gateways and the distribution switches
- **Distribution ports:** Uplink ports that connect to access switches and core switches
- **Core ports:** Uplink ports that connect the distribution switches to the core campus switches

Separation of Queues

This topic explains how you can use multiple queues to avoid dropped packets.



VoIP traffic is sensitive to both delayed packets and dropped packets. Even though a campus may be using gigabit Ethernet trunks, which have extremely fast serialization times, delay and dropped packets may still be an issue because of the type of packets, type of applications, and number of packets. However, drops *always* adversely affect voice quality in the campus. Using multiple queues on transmit interfaces is the only way to eliminate the potential for dropped traffic caused by buffers operating at 100 percent capacity. By separating voice (which is sensitive to delays and drops) into its own queue, you can prevent flows from being dropped at the ingress interface even if data flows fill up the data transmit buffer. If video is present, it should also have its own queue because it has similar delay characteristics.

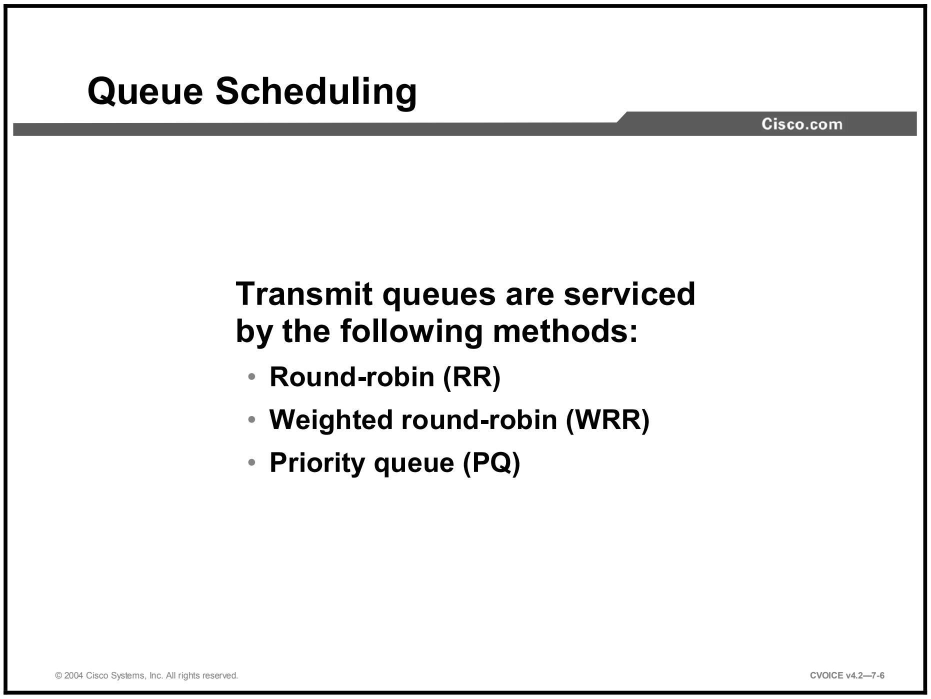
Caution It is critical that you verify that flow control is disabled when enabling QoS (multiple queues) on Catalyst switches. Flow control interferes with the configured queuing behavior by acting on the ports before activation of queuing. Flow control is disabled by default.

Example: Separation of Queues

The figure shows a voice packet that may be dropped if there is only one queue and it is full. By separating the voice and data queues, voice is put into its own delay-and-drop sensitive queue.

Queue Scheduling

Queues should schedule packet dispatching to ensure the emptying of queues in a fair and timely fashion. This topic describes queue scheduling as an approach to management of QoS in a campus network.



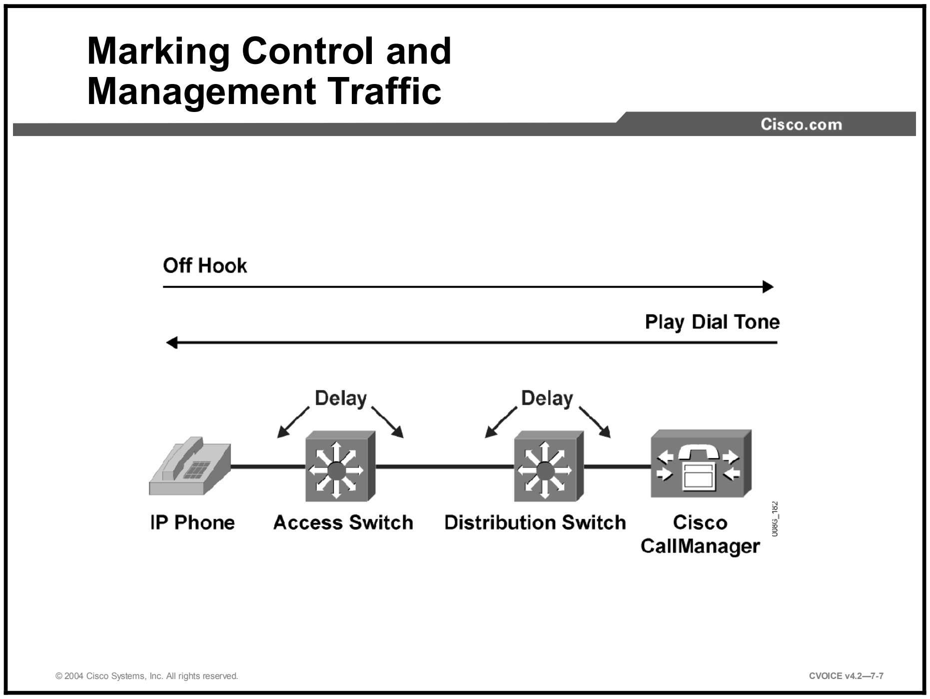
The scheduler process consists of a variety of methods to service each of the transmit queues (voice and data). The easiest method is a round-robin (RR) algorithm, which services queue 1 through queue *n* in a sequential manner. While not robust, this is an extremely simple and efficient method that you can use for branch office and wiring-closet switches. Distribution-layer switches use a weighted round-robin (WRR) algorithm in which higher-priority traffic is given a scheduling “weight.” WRR provides bandwidth to higher-priority applications using CoS bit settings and grants access to lower-priority queues. The frame schedule affords the bandwidth that each queue is allotted by the network administrator. This mapping is configurable, both at the system and interface levels. Care should be taken when configuring WRR, because it may affect bandwidth provided to other lower-priority queues.

Another option is to combine RR or WRR scheduling with priority scheduling for applications that are sensitive to packet delay and drop. This uses priority queuing (PQ) that is always served first when there are packets in the queue. If there are no frames in the PQ, the additional queues are scheduled using RR or WRR. You should take care when implementing a PQ, because it may starve traffic in lower queues if there is too much priority traffic.

The preferred method of queue scheduling is WRR.

Marking Control and Management Traffic

Marking control and management traffic is critical to VoIP network performance. Control and management traffic should be marked for preferential dispatching throughout the campus. This topic describes marking control and management traffic as an approach to the management of QoS in a campus network.



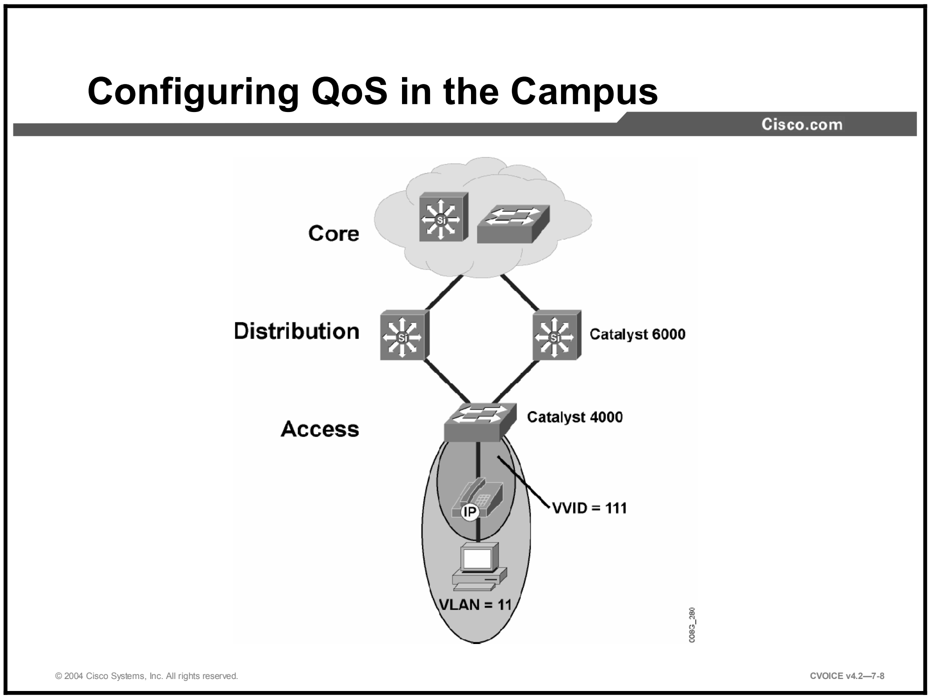
In networks with high traffic loads, managing the delivery of control traffic is critical to ensuring a positive user experience with VoIP. Delay to Dial-Tone (DTT) time is an example of this type of delivery management. When a Cisco IP Phone goes off hook, it "asks" Cisco CallManager for instructions. Cisco CallManager then instructs the Cisco IP Phone to play a dial tone. If this management and control traffic is dropped or delayed within the network, the user experience is adversely affected. The same logic applies to all signaling traffic for gateways and telephones.

Example: Marking Control and Management Traffic

To ensure that this control and management traffic is marked as important (but not as important as the actual RTP stream), access control lists (ACLs) are used to classify these streams on Catalyst 3500, 4000, and 6000 switches that are enabled for Layers 3 and 4.

Configuring QoS in the Campus

This topic describes how to configure QoS in a campus network.



The figure illustrates a network. Note that the access switch is a Catalyst 4000 and the distribution and core switches are Catalyst 6000. Also note that the Cisco IP Phone operates in voice VLAN ID (VVID) 111, while the workstation operates in voice VLAN 11. VLAN increases the quality of the voice traffic and allows a large number of telephones to be added to an existing network (in which there are not enough IP addresses), by isolating the telephones on a separate auxiliary. A new VLAN means a new subnet and a new set of IP addresses.

In the access-switch configuration, the distribution and core switches must also be configured; however, the configuration is very similar to the access-switch configuration.

Configuring a Voice VLAN

```
Router# configure terminal
Router(config)# interface fastethernet 5/1
Router(config-if)# switchport voice vlan 101
Router(config-if)# exit
```

 CVOICE v4.2—7-9

The figure shows how to configure Fast Ethernet port 5/1 to send Cisco Discovery Protocol (CDP) packets that tell the Cisco IP Phone to use VLAN 101 as the voice VLAN.

Verifying the Configuration

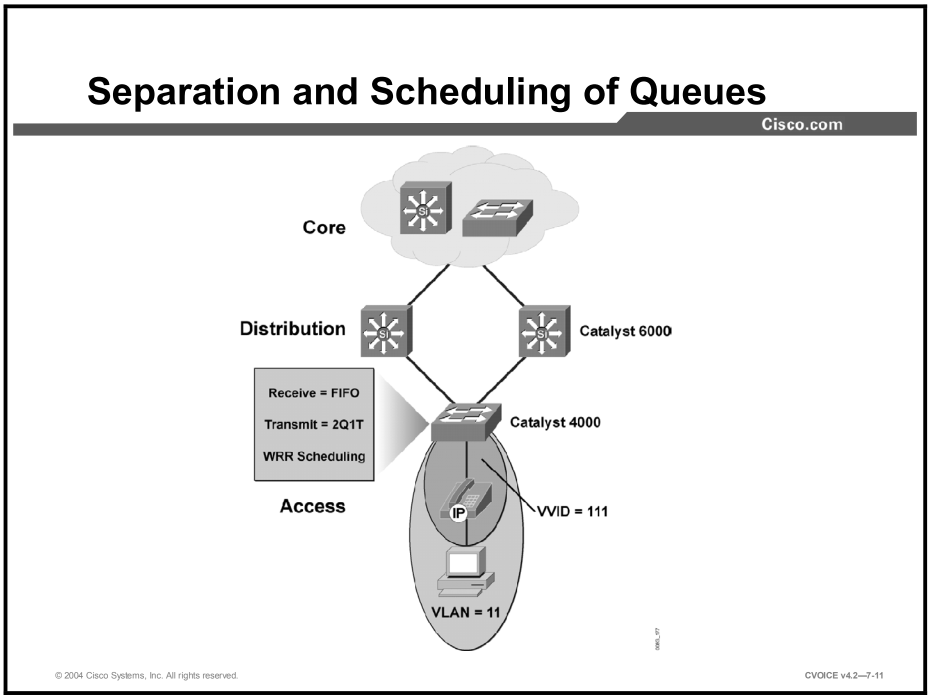
```
Router# show interfaces fastethernet 5/1 switchport
Name: Fa5/1
Switchport: Enabled
Administrative Mode: access
Operational Mode: access
Administrative Trunking Encapsulation: dot1q
Operational Trunking Encapsulation: dot1q
Negotiation of Trunking: off
Access Mode VLAN: 100
Voice VLAN: 101
Trunking Native Mode VLAN: 1 (default)
Administrative private-vlan host-association: none
Administrative private-vlan mapping: 900 ((Inactive)) 901
((Inactive))
Operational private-vlan: none
Trunking VLANs Enabled: ALL
Pruning VLANs Enabled: 2-1001
Capture Mode Disabled
Capture VLANs Allowed: ALL
```

 CVOICE v4.2—7-10

This example shows how to verify the configuration of Fast Ethernet port 5/1.

Separation and Scheduling of Queues

This topic presents several suggested configurations for port scheduling and queuing on the Catalyst 4000 at the access layer.



Switchwide Queuing

Configuration for switchwide queuing depends upon the type of interface used. The two types of interfaces used are:

- **Receive interface:** The recommended configuration for the receive interface is one standard FIFO queue.
- **Transmit interface:** The recommended configuration for the transmit interface is two standard queues with a single threshold (2Q1T). This method is preferred over other methods because it places traffic with specific CoS in one queue (as configured) and all other traffic in the second queue, thus simplifying configuration. The threshold indicates that traffic is dropped when the buffer capacity reaches 100 percent. Scheduling is done on a WRR basis. Admission to the queues is based on the 802.1p CoS value and is user-configurable in pairs. If you enable QoS, but do not modify the CoS-to-transmit queue mappings, switch performance could be affected because all traffic goes to queue 1.

Note	802.1p is a Layer 2 mechanism for classifying traffic using 3 bits. 802.1p adds 16 bits to the Layer 2 header.

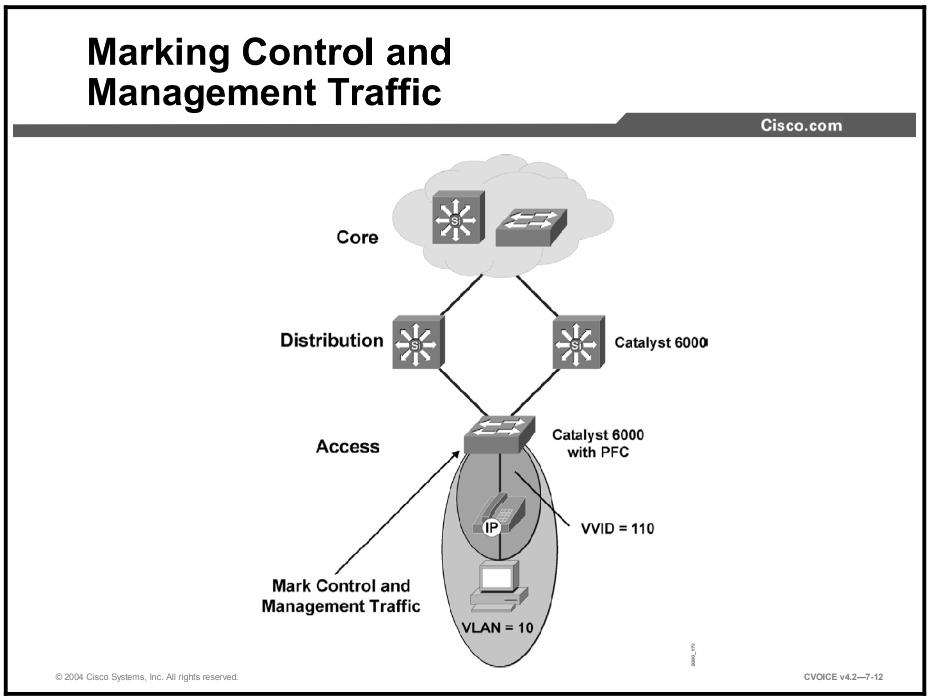
IP Phone Port Queuing

In Cisco Catalyst software Release 5.5.1, the Catalyst 4000 line does not offer any advanced IP Phone-queuing features. Because of this, the Catalyst 4000 depends on the default CoS marking and enforcement of the Cisco IP Phone. Other access switches, such as the Catalyst 3500, offer more advanced IP port phone-queuing options.

Uplink Interface to the Distribution Switch

You do not need to configure special queuing or scheduling commands on the Catalyst 4000 side of the link (from the access layer Catalyst 4000 to the distribution layer Catalyst 6000). Queuing is automatically enabled after QoS has been enabled and the classification and queue admission have been configured.

Marking Control and Management Traffic

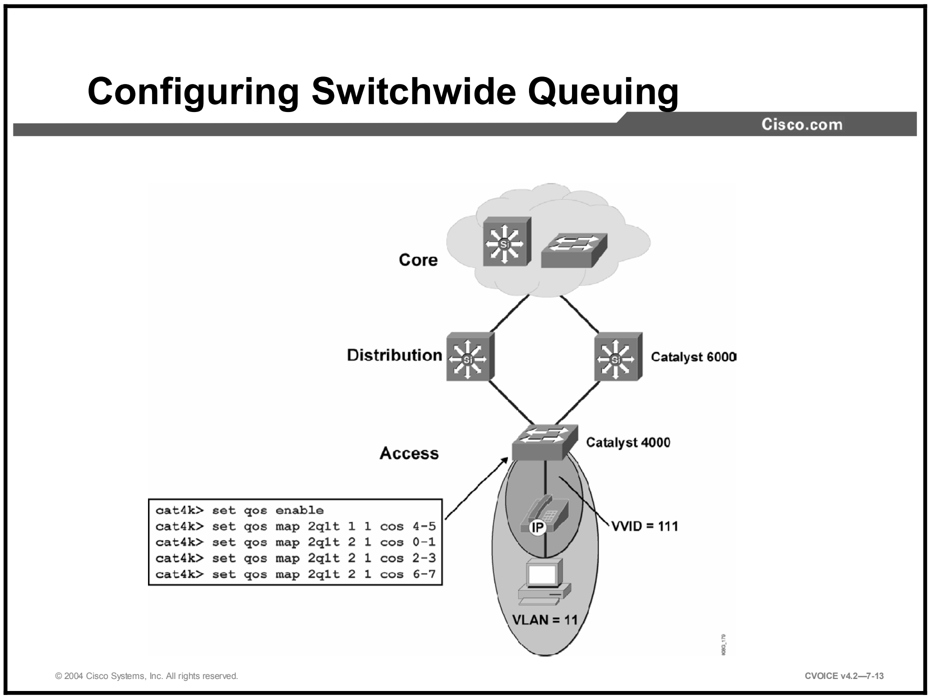


If the access switch is a Catalyst 5000 or 6000, you can classify control and management traffic in the campus. To ensure that this control and management traffic is marked as important (but not as important as the actual RTP stream), ACLs are used to classify these streams on Catalyst 5000 and 6000 switches that are enabled for Layers 3 and 4. You should enable this classification and marking for all capable switches in the campus.

The addition of the Policy Feature Card (PFC) means that the Catalyst 6000 is capable of handling Layer 2, 3, and 4 QoS issues. You can use the PFC to enable advanced QoS tools, such as packet classification and marking, scheduling, and congestion avoidance, based on either Layer 2, Layer 3, or Layer 4 header information. You can configure multiple receive and transmit queues with thresholds and use them according to the QoS policy rules that are configured in the switch.

Configuration Examples

This topic illustrates configuration examples for QoS in the campus network.



By default, only one queue is enabled on the Catalyst 4000 line of switches. Use the **set qos map** commands to enable the use of the second queue in CatOS Release 5.5.1. VoIP control frames (CoS = 3) should be placed into the second queue in the Catalyst 4000. These maps must be configured in pairs of CoS values, because the Catalyst 4000 examines the first two CoS bits only.

Verifying Queue Admission

Cisco.com

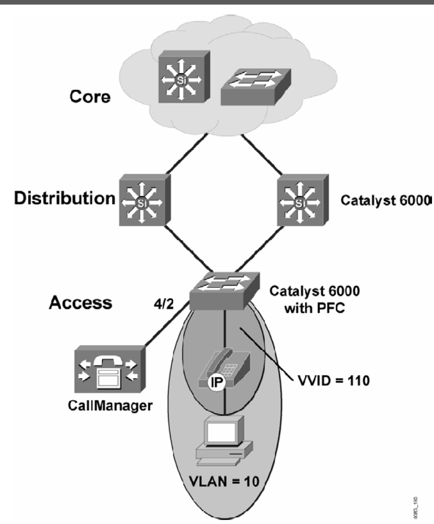
```
cat4k> (enable) show qos info runtime
Run time setting of QoS:
QoS is enabled
All ports have 2 transmit queues with 1 drop thresholds
(2q1t).
Default CoS = 0
Queue and Threshold Mapping:
Queue Threshold CoS
----- --------- ---------------
1     1         4 5
2     1         0 1 2 3 6 7
```

 CVOICE v4.2—7-14

Output in this figure shows verification of the queue admission configuration.

Configuring Marking of Control and Management Traffic

Cisco.com



 CVOICE v4.2—7-15

To demonstrate marking of control and management traffic, you must assume that the access switch is a Catalyst 6000. The VVID is 110 and the data VLAN is 10. Assume also that the network is using the Skinny Client Control Protocol (SCCP) and that the Cisco CallManager is on port 4/2.

Configuring Marking of Control and Management Traffic (Cont.)

Cisco.com

Catalyst 6000 Access Switch

```
cat6k-access> (enable) set qos enable
cat6k-access> (enable) set qos acl ip ACL_IP-PHONES dscp 26 tcp any any range 2000 2002
cat6k-access> (enable) set qos acl ip ACL_IP-PHONES trust-cos ip any any
cat6k-access> (enable) set qos acl ip ACL_VOIP_CONTROL dscp 26 tcp any any range 2000 2002
cat6k-access> (enable) set port qos 5/1-48 trust trust-cos
cat6k-access> (enable) set port qos 5/1-48 vlan-based
cat6k-access> (enable) set port qos 5/1-48 trust-ext untrusted
cat6k-access> (enable) set port qos 4/2 port-based
cat6k-access> (enable) commit qos acl all
cat6k-access> (enable) set qos acl map ACL_IP-PHONES 110
cat6k-access> (enable) set qos acl map ACL_VOIP_CONTROL 4/2
```


CVOICE v4.2—7-16

Example: Configuring Marking of Control and Management Traffic

The commands in the figure show an example configuration for the network.

Cisco CallManager communicates with Cisco IP Phones and gateways using TCP ports 2000 to 2002. The following sample commands classify all Skinny Protocol traffic from Cisco IP Phones and gateways (VLAN 110) and Cisco CallManager (port 4/2) as differentiated services code point (DSCP) 26:

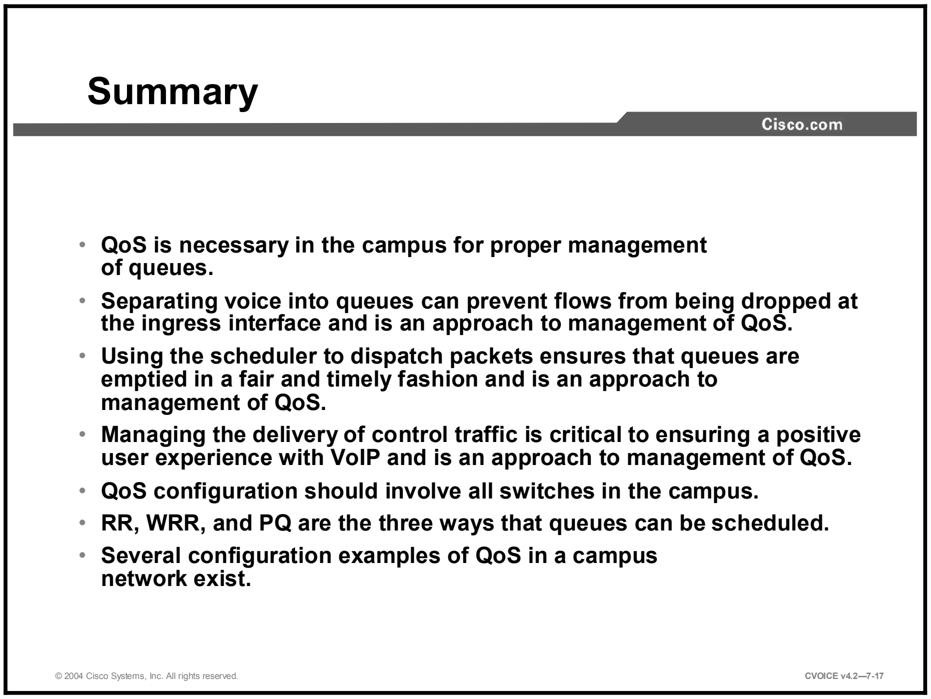
1. Enable switchwide QoS.
2. Create an ACL (**ACL_IP-PHONES**), marking all Skinny client and gateway protocol traffic from the IP Phones and from Skinny protocol gateways with a DSCP value of 26 (AF31).
3. Add to the **ACL_IP-PHONES** access list, trusting all DSCP markings from the IP Phone, so that the CoS = 5 RTP traffic is not rewritten.
4. Create an ACL (**ACL_VOIP_CONTROL**), marking all Skinny client and gateway protocol traffic from Cisco CallManager with a DSCP value of 26 (AF31).
5. Accept incoming Layer 2 CoS classification. (Current 10/100 version 1 line cards must have **trust-cos** enabled even though the parser returns an error.)
6. Configure the port so that all QoS associated with the port will be implemented on a VLAN basis.
7. Instruct the Cisco IP Phone to rewrite CoS from the PC to CoS = 0 within the IP Phone Ethernet application-specific integrated circuit (ASIC).
8. Inform Cisco CallManager port 4/2 that all QoS associated with the port will be done on a port basis.

9. Write the ACL to hardware.
10. Map the **ACL_IP-PHONES** ACL to the auxiliary VLAN.
11. Map the **ACL_VOIP_CONTROL** ACL to the Cisco CallManager port.

Note Beginning with Release 3.0(5), Cisco CallManager includes the ability to configure the CoS values for all VoIP control and management traffic from Cisco CallManager, the IP Phones, and the Skinny Protocol gateways (this does not include the AT and AS model analog gateways). This user-configurable classification means that network element access lists are no longer required for marking Skinny protocol VoIP control traffic. H.323 and Media Gateway Control Protocol (MGCP) traffic still require external network element marking.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to these resources:

- *Cisco AVVID Network Infrastructure Enterprise Quality of Service Design*
  http://www.cisco.com/application/pdf/en/us/guest/netsol/ns17/c649/ccmigration_09186a00800d67ed.pdf
- *Deploying QoS for Enterprise Networks* (DQOS) course

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which ports connect to access switches and core switches?

A) access ports

B) core ports

C) distribution ports

D) fast ports

Q2) What is the result of voice packet loss?

A) clipping and skips

B) echo

C) jitter

D) overlap

Q3) Which QoS service is best for high-speed campus networks?

A) fair queuing

B) queue scheduling

C) separation of queues

D) marking control and management traffic

Q4) What must be disabled before you enable multiple queues on Catalyst switches?

A) flow control

B) signaling

C) priority queuing

D) streaming

Q5) Which type of queue scheduling provides QoS by assigning bandwidth to higher priority applications?

A) priority queue

B) interleaving

C) round robin

D) weighted round-robin

Q6) Which queue scheduling method is best suited for branch office and wiring-closet switches?

A) interleaving

B) round robin

C) weighted round-robin

D) priority queue

Q7) What is used to mark control and management traffic on Catalyst 3500 and 4000 switches that are enabled for Layers 3 and 4?

A) transmission control protocols

B) access control lists

C) signaling protocols

D) firewalls

Q8) What network problems adversely affect signaling traffic for gateways and telephones?

A) echo

B) congestion

C) delay and loss

D) network speed

Q9) Which strategy can you use to allow a large number of telephones on a network that does not have enough IP addresses?

A) disabling routing on the network

B) configuring stub areas

C) using class C addressing only

D) isolating the telephones on an auxiliary VLAN

Q10) Which three of the following categories of QoS are suited for a campus network? (Choose three.)

A) separation of queues

B) link efficiency

C) queue scheduling

D) marking control and management traffic

E) policing

F) compression

Q11) What is the recommended configuration for the receive interface in switchwide queuing?

A) 2Q1T

B) 802.1p CoS

C) one standard FIFO queue

D) multiple receive and transmit queues with thresholds

Q12) What type of configuration can be used for queuing with Catalyst 6000 switches?

A) 2Q1T

B) 802.1p CoS

C) one standard FIFO queue

D) multiple receive and transmit queues with thresholds

Q13) Which command can be used to enable a second queue on the Catalyst 4000 series switches?

A) **set qos**

B) **set qos queue 2**

C) **set qos map**

D) **set qos 0-1**

Q14) Which command is used to enable switchwide queuing?

A) **set qos enable**

B) **enable set qos**

C) **qos set enable**

D) **qos enable set**

Quiz Answer Key

Q1) C

Relates to: Need for QoS in the Campus

Q2) A

Relates to: Need for QoS in the Campus

Q3) C

Relates to: Separation of Queues

Q4) A

Relates to: Separation of Queues

Q5) D

Relates to: Queue Scheduling

Q6) C

Relates to: Queue Scheduling

Q7) B

Relates to: Marking Control and Management Traffic

Q8) C

Relates to: Marking Control and Management Traffic

Q9) D

Relates to: Configuring QoS in the Campus

Q10) A, C, D

Relates to: Configuring QoS in the Campus

Q11) C

Relates to: Separation and Scheduling of Queues

Q12) D

Relates to: Separation and Scheduling of Queues

Q13) C

Relates to: Configuration Examples

Q14) A

Relates to: Configuration Examples

Understanding QoS Tools in the WAN

Overview

This lesson introduces the need for QoS in a WAN and describes some of the tools that can be applied.

Relevance

As VoIP networks grow to encompass the geographical areas served by WANs, it is essential that network administrators understand QoS tools available for implementation in the WAN.

Objectives

Upon completing this lesson, you will be able to describe the tools available for implementing QoS in the WAN. This includes being able to meet these objectives:

- Describe six reasons for implementing QoS in a WAN
- List five QoS mechanisms used in a WAN
- Describe how bandwidth provisioning provides QoS in a WAN
- Illustrate how optimized queuing provides QoS in a WAN
- Describe how IP precedence and DSCP provide link efficiency in a WAN
- List three link fragmentation and interleaving techniques that provide QoS in a WAN
- Identify the types of links that require traffic shaping
- Describe how CAC provides QoS in a WAN

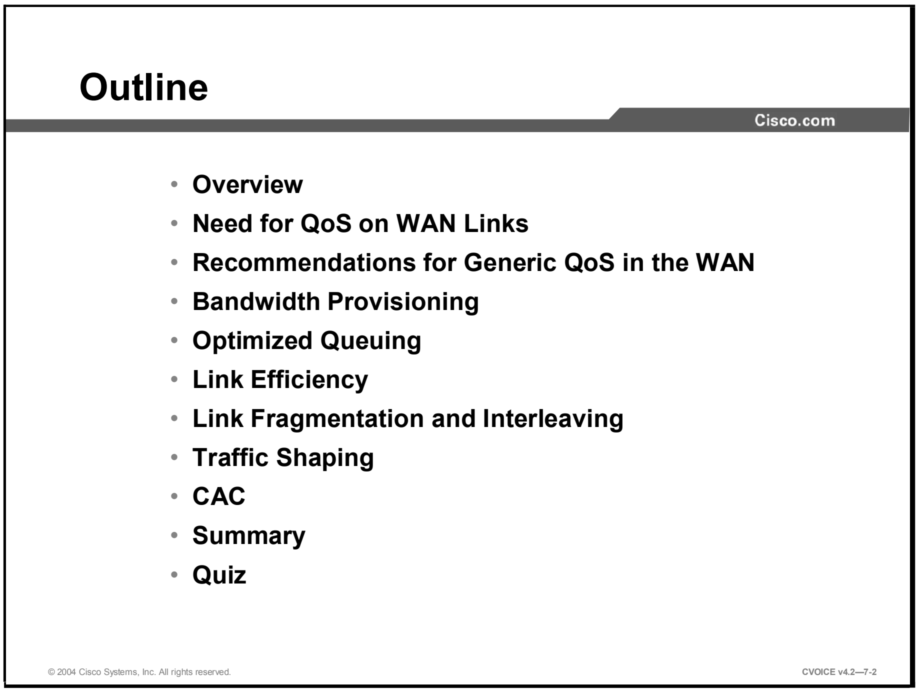
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of WAN technologies
- Knowledge of TCP/IP
- Basic knowledge of IP telephony

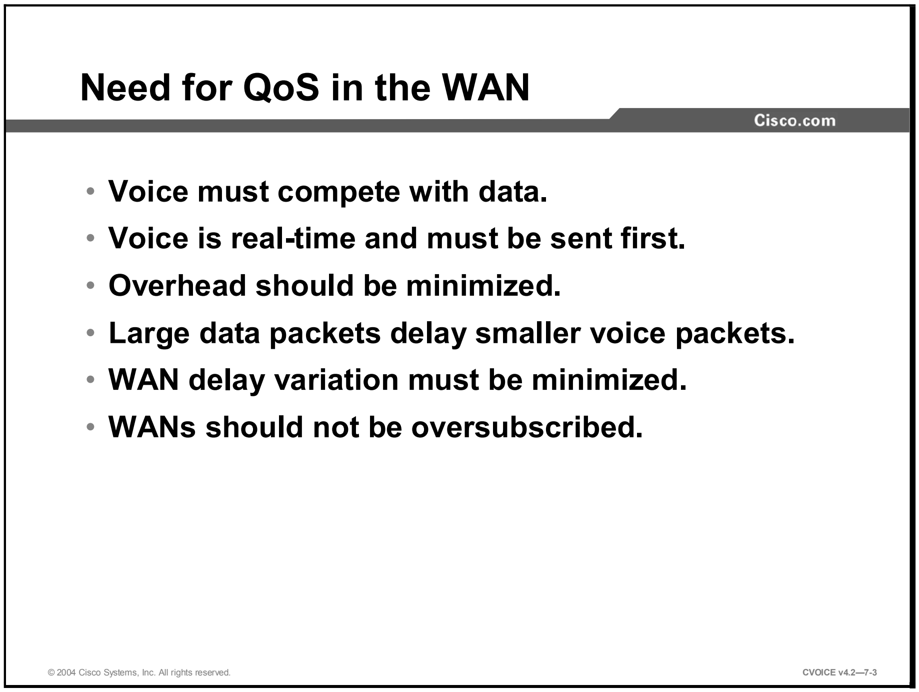
Outline

The outline lists the topics included in this lesson.



Need for QoS on WAN Links

This topic defines the need for QoS in a WAN.



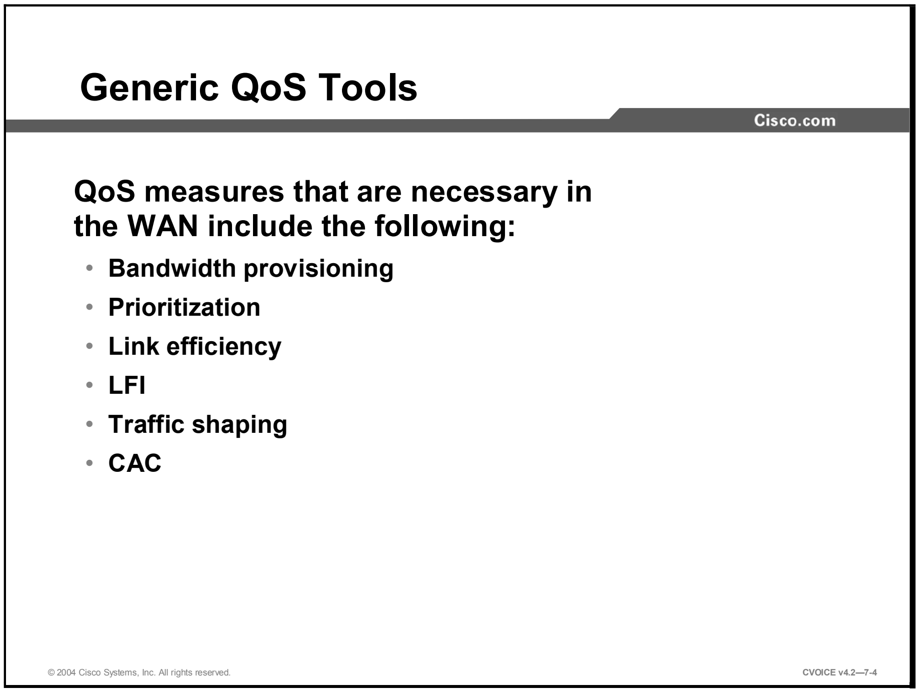
You must consider and implement certain QoS measures before voice can be placed on a WAN to ensure the proper performance of voice and data applications. The following factors explain the need for QoS:

- **Voice must compete with data:** Voice traffic demand on the WAN is typically smooth; each voice call consumes a relatively fixed amount of bandwidth for the duration of the call. On the other hand, data traffic is bursty, peaking and leveling off based on user access and application type. The amount of total bandwidth that you provision must have the capacity to carry all of the expected traffic.
- **Voice is real-time and must be sent first:** Because voice is a real-time application and delayed packets have a severe impact on performance, you must prioritize voice ahead of data traffic that is not real-time traffic.
- **Overhead should be minimized:** Voice is sent using UDP. Unfortunately, this requirement adds significant overhead to the overall packet size. You must identify a means for numbering the packets that are needed, because minimizing the overhead with compression allows for smaller, more efficient packets.
- **Large data packets delay smaller voice packets:** With certain applications such as file transfers and web browsing, data packets that compete with voice can be relatively large. Voice packets are forced to wait until the larger data packets are sent. By fragmenting the data packets into smaller, more manageable sizes, you can reduce the delay that is experienced by voice packets.
- **WAN delay variation must be minimized:** The natural behavior of certain WAN technologies, such as Frame Relay, ATM, and PPP can exacerbate delay variation and result in poor voice performance. By using tools to reduce the delay variation on WAN technologies, you can add significant performance to voice applications.

- **WANs should not be oversubscribed:** If you place too many voice calls on a WAN, the necessary bandwidth that is required by data applications will be choked. You must take care to ensure that data still gets its deserved bandwidth. Excess voice calls can severely impact all calls on the network.

Recommendations for Generic QoS in the WAN

This topic lists generic QoS tools.

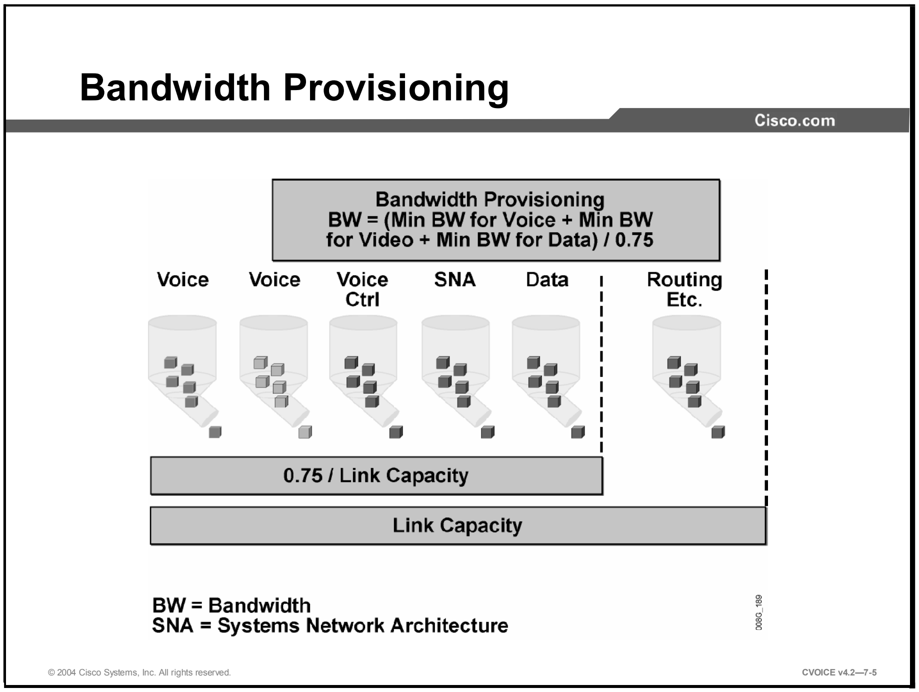


Depending on the Layer 2 technology that you use, the following QoS measures may be necessary in the WAN. Each of these QoS tools will be covered in this lesson:

- Bandwidth provisioning
- Prioritization
- Link efficiency
- LFI
- Traffic shaping
- CAC

Bandwidth Provisioning

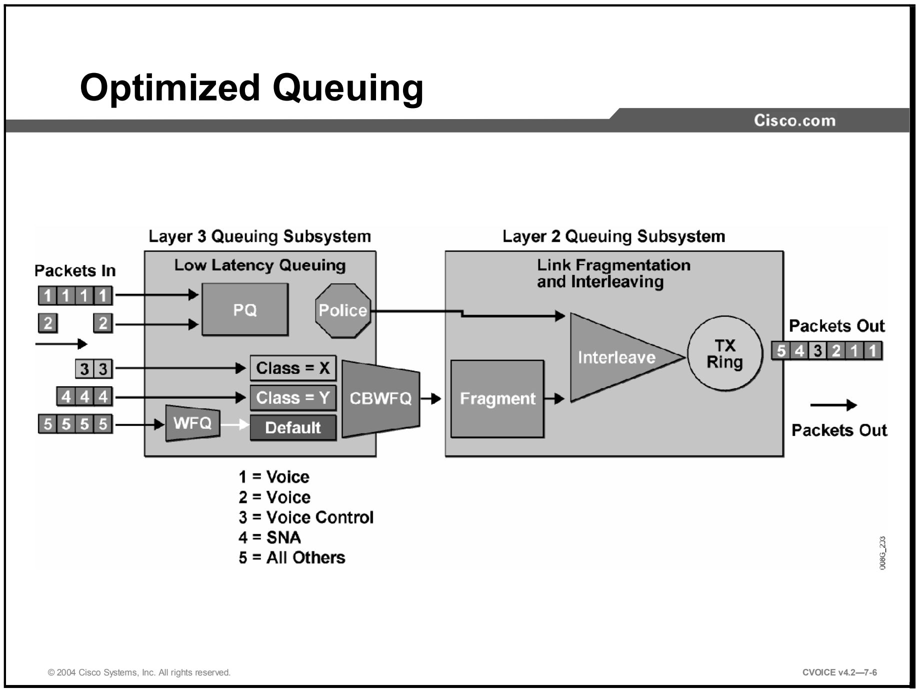
This topic describes bandwidth provisioning as a tool for implementing QoS in the WAN.



The figure represents a typical network. This network has a number of delay-sensitive traffic types: voice, video, and Systems Network Architecture (SNA). Before you place voice and video on a network, you must ensure that adequate bandwidth exists for all required applications; for example, SNA is also delay-sensitive but should not be queued *ahead* of voice and video. To begin, the minimum bandwidth requirements for each major application (for example, voice media streams, video streams, voice control protocols, and all data traffic) should be summed. This sum represents the minimum bandwidth requirement for any given link, and it should consume no more than 75 percent of the total bandwidth that is available on that link. This 75 percent rule assumes that some bandwidth is required for overhead traffic, such as routing and Layer 2 keepalives, as well as for additional applications such as e-mail, HTTP traffic, and other data traffic that is not easily measured.

Optimized Queuing

This topic describes optimized queuing as a tool for implementing QoS in the WAN.



When you are choosing from among the many available prioritization schemes, the major factors that you must consider include the type of traffic on the network and the wide-area media that is being traversed.

For multiservice traffic over an IP WAN, Cisco recommends LLQ for low-speed links. This approach allows up to 64 traffic classes with the ability to specify, for example, PQ behavior for voice and interactive video, a minimum bandwidth for SNA data and market data feeds, and, as illustrated in the figure, WFQ to other traffic types.

Example: Optimized Queuing

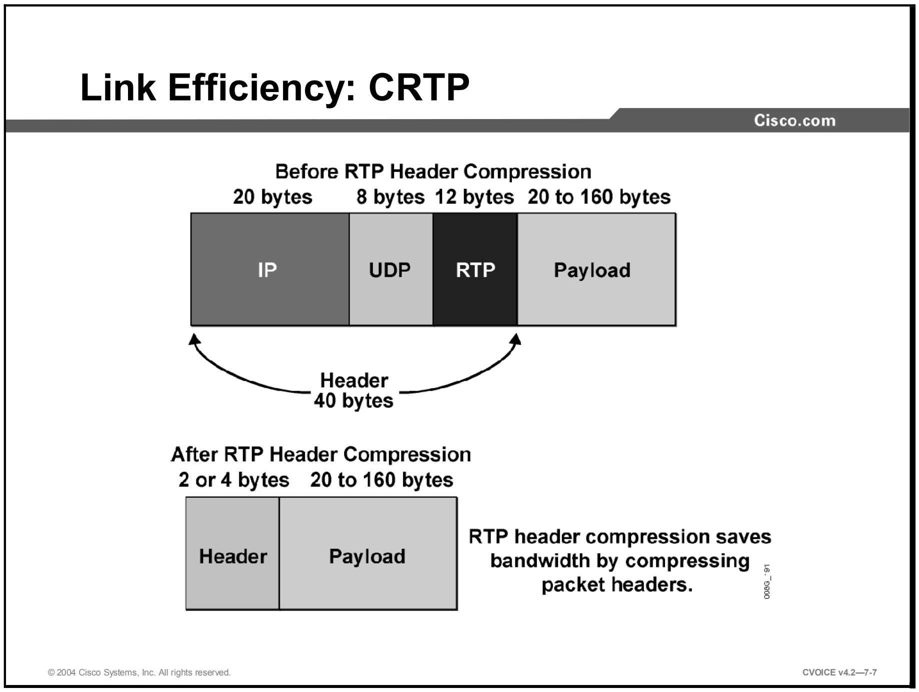
The figure illustrates LLQ, which works as follows:

- Voice is placed into a queue with PQ capabilities and allocated a bandwidth of 48 kbps, for example. This bandwidth is based on the total number of expected calls. The entrance criterion to this queue should be the DSCP value of Express Forwarding (EF), or IP precedence value of 5. Traffic in excess of 48 kbps is dropped if the interface becomes congested. Therefore, you must use an admission control mechanism to ensure that you do not exceed this value.
- As the WAN links become congested, it is possible to completely starve the voice control signaling protocols and therefore eliminate the capacity of the IP Phones to complete calls across the IP WAN. Voice control protocol traffic, such as H.323 and SCCP, requires its own CBWFQ with a minimum configurable bandwidth equal to a DSCP value of AF31, which correlates to an IP precedence value of 3.

- SNA traffic is placed into a queue that has a specified bandwidth of 56 kbps, for example. This bandwidth represents the expected SNA demand on the network. Queuing operation within this class is FIFO with a minimum allocated bandwidth of 56 kbps. Traffic in this class that exceeds 56 kbps is placed in the default queue. The entrance criterion to this queue could be TCP port numbers, a Layer 3 address, IP precedence, or a DSCP.
- You can place all remaining traffic in a default queue. If a bandwidth is specified, the queuing operation is FIFO. Alternatively, specifying the keyword **fair** assigns WFQ to the operation.

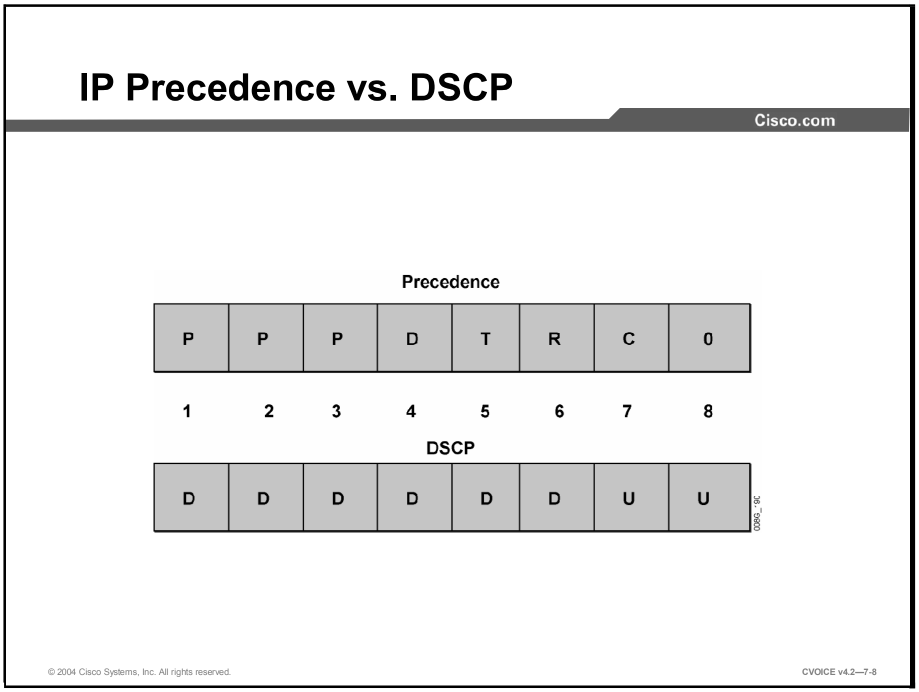
Link Efficiency

This topic describes link efficiency as a tool for implementing QoS in the WAN.



Because wide-area bandwidth is often prohibitively expensive, only low-speed circuits may be available or cost-effective when you are interconnecting remote sites. In this scenario, it is important to achieve the maximum savings by transmitting as many voice calls as possible over the low-speed link. Many compression schemes, such as G.729, can squeeze a 64-kbps call down to an 8-kbps payload. Cisco gateways and IP Phones support a range of codecs that enhance efficiency on these low-speed links.

The link efficiency is further increased by using CRTP, which compresses a 40-byte IP + UDP + RTP header to approximately 2 to 4 bytes. In addition, VAD takes advantage of the fact that in most conversations, only one party is talking at a time. VAD recovers this empty time and allows data to use the bandwidth.



IP Precedence

The 8-bit CoS field in an IP packet header was originally defined in RFC 791 (superseded by RFC 1122). It defined the most significant bits as shown in the table.

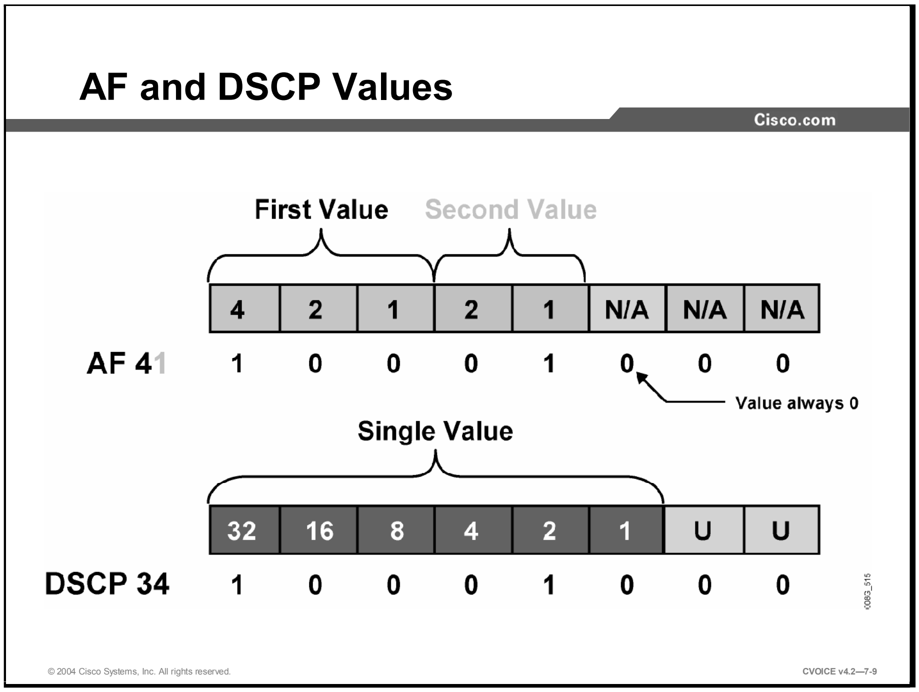
IP Precedence

Binary Value	Precedence Value	Description
111	7	Network Control
110	6	Internetwork Control
101	5	Critical
100	4	Flash Override
011	3	Flash
010	2	Immediate
001	1	Priority
000	0	Routine

In addition, the next 4 bits, when turned on, refer to the following:

- **Bit 4, D:** Instructs the network to minimize delay
- **Bit 5, T:** Instructs the network to maximize throughput
- **Bit 6, R:** Instructs the network to maximize reliability
- **Bit 7, C:** Instructs the network to minimize costs
- **Bit 8:** Reserved for future use

DSCP



The newest use of the eight CoS bits is commonly called the DiffServ standard. It uses the same precedence bits (the most significant bits: 1, 2, and 3) for priority setting, but further clarifies their functions and definitions and offers finer priority granularity through use of the next three bits in the CoS field. DiffServ reorganizes and renames the precedence levels (still defined by the three most significant bits of the CoS field) into the categories shown in the table.

DiffServ Standard

Precedence Bits	Description
Precedence 7	Stays the same (link layer and routing protocol keepalive)
Precedence 6	Stays the same (used for IP routing protocols)
Precedence 5	EF
Precedence 4	Class 4
Precedence 3	Class 3
Precedence 2	Class 2
Precedence 1	Class 1
Precedence 0	Best effort

Bits 3 and 4 of the CoS field (now called the DSCP in the DiffServ standard) allow further priority granularity through the specification of packet-drop probability for any of the defined classes. Collectively, classes 1 through 4 are referred to as Assured Forwarding (AF). The table illustrates the DSCP coding for specifying the priority level (class) plus the drop percentage. (Bits 1, 2, and 3 define the class; bits 4 and 5 specify the drop percentage; bit 6 is always 0.) As an example, AF41 is expressed as 100010, where the first three bits represent class 4, the next two bits specify a low drop percentage, and the last bit is always 0. AF41 has a higher priority than any class 3, 2, or 1, and enjoys the lowest drop percentage. AF41 is also referred to as DSCP 34, as shown in the figure.

DSCP Coding

Drop Percentage	DSCP Coding			
	Class 1	Class 2	Class 3	Class 4
Low drop percentage	001010	010010	011010	100010
Medium drop percentage	001100	010100	011100	100100
High drop percentage	001110	010110	011110	100110

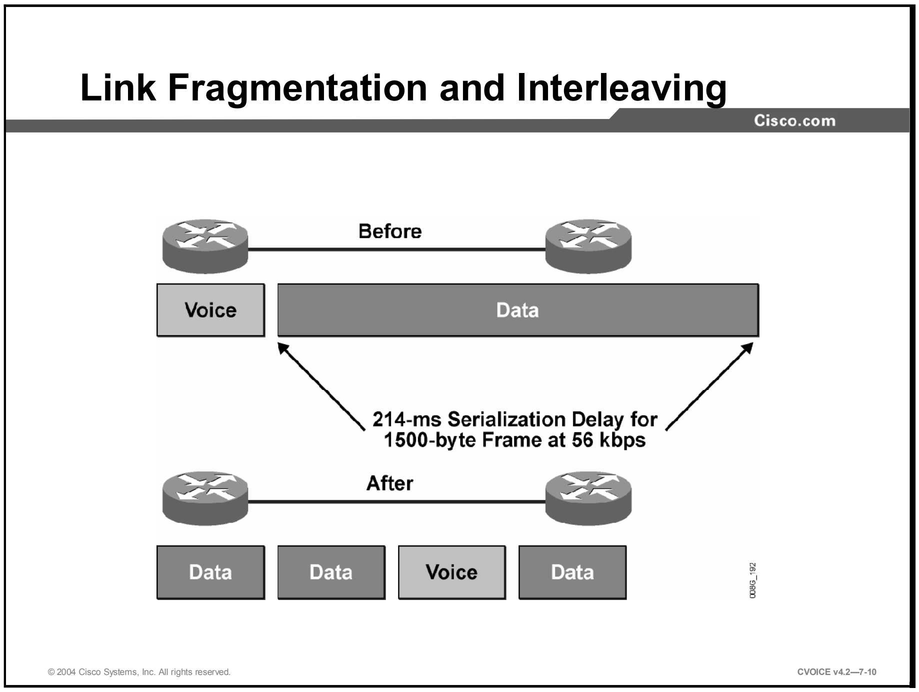
Using this system, a device first prioritizes traffic by class, then differentiates and prioritizes traffic that is in the same class by considering the drop percentage. It is important to note that this standard does not offer a precise definition of low, medium, and high drop percentages. Additionally, not all devices recognize the DiffServ bit 4 and 5 settings. In fact, even when the settings are recognized, they do not necessarily trigger the same forwarding action from each device on the network.

Each device implements its own response in relation to the packet priorities that it detects. The DiffServ proposal is meant to allow a finer granularity of priority setting for the applications and devices that can use it, but it does not specify interpretation (that is, action to be taken). In this application, bits 7 and 8 are unused.

The figure shows the relationship between AF values and DSCP values. AF41 can also be referred to as DSCP 34.

Link Fragmentation and Interleaving

This topic describes link fragmentation and interleaving (LFI) as a tool for implementing QoS in the WAN.



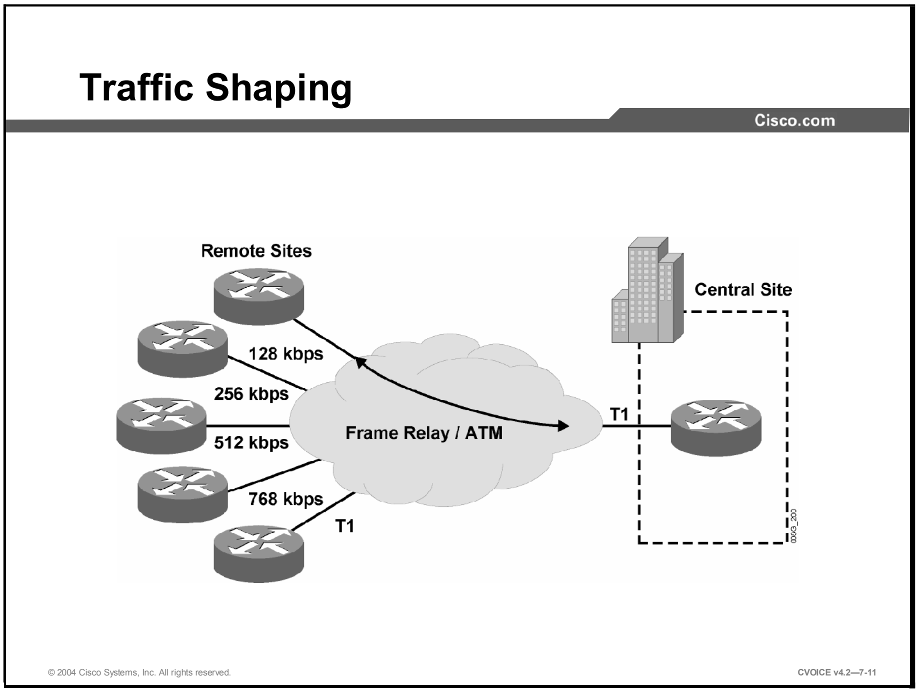
For low-speed links (less than 768 kbps), it is necessary to use techniques that provide LFI. This approach places bounds on jitter by preventing the delay of voice traffic behind large data frames. The following three techniques exist for this purpose:

- MLP for point-to-point serial links
- FRF.12 for Frame Relay
- MLP over ATM for ATM connections

The figure shows the necessity of fragmenting the large data packet into smaller segments. Data packets should be fragmented to the same size as voice packets. If the data packets are fragmented to a size smaller than voice packets, voice packets will also be fragmented. Determining the size of the voice packet will depend on the codec type selected. Also, remember that using CRTP can reduce the size of a voice packet significantly.

Traffic Shaping

This topic describes traffic shaping as a tool for implementing QoS in the WAN.



Traffic shaping uses a rate control mechanism called a token bucket filter. This token bucket filter is set as follows:

excess burst plus committed burst (Be + Bc) = maximum speed for the virtual circuit (VC)

Traffic above the maximum speed is buffered in a traffic-shaping queue, which is equal to the size of the WFQ. The token bucket filter does not filter traffic, but controls the rate at which traffic is sent on the outbound interface.

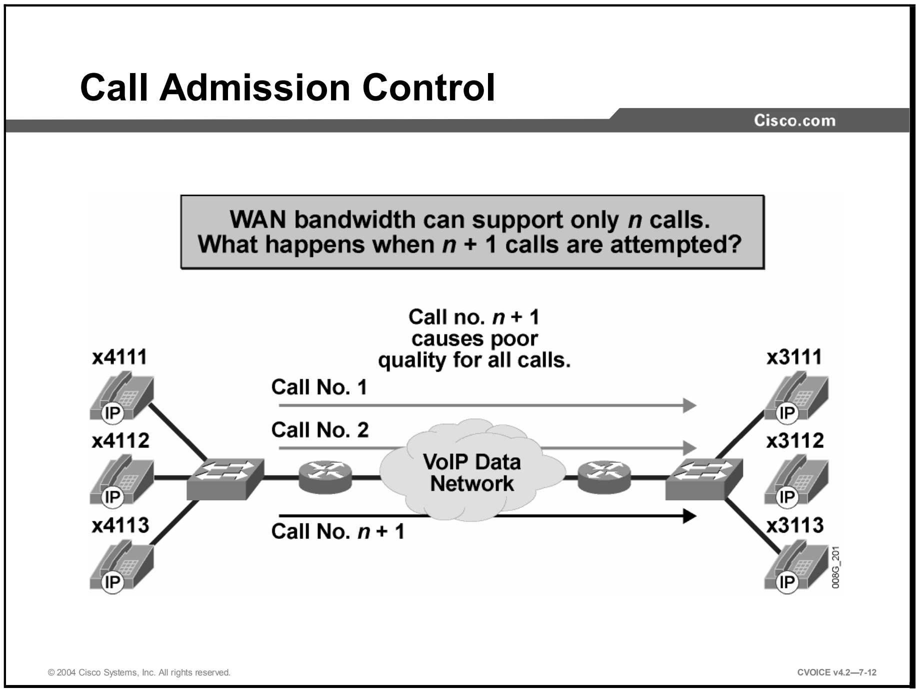
Traffic shaping is required for multiple-access nonbroadcast media, such as ATM and Frame Relay, in which the physical access speed varies between two endpoints. Traffic-shaping technology accommodates mismatched access speeds. In the case of Frame Relay with FRF.12, traffic shaping also allows delay variation, or jitter, to be bounded appropriately.

Example: Traffic Shaping

The figure shows an example of why traffic shaping is necessary. First, the remote sites should not oversubscribe the central site, as is shown. Second, traffic should be shaped to the lowest speed remote site, in this case, 128 kbps.

CAC

This topic describes CAC as a tool for implementing QoS in the WAN.



CAC is required to ensure that network resources are not oversubscribed. CAC could be described as a way to protect voice from voice. Calls that exceed the specified bandwidth are either rerouted using an alternative route such as the PSTN, or a fast busy tone is returned to the calling party. This way the next voice call does not degrade the quality of all the calls on the link. You can implement CAC on a gatekeeper or by using Cisco CallManager.

Reference For more information on VoIP CAC, see the following topic on the Cisco.com website: http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/cac.htm

Example: CAC

The figure illustrates the need for CAC. If the network segment was designed to carry two VoIP calls, then when the third call arrives at either the gatekeeper or the CallManager, the call is either rerouted to the PSTN or the caller is sent a fast busy tone. This way, the third call does not impact the quality of the two existing calls.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- Factors that necessitate QoS in WANs include voice competing with data, voice as a real-time application, minimizing delay, delay variation, and overhead.
- WAN QoS tools include bandwidth provisioning, prioritization, link efficiency, LFI, traffic shaping, and CAC.
- Bandwidth provisioning ensures adequate bandwidth for all applications.
- LLQ on low-speed links allows up to 64 classes.
- IP precedence and DSCP are used to provide link efficiency.

 CVOICE v4.2—7-13

Summary (Cont.)

Cisco.com

- MLP, FRF.12, and MLP over ATM provide LFI over low-speed links.
- Traffic shaping is required for multiple-access nonbroadcast media such as ATM and Frame Relay.
- CAC ensures that network resources are not oversubscribed.

 CVOICE v4.2—7-14

References

For additional information, refer to these resources:

- *Voice over IP (VoIP) Frame Relay (VoFR) and ATM (VoATM)*
  http://www.cisco.com/pcgi-bin/Support/browse/index.pl?i=Technologies&f=775

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Which characteristic describes voice traffic?

A) bursty

B) real-time

C) delay tolerant

D) bandwidth hungry

Q2) Which requirement of voice adds significant overhead to the voice packets?

A) prioritization of voice packets

B) fragmenting the packets

C) numbering the packets

D) reducing the delay variation

Q3) Which two QoS mechanisms are commonly used on the WAN? (Choose two.)

A) bandwidth provisioning

B) queue scheduling

C) traffic shaping

D) separation of queues

E) marking of control and management traffic

Q4) Which OSI layer technology affects the type of QoS measures used in the WAN?

A) Layer 1

B) Layer 2

C) Layer 3

D) Layer 4

Q5) How much of the total available bandwidth should be consumed by minimum bandwidth requirements?

A) 100 percent of the bandwidth

B) no more than 99 percent of the bandwidth

C) no more than 80 percent of the bandwidth

D) no more than 75 percent of the bandwidth

Q6) Which two types of traffic are NOT included in the minimum bandwidth requirement calculation? (Choose two.)

A) voice media streams

B) video streams

C) HTTP traffic

D) H.323 traffic

E) overhead traffic

Q7) In optimized queuing, what kind of queuing do you use for voice control signaling protocols?

A) DSCP

B) FIFO

C) WFQ

D) CBWFQ

Q8) How many traffic classes does Low Latency Queuing allow for?

A) 32

B) 48

C) 56

D) 64

Q9) Describe two ways that DiffServ improves link efficiency in a WAN. (Choose two.)

A) by specifying how the device should forward the packet

B) by providing a finer granularity of priority setting for the applications

C) by providing a finer granularity of priority setting for the devices

D) by specifying the range of acceptable link speeds

Q10) What is the size of the CoS field that is used for IP precedence in an IP packet header?

A) 3 bits

B) 4 bits

C) 7 bits

D) 8 bits

Q11) When should you use QoS techniques that provide LFI?

A) when the speed of the link is less than 56 kbps

B) when the speed of the link is less than 768 kbps

C) when the speed of the link is greater than 786 kbps

D) when the speed of the link is less than 1544 kbps

Q12) Which technique is used for providing LFI on Frame Relay links?

- A) MLP
- B) MLP over ATM
- C) FRF.12
- D) G.711
- E) G.729

Q13) Which type of link is traffic shaping NOT suitable for?

- A) ATM links
- B) Frame Relay links
- C) Ethernet
- D) links with variable access speeds

Q14) Which three types of links can traffic-shaping mechanisms be used to limit jitter on? (Choose three.)

- A) Frame Relay links
- B) ATM links
- C) point-to-point links
- D) on all links that support CAC
- E) only on links that have link speeds of 768 KB or less

Q15) Which two statements describe CAC as a QoS technique? (Choose two.)

- A) specifies bandwidth for a link
- B) frees up bandwidth for higher-priority traffic
- C) reroutes calls that exceed the specified bandwidth
- D) protects voice from voice
- E) can be implemented by either a CallManager or a gateway

Q16) Which two tools will allow you to implement CAC on an entire network? (Choose two.)

- A) Cisco IOS
- B) Cisco CallManager
- C) Cisco Voice Health Monitor
- D) IVR
- E) Cisco Performance Monitor
- F) Cisco Quality Policy Manager

Quiz Answer Key

Q1) B

Relates to: Need for QoS on WAN Links

Q2) B

Relates to: Need for QoS on WAN Links

Q3) A, C

Relates to: Recommendations for Generic QoS in the WAN

Q4) B

Relates to: Recommendations for Generic QoS in the WAN

Q5) D

Relates to: Bandwidth Provisioning

Q6) C, E

Relates to: Bandwidth Provisioning

Q7) D

Relates to: Optimized Queuing

Q8) D

Relates to: Optimized Queuing

Q9) B, C

Relates to: Link Efficiency

Q10) D

Relates to: Link Efficiency

Q11) B

Relates to: Link Fragmentation and Interleaving

Q12) C

Relates to: Link Fragmentation and Interleaving

Q13) C

Relates to: Traffic Shaping

Q14) A, B, C

Relates to: Traffic Shaping

Q15) C, D

Relates to: CAC

Q16) A, B

Relates to: CAC

Configuring QoS in the WAN

Overview

This lesson illustrates configuration examples for QoS tools that are used in the WAN in both Frame Relay and PPP and introduces the Quality Policy Manager for CiscoWorks.

Relevance

As VoIP networks grow to encompass the geographical areas served by WANs, it is essential that network administrators are able to apply QoS tools to the WAN configuration and manage the tools properly.

Objectives

Upon completing this lesson, you will be able to configure QoS parameters in VoIPovFR and VoIP over PPP applications. This includes being able to meet these objectives:

- List five major components of QoS in a VoIPovFR network and describe three of those components
- List four major components of QoS in a VoIP over PPP network and describe one of those components
- Configure all QoS components on a VoIPovFR network using map classes and policies on the router
- Configure all QoS components on a VoIP over PPP network using map classes and policies on the router
- Configure AutoQoS to support VoIP
- Describe the basics of QoS Policy Manager as a management tool for QoS

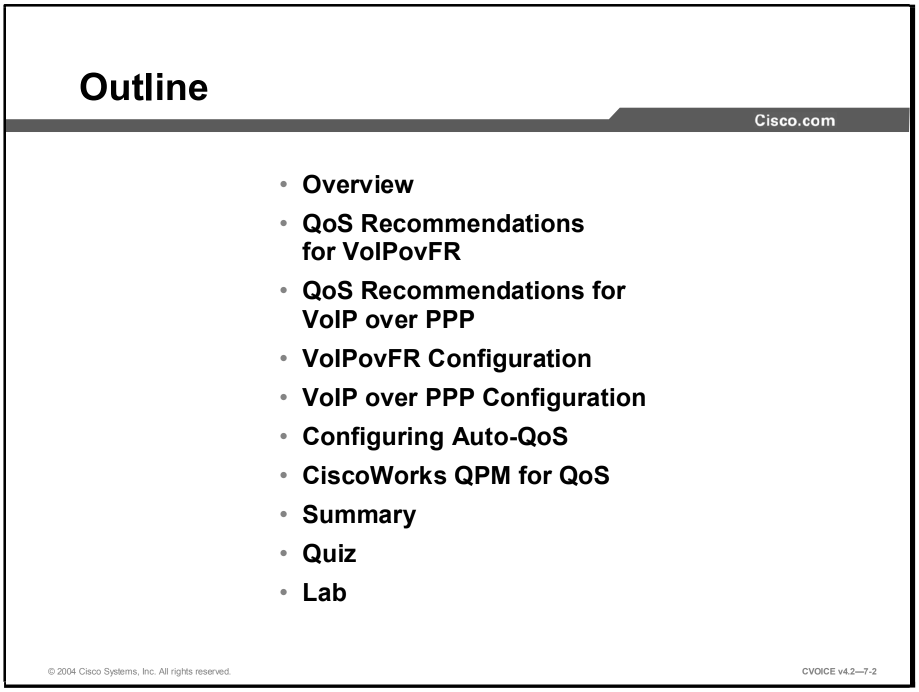
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of WAN technologies
- Knowledge of TCP/IP
- Basic knowledge of IP telephony

Outline

The outline lists the topics included in this lesson.



QoS Recommendations for VoIPovFR

This topic identifies some of the QoS tools implemented in a Frame Relay network to support VoIP.

VoIP over Frame Relay QoS

Cisco.com

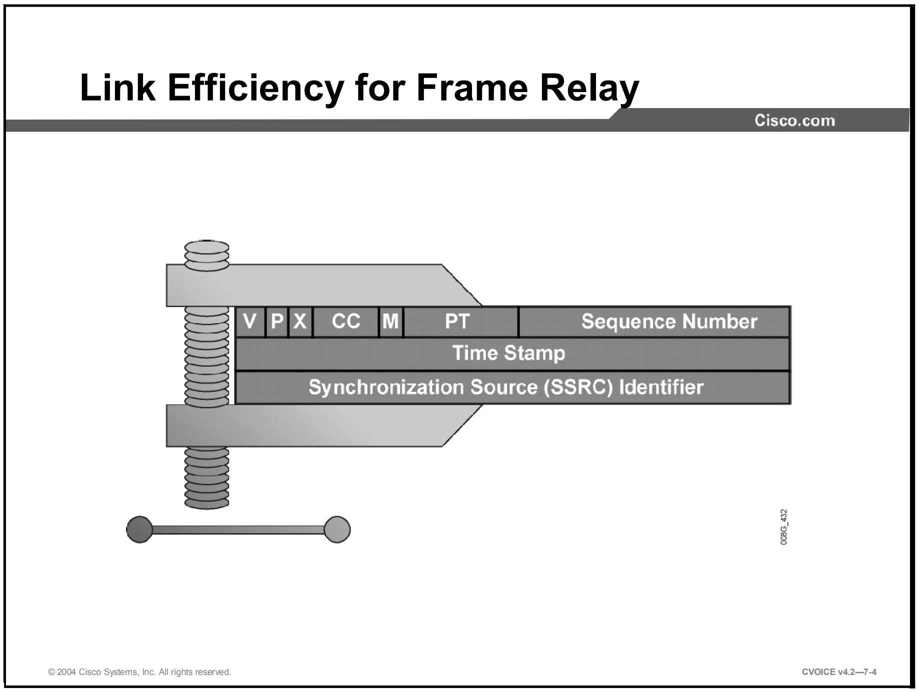
QoS components for VoIPovFR include:

- **Link efficiency with CRTP**
- **Fragmentation**
- **Traffic shaping**
- **LLQ**
- **CAC**

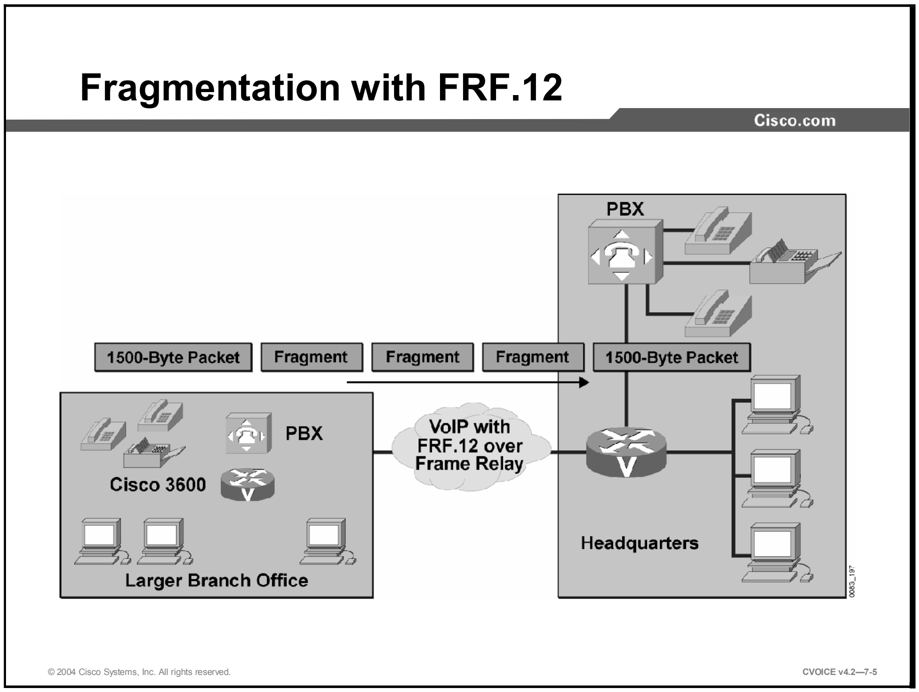
CVOICE v4.2—7-3

The five major components of QoS that are particular to VoIPovFR are as follows:

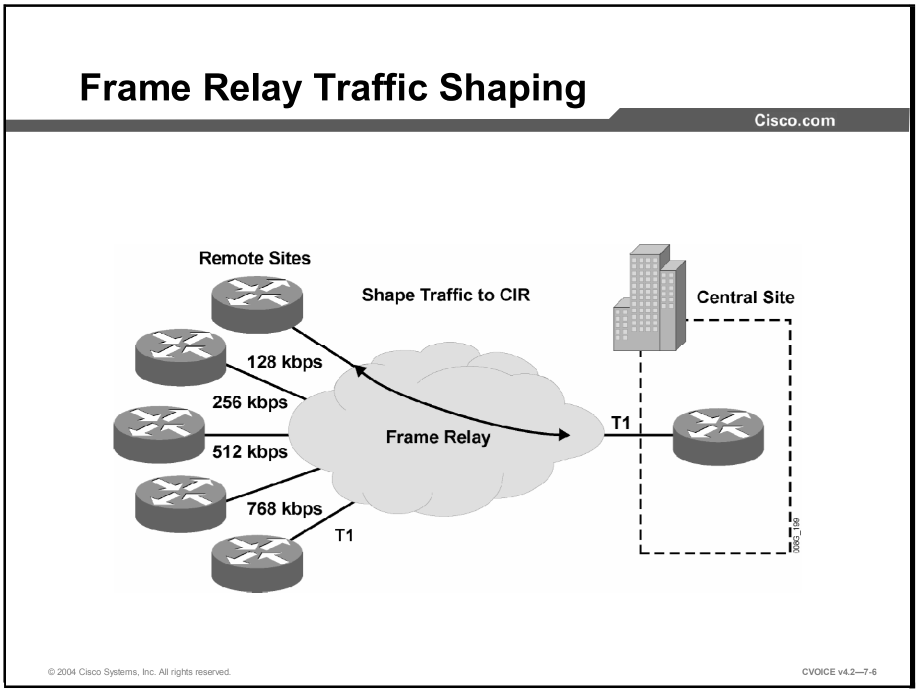
- Link efficiency with CRTP
- Fragmentation
- Traffic shaping
- LLQ
- CAC



CRTP is explicitly enabled on a Frame Relay interface. Compressing the RTP header can reduce it from 40 bytes to 2 or 4 bytes, depending on whether the cyclic redundancy check (CRC) is included.



When running VoIP over Frame Relay, you must facilitate fragmentation of larger data frames to allow interleaving of smaller voice frames. This fragmentation is accomplished with FRF.12. FRF.12 adds an extended header to the Frame Relay frame that identifies the frame as fragmented and indicates a sequence number for reassembly. It provides end-to-end fragmentation on a per-VC basis. As stated earlier, fragmentation is necessary only on links of 768 kbps or less.



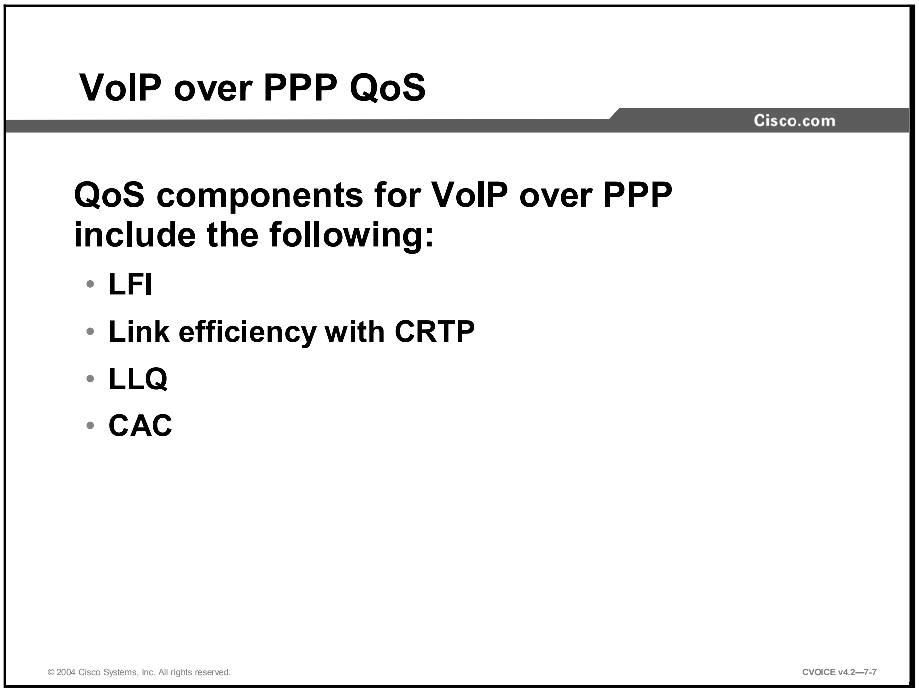
You must enable Frame Relay traffic shaping (FRTS) to shape the VC to the CIR value. Shaping to the CIR for each VC that carries voice ensures that the service provider does not mark any voice frames discard eligible (DE). FRTS provides useful parameters for managing network traffic congestion on Frame Relay networks. FRTS eliminates bottlenecks in Frame Relay networks with high-speed connections to the central site and low-speed connections to the branch sites. FRTS ensures that traffic above the CIR on the link is not dropped but is buffered and sent when there is capacity on the link.

Example: Frame Relay Traffic Shaping

You can configure rate enforcement values to limit the rate at which data is sent from the VC at the central site. Without traffic shaping, certain VCs may overuse the outbound interface, thereby starving other VCs and causing delay and jitter.

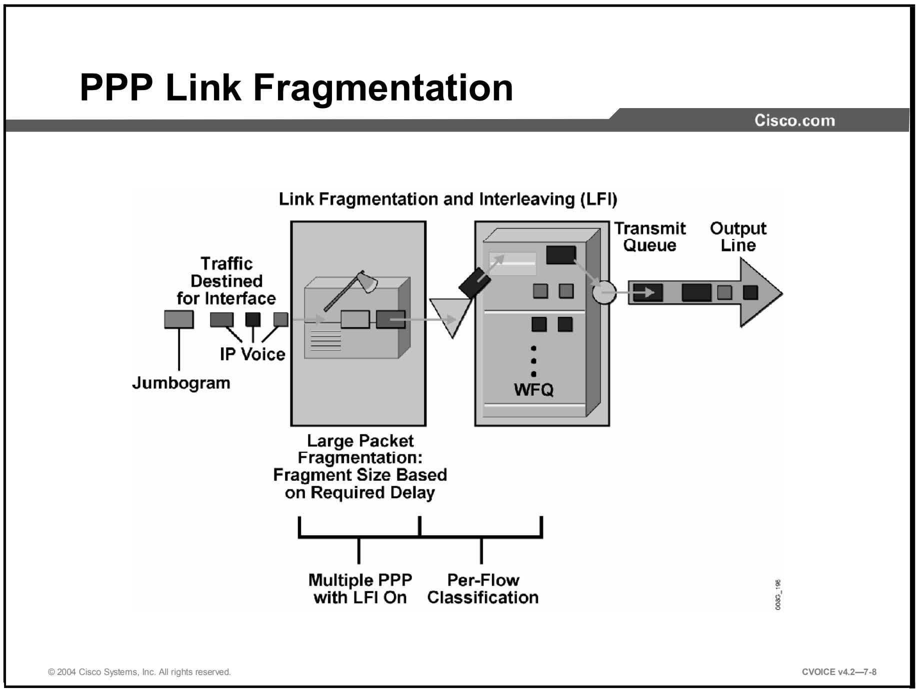
QoS Recommendations for VoIP over PPP

This topic identifies some of the QoS tools implemented in a PPP network to support VoIP.



Following are the four major components of QoS that are particular to VoIP over PPP:

- LFI
- Link efficiency with CRTP
- LLQ
- CAC



While 1500 bytes is a common size for data packets, a typical VoIP packet (carrying G.729 voice frames) is approximately 66 bytes (20 bytes voice payload, 6 bytes Layer 2 header, 20 bytes RTP and UDP header, and 20 bytes IP header).

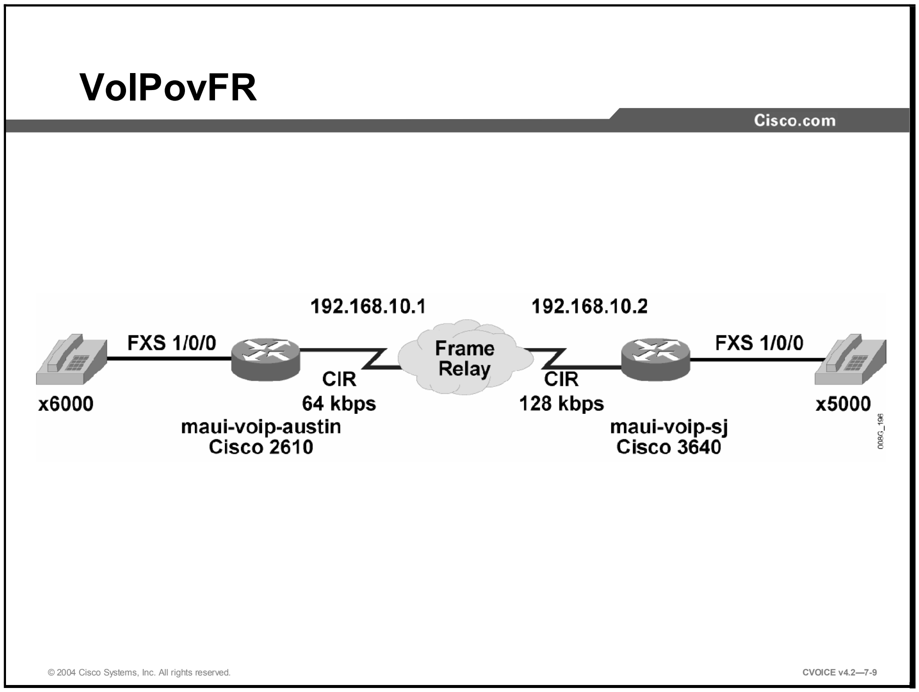
Now, imagine a 56-kbps leased-line link in which voice and data traffic coexist. If a voice packet is ready to be serialized just when a data packet starts transmission over the link, a problem occurs. The delay-sensitive voice packet has to wait 214 ms before transmission (it takes 214 ms to serialize a 1500-byte packet over a 56-kbps link).

Example: PPP Link Fragmentation

Large data packets can significantly delay delivery of small voice packets and therefore reduce speech quality. Fragmenting these large data packets into smaller packets and interleaving voice packets among the fragments reduces delay and jitter. The Cisco IOS LFI feature helps satisfy the real-time delivery requirements of VoIP. The figure illustrates the operation of LFI.

VoIPovFR Configuration

This topic includes configuration examples of QoS for a VoIPovFR network.



For this example, use the network shown in the figure. Assume that a customer wants to set up all components of QoS for a VoIPovFR network.

VoIPovFR QoS

Cisco.com

Access List Configuration

```
access-list 102 permit udp any any range 16384 32767
access-list 103 permit tcp any eq 1720 any
access-list 103 permit tcp any any eq 1720
```

Notes: **access-list 102** matches VoIP traffic based on the UDP port range. Both odd and even ports are put into the PQ. **access-list 103** matches VoIP signaling protocol. In this case, H.323v2 is used with the Fast Start feature.

CVOICE v4.2—7-10

VoIPovFR QoS (Cont.)

Cisco.com

Class Maps and Policy Maps Configuration

```
class-map match-all voice-signaling
 match access-group 103
class-map match-all voice-traffic
 match access-group 102
```

Notes: Definition of the voice-signaling and traffic class maps. **voice-traffic** class uses **access-list 102** for its matching criteria. **voice-signaling** class uses **access-list 103** for its matching criteria.

```
policy-map VOICE-POLICY
 class voice-traffic
  priority 45
 class voice-signaling
  bandwidth 8
class default-class
  fair-queue
```

Notes: The policy map defines how the link resources are assigned to the different map classes. In this configuration, strict PQ is assigned to the **voice-traffic** class with a maximum bandwidth of 45 kbps. This also assigns a queue for **voice-signaling** traffic that ensures 8 kbps. Note that this is optional and has nothing to do with good voice quality. Instead, it is a way to secure signaling.

CVOICE v4.2—7-11

These figures illustrate the commands used for implementing Frame Relay QoS using map classes and policies on the router.

VoIPovFR QoS (Cont.)

Cisco.com

Map Class Configuration

```
map-class frame-relay VOIPovFR
 no frame-relay adaptive-shaping becn
```

Notes: Disables Frame Relay BECNS.

```
frame-relay cir 64000
frame-relay bc 640
```

Notes: Tc = BC/CIR. In this case Tc is forced to its minimal configurable value of 10 ms.

```
frame-relay be 0
frame-relay mincir 64000
```

Notes: Although adaptive shaping is disabled, make CIR equal minCIR as a double safety. By default minCIR would be half of CIR.

```
service-policy output VOICE-POLICY
```

Notes: Enables LLQ on the PVC (see policy map configuration from previous page).

```
frame-relay fragment 80
```

Notes: Turns on FRF.12 fragmentation and sets fragment size equal to 80 bytes. This value is based on port speed of the slowest path link. This command also enables dual-FIFO.

 CVOICE v4.2—7-12

VoIPovFR QoS (Cont.)

Cisco.com

Interface Configuration

```
interface Serial0/0
 bandwidth 128
 no ip address
 encapsulation frame-relay
 no fair-queue
 frame-relay traffic-shaping
 frame-relay ip rtp header-compression
```

Notes: Turns on traffic shaping and CRTP. If **traffic-shaping** is not enabled, then **map-class** does not start and FRF.12 and LLQ will not work.

```
interface Serial0/0.1 point-to-point
 bandwidth 128
 ip address 192.168.10.2 255.255.255.252
 frame-relay interface-dlci 300 class VOIPovFR
```

Notes: This command links the subinterface to a VoIPovFR map class. See the **map-class frame-relay VoIPovFR** command.

 CVOICE v4.2—7-13

These figures continue the QoS commands.

VoIPovFR QoS (Cont.)

Cisco.com

Dial-Peer Configuration—Precedence

```
dial-peer voice 1 pots
 destination-pattern 6000
 port 1/0/0
!
dial-peer voice 2 voip
 destination-pattern 5000
 session target ipv4:192.168.10.2
 ip precedence 5
```

Notes: Sets the precedence value to critical for systems that do not understand DSCP.

Or:

 CVOICE v4.2—7-14

VoIPovFR QoS (Cont.)

Cisco.com

Dial-Peer Configuration—DSCP

```
dial-peer voice 1 pots
 destination-pattern 6000
 port 1/0/0
!
dial-peer voice 2 voip
 destination-pattern 5000
 session target ipv4:192.168.10.2
 ip qos dscp ef media
```

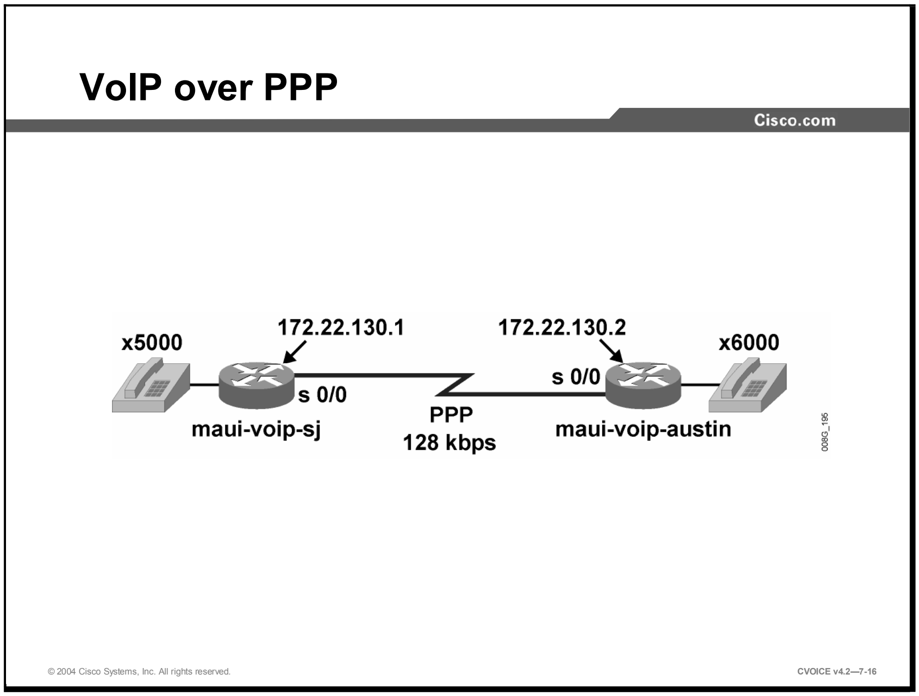
Notes: Sets the DSCP value to Express Forward for systems that understand DSCP. The **media** term applies DSCP to media payload packets.

 CVOICE v4.2—7-15

These figures continue the QoS commands.

VoIP over PPP Configuration

This topic includes configuration examples for VoIP over PPP.



For the next example, use the "VoIP over PPP" network displayed here. Assume that a customer wants to set up all components of QoS for a VoIP over PPP network.

VoIP over PPP QoS

Cisco.com

Access List Configuration

```
access-list 102 permit udp any any range 16384 37276
access-list 103 permit tcp any eq 1720 any
access-list 103 permit tcp any any eq 1720
```

Notes: **access-list 102** matches VoIP traffic based on the UDP port range Both odd and even ports are put into the PQ. **access-list 103** is used to match VoIP signaling protocol. In this case we are using H.323v2 with Fast Start feature.

CVOICE v4.2—7-17

VoIP over PPP QoS (Cont.)

Cisco.com

Class Maps and Policy Maps Configuration

```
class-map match-all voice-signaling
 match access-group 103
class-map match-all voice-traffic
 match access-group 102
```

Notes: Definition of the voice-signaling and traffic class maps. **voice-traffic** class uses access-list 102 for its matching criteria. **voice-signaling** class uses access-list 103 for its matching criteria.

```
policy-map VOICE-POLICY
 class voice-traffic
  priority 45
 class voice-signaling
  bandwidth 8
class default-class
  fair-queue
```

Notes: The policy map defines how the link resources are assigned to the different map classes. In this configuration, strict PQ is assigned to the voice-traffic class with a maximum bandwidth of 45 kbps. This also assigns a queue for voice-signaling traffic that ensures 8 kbps. Note that this is optional and has nothing to do with good voice quality. Instead, it is a way to secure signaling.

CVOICE v4.2—7-18

These figures illustrate the commands that are used for implementing PPP QoS using the map classes and policies on the router.

VoIP over PPP QoS (Cont.)

Cisco.com

Multilink Interface Configuration

Notes: MLP is strictly an LFI mechanism. It does not bundle multiple serial interfaces to the same virtual interface as the name suggests. (This bundling is done for data and is *not* recommended for voice.) The end result may manifest itself as jitter and no audio.

```
interface Multilink1
 bandwidth 128
```

Notes: The bandwidth command needs to be set correctly for the right fragment size to be calculated.

```
 ip address 172.22.130.1 255.255.255.252
 ip tcp header-compression iphc-format
 service-policy output VOICE-POLICY
```

Notes: Sets up LLQ, an outbound operation applied to the outbound WAN interface.

```
 no cdp enable
 ppp multilink
 ppp multilink fragment-delay 10
```

Notes: The configured value of 10 sets the fragment size such that all fragments will have a 10ms maximum serialization delay.

```
ppp multilink interleave
multilink-group 1
```

Notes: This command links the multilink interface to the physical serial interface.

```
ip rtp header-compression
```

 CVOICE v4.2—7-19

The **ppp multilink interleave** command invokes interleaving of voice packets among the fragmented data packets. The **ip rtp header-compression** command invokes header compression for the RTP header. The **ppp multilink fragment-delay 10** command sets the fragment size such that the fragment delay does not exceed 10 ms.

VoIP over PPP QoS (Cont.)

Cisco.com

Interface Configuration

```
interface Serial0/0
 bandwidth 128
 no ip address
 encapsulation ppp
 clockrate 128000
 ppp multilink
 multilink-group 1
```

Notes: This command links the multilink interface to the physical serial interface.

 CVOICE v4.2—7-20

These figures continue the QoS commands.

VoIP over PPP QoS (Cont.)

Cisco.com

Dial-peer Configuration—Precedence

```
dial-peer voice 1 pots
 destination-pattern 6000
 port 1/0/0
!
dial-peer voice 2 voip
 destination-pattern 5000
 session target ipv4:192.168.10.2
 ip precedence 5
```

Notes: Sets the precedence value to critical for systems that do not understand DSCP.

Or:

CVOICE v4.2—7-21

VoIP over PPP QoS (Cont.)

Cisco.com

Dial-peer Configuration—DSCP

```
dial-peer voice 1 pots
 destination-pattern 6000
 port 1/0/0
!
dial-peer voice 2 voip
 destination-pattern 5000
 session target ipv4:192.168.10.2
 ip qos dscp ef media
```

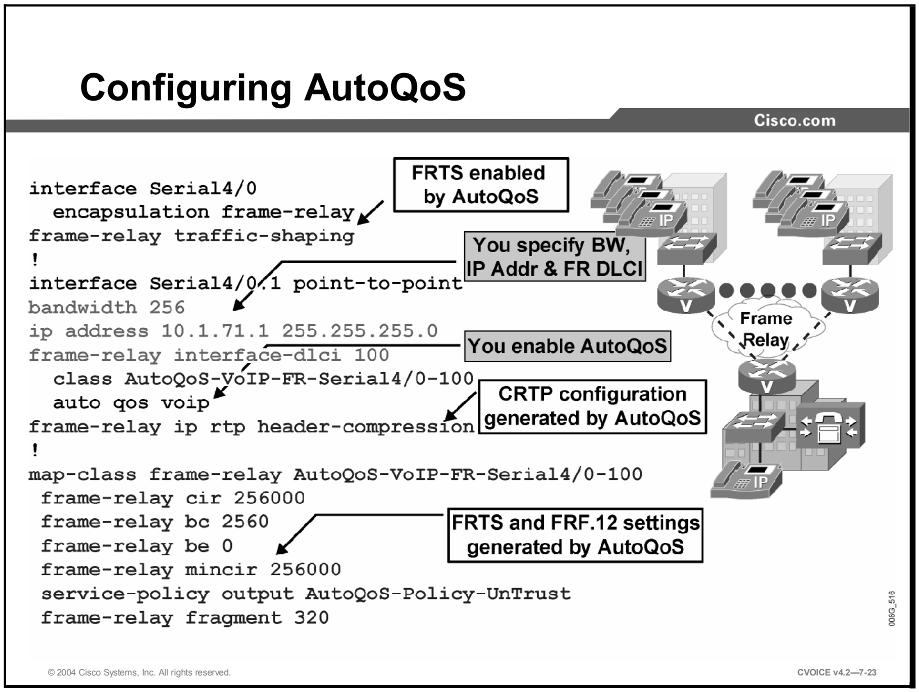
Notes: Sets the DSCP value to Express Forward for systems that understand DSCP. The **media** term applies DSCP to media payload packets.

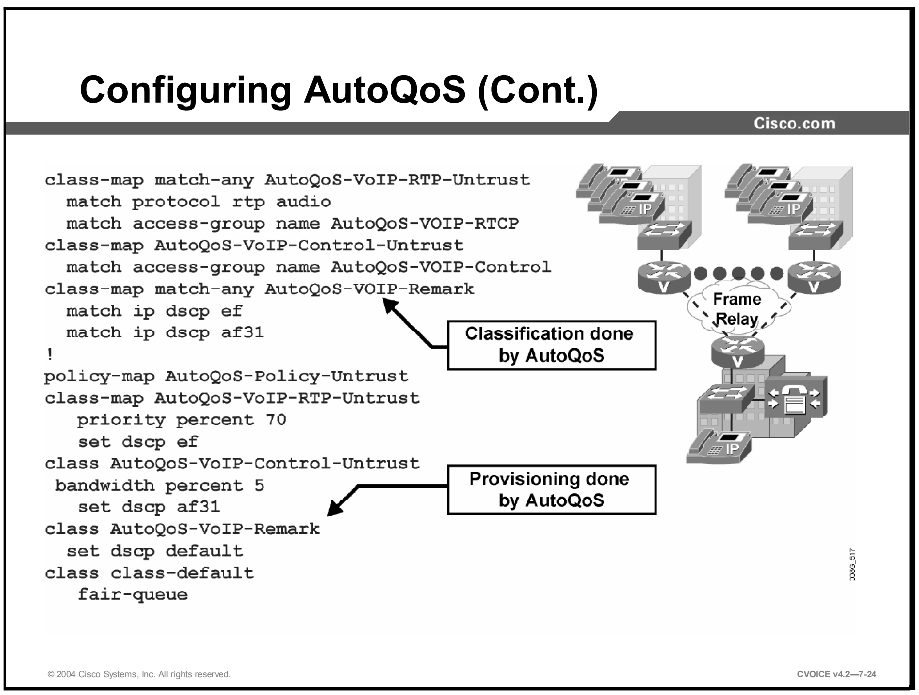
CVOICE v4.2—7-22

These figures illustrate the required configuration on dial peers.

Configuring AutoQoS

This topic describes Cisco AutoQoS and its associated command structure for routers.





Cisco AutoQoS is innovative technology that minimizes the complexity, time, and operating cost of QoS deployment. Cisco AutoQoS incorporates value-added intelligence into Cisco IOS software and Cisco Catalyst Operating Service software to provision and manage large-scale QoS deployments.

The first phase of Cisco AutoQoS targets VoIP deployments for customers who want to deploy IP telephony, but who lack the expertise or staffing to plan and deploy IP QoS and IP services.

Cisco AutoQoS automates consistent deployment of QoS features across Cisco routers and switches. It enables various Cisco QoS components based on the network environment and Cisco best-practice recommendations. With the increased prominence of delay-sensitive applications (voice, video, and other multimedia applications) deployed in networks today, proper QoS configuration will ensure high-quality application performance.

Currently, this presupposes a deep understanding of various QoS features (that is, queuing, dropping, traffic conditioning, queue-depth, drop thresholds, burst parameters, LFI, and CRTP) and the complexities of configuring many parameters associated with these features. Cisco AutoQoS helps overcome these difficulties by automatically configuring the device for Cisco QoS features and variables with the correct parameters.

Users can subsequently tune parameters that are generated by Cisco AutoQoS to suit their particular application needs, as desired.

Cisco AutoQoS performs the following functions:

On WAN interfaces:

- Automatically classifies RTP payload and VoIP control packets (H.323, H.225 Unicast, Skinny, session initiation protocol [SIP], MGCP)
- Builds service policies for VoIP traffic that are based on Cisco Modular QoS Command Line Interface (MQC)
- Provisions LLQ—Priority Queuing for VoIP bearer and bandwidth guarantees for control traffic
- Enables WAN traffic shaping that adheres to Cisco best practices, where required
- Enables link efficiency mechanisms, such as LFI and RTP header compression (CRTP), where required
- Provides SNMP and SYSLOG alerts for VoIP packet drops

On LAN interfaces:

- Enforces the trust boundary on Cisco Catalyst switch access ports, uplinks, and downlinks
- Enables Cisco Catalyst strict priority queuing (also known as expedite queuing) with WRR scheduling for voice and data traffic, where appropriate
- Configures queue admission criteria (maps CoS values in incoming packets to the appropriate queues)
- Modifies queue sizes and weights where required

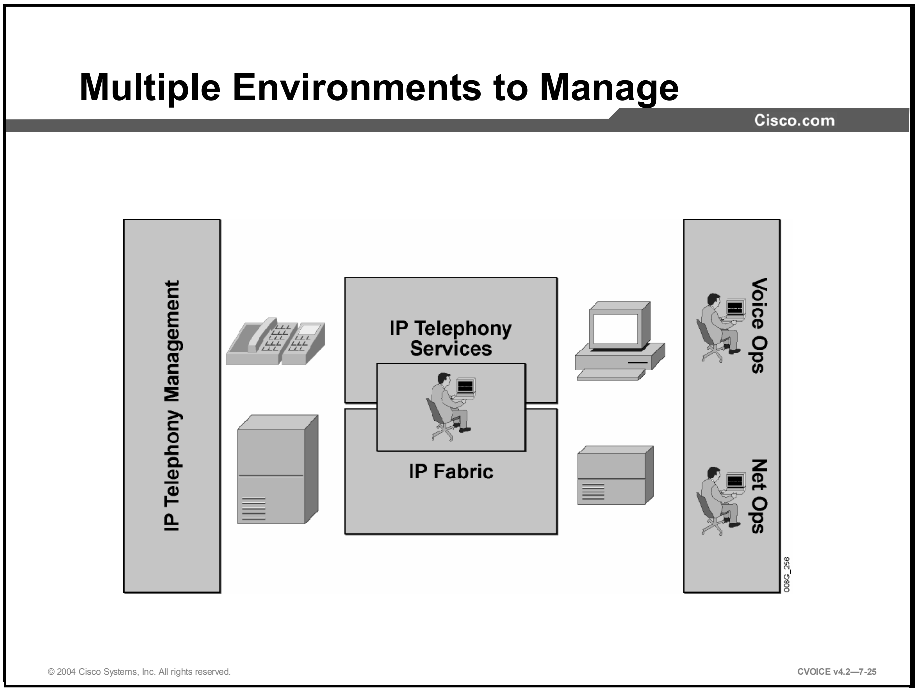
Note Cisco AutoQoS is available in the following Cisco IOS software releases: Cisco IOS Software Release 12.1E or later for the Cisco Catalyst 2950 and 3550 series switches; Cisco IOS Software Release 12.2T or later for the Cisco 2600, 2600XM, 3600, 3700, and 7200 series routers; Cisco IOS Software Release 12.1E or later for the Cisco Catalyst 4500 series switches; and Cisco Catalyst OS 7.5.1 or later for the Cisco Catalyst 6500 series switches.

Typically, QoS network design and implementation over multiple LAN and WAN sites is fairly complex and labor intensive. Customers wish to reduce deployment time, provisioning errors, and operating expenses to optimize their network for the applications, while retaining the flexibility to subsequently fine-tune QoS.

To expedite QoS deployment, the user interface must be simplified. Cisco AutoQoS addresses this by automating the five main aspects of QoS deployment (Application Classification, Policy Generation, Configuration, Monitoring and Reporting, and Consistency) while adding control plane intelligence to create a simple, accelerated, and tunable solution.

CiscoWorks QPM for QoS

This topic describes the basics of CiscoWorks QoS Policy Manager (QPM) as a management tool for QoS.

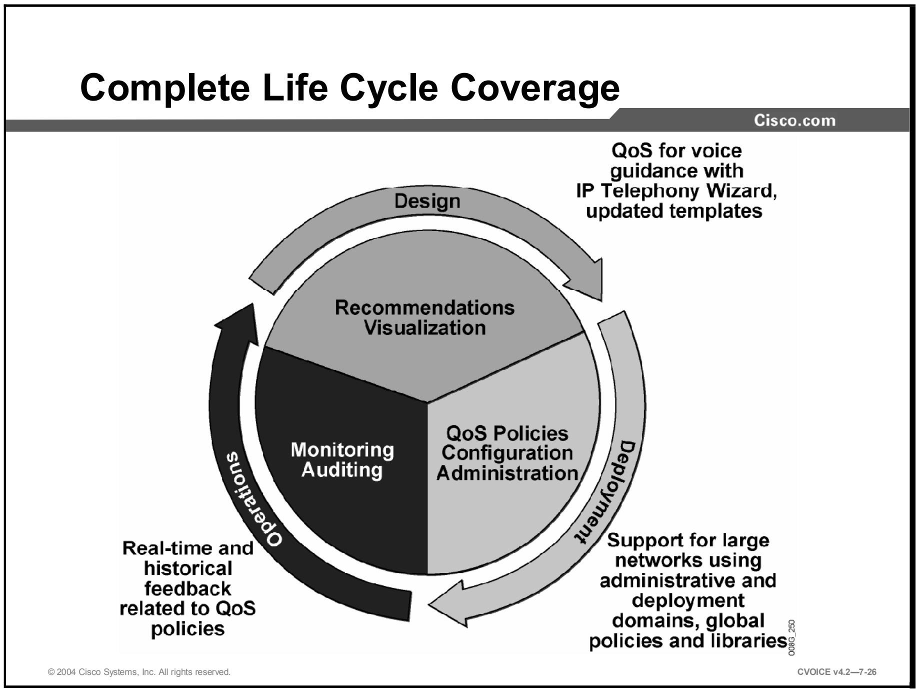


While the need for managing QoS in a voice-enabled environment is obvious, the configuration, deployment, and analysis of the requisite QoS policies may not be as intuitive.

The primary challenge for network managers is to consistently deploy policies in the devices that are located throughout a network to operate voice correctly. However, using the QoS features within intelligent network devices can be a complex exercise for network managers.

The sheer number of devices involved, coupled with the additional complexity of having different devices support different sets of QoS mechanisms, makes this a daunting task. To deploy these end-to-end intelligent services, network managers must use automated tools only. Without automated policy tools like QPM, it is highly likely that a user who tries to apply policies across a complex network will end up with inconsistencies.

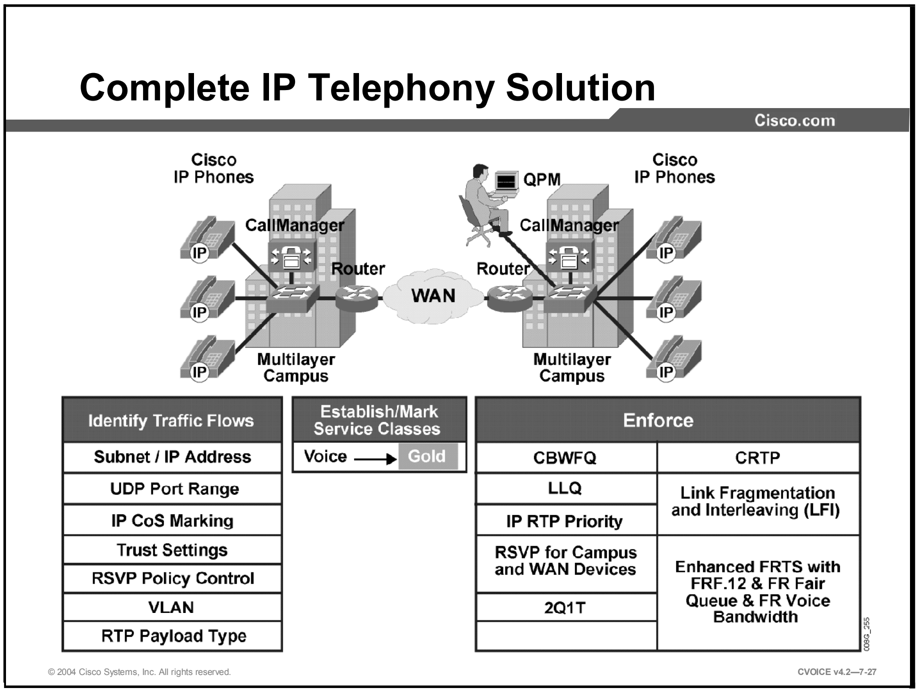
In networks today, users generally limit their application of intelligent tools in switches and routers for two reasons: fear of having inconsistent policies, or complexity and cost of implementing those policies in networks through Cisco IOS-based ACLs. Therefore, the solution is to have a centralized policy to manage networkwide QoS.



To effectively manage a voice and telephony-enabled network, administrators need tools that can simplify the configuration, deployment, management, and monitoring of QoS policies. Effective QoS management is an ongoing process. It entails the following tasks:

- Baseline monitoring of critical traffic flows to determine the policies that are needed
- Classifying applications into service classes
- Provisioning QoS with networkwide enforcement
- Validating QoS settings and results

The management tool must be able to deliver centralized QoS analysis and policy control for voice, video, and data networks. Additionally, the tool must enable networkwide, content-based DiffServ, and campus-to-WAN automated QoS configuration and deployment.

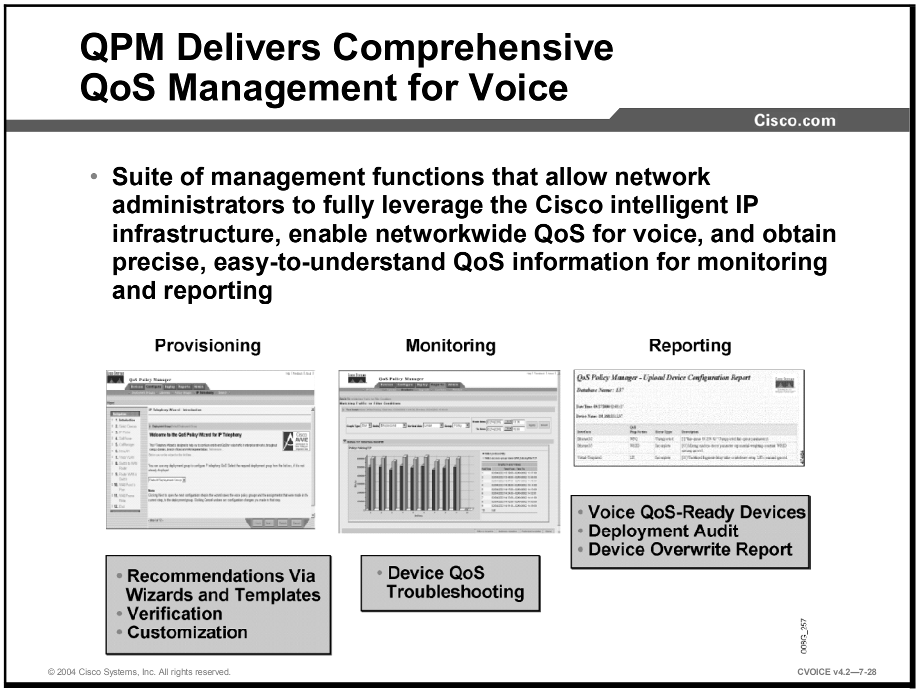


Using QPM, a network administrator quickly constructs rules-based QoS policies that identify and partition application traffic into multiple levels of service, ensuring that the most important applications receive priority service.

Example: QPM Applied

A business may establish differentiated "Gold," "Silver," and "Bronze" levels of IP services. A Gold service guarantees latency and delivery for the transport of real-time traffic, such as VoIP. A Silver service guarantees delivery for business-critical applications that require certain response times but are not as latency-sensitive, such as enterprise resource planning (ERP) or e-commerce. A Bronze service can support certain web and e-mail sessions while other traffic is treated on a best-effort basis.

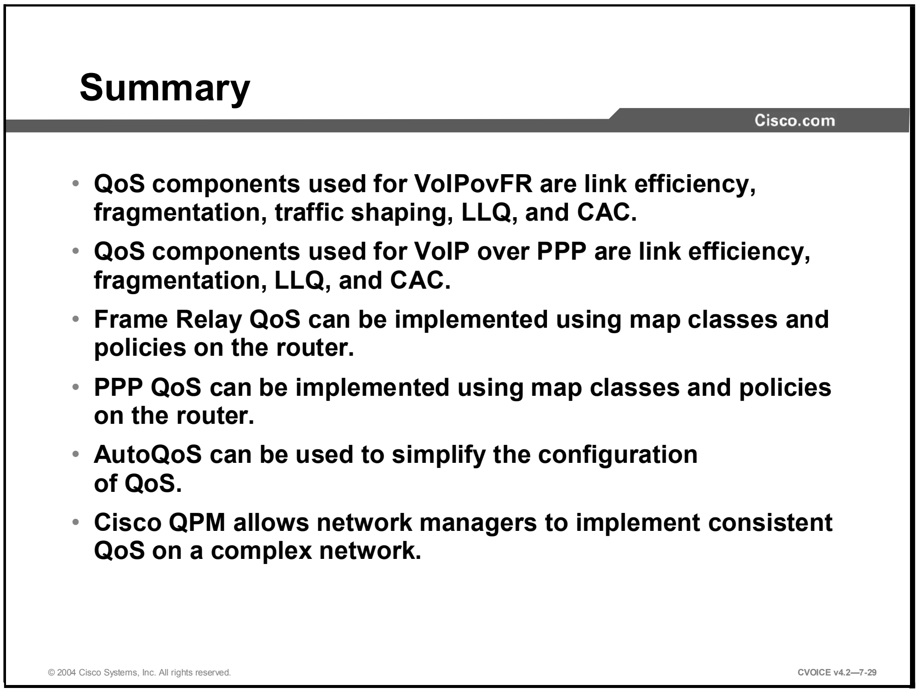
There are many QoS mechanisms built into Cisco Systems switches and routers to ensure that voice gets the required level of service in the network. QPM is an enabler for optimal voice quality in enterprise IP networks because it leverages the intelligent Cisco infrastructure to identify voice flows, such as IP precedence, DSCP value, or RTP payload type. It also ensures that voice gets Gold service, and enforces low latency and low packet drops through various mechanisms, such as placing LLQ on routers and voice packets in the PQ on Catalyst switches.



IP telephony usually requires a rather complex QoS configuration for the devices that carry voice traffic flows. IP telephony QoS management is one of the primary features of QPM. QPM includes a rules-based, step-by-step wizard with a built-in library of predefined QoS policy templates that are based on Cisco design recommendations. These policies allow for the accurate, speedy configuration of voice QoS mechanisms on switches and routers; they can also be modified as needed to meet specific network requirements. QPM includes a network voice-readiness report that lists devices that have all the required software and hardware to support QoS for voice. You can leverage QoS monitoring for troubleshooting and adjusting policies.

Summary

This topic summarizes the key points discussed in this lesson.



References

For additional information, refer to these resources:

- Technical Support
  http://www.cisco.com/pcgi-bin/Support/PSP/psp_view.pl?p=Internetworking:Voice:QoS
- AutoQoS
  http://www.cisco.com/en/US/products/hw/video/ps1870/products_qanda_item09186a0080134a66.shtml

Next Steps

For the associated lab exercise, refer to the following section of the course Lab Guide:

- Lab Exercise 7-1: QoS

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) Describe two ways that FRF.12 identifies fragmented frames. (Choose two.)

- A) by adding an extended header to the frame
- B) by including a 1-byte flag in the header
- C) by fragmenting the packets into fixed-size frames
- D) by adding sequence numbers to the frames
- E) by sending FRF.12 signaling to the destination

Q2) If FRTS is enabled on a link, what happens to voice frames when the network becomes congested?

- A) Frames that exceed specified bandwidth are rerouted over another link, such as a PSTN.
- B) Frames that exceed specified bandwidth are buffered and transmitted when congestion is reduced.
- C) Frames that exceed specified bandwidth are marked DE and dropped.
- D) Frames that exceed specified bandwidth are fragmented and given sequence numbers.

Q3) What is the size of a typical G.729 voice packet?

- A) 20 bytes
- B) 66 bytes
- C) 214 bytes
- D) 1500 bytes

Q4) How does LFI reduce delay and jitter for voice traffic in a WAN?

- A) It queues high-priority voice traffic ahead of data traffic.
- B) It fragments large voice packets and interleaves data packets among the fragments.
- C) It fragments large data packets and interleaves voice packets among the fragments.
- D) It reroutes voice traffic over faster connections.

Q5) What is the purpose of a policy map?

- A) defines the voice-signaling and traffic class maps
- B) defines how the link resources are assigned to the map classes
- C) assigns a queue for voice-signaling traffic
- D) defines how to classify traffic that does not fall into a defined class

Q6) Which QoS mechanism must be enabled for **map-class** to start?

A) LFI

B) LLQ

C) CAC

D) traffic shaping

Q7) What is the purpose of the **class class-default** command?

A) defines the voice-signaling and traffic class maps

B) defines how the link resources are assigned to the map classes

C) assigns a queue for voice-signaling traffic

D) defines how to classify traffic that does not fall into a defined class

Q8) During VoIP over PPP configuration, which command must you set properly to calculate the correct fragment size?

A) **bandwidth**

B) **fragment-delay**

C) **class voice-traffic**

D) **ip precedence**

Q9) Cisco AutoQoS automates consistent deployment of _____ features across Cisco routers and switches.

A) fragmentation

B) traffic shaping

C) FRTS

D) QoS

Q10) AutoQoS provides which three of the following? (Choose three.)

A) provides automatic configuration of bandwidth parameter on serial interfaces

B) builds service policies for VoIP traffic that are based on Cisco Modular QoS CLI (MQC)

C) provisions Low Latency Queuing (LLQ)—Priority Queuing for VoIP bearer and bandwidth guarantees for control traffic

D) enables WAN traffic shaping that adheres to Cisco best practices, where required

E) enables policing on WAN interfaces to reduce jitter and delay

Q11) What is the biggest advantage of using an automated policy tool such as CiscoWorks QPM?

A) It is less expensive than applying the policies manually.

B) It avoids inconsistencies in end-to-end policy application on a complex network.

C) It uses factory-recommended default settings for all devices.

D) It integrates well with all types of networks.

Q12) What type of information is supplied by the network voice-readiness report provided by QPM?

A) devices with software and hardware required to support QoS for voice

B) times of least congestion on the network

C) links that are best suited for transporting voice traffic

D) all the devices in the network that are experiencing congestion

Quiz Answer Key

Q1) A, D

Relates to: QoS Recommendations for VoIPovFR

Q2) B

Relates to: QoS Recommendations for VoIPovFR

Q3) B

Relates to: QoS Recommendations for VoIP over PPP

Q4) C

Relates to: QoS Recommendations for VoIP over PPP

Q5) B

Relates to: VoIPovFR Configuration

Q6) D

Relates to: VoIPovFR Configuration

Q7) D

Relates to: VoIP over PPP Configuration

Q8) A

Relates to: VoIP over PPP Configuration

Q9) D

Relates to: Configuring AutoQoS

Q10) B, C, D

Relates to: Configuring AutoQoS

Q11) B

Relates to: CiscoWorks QPM for QoS

Q12) A

Relates to: CiscoWorks QPM for QoS

Configuring CAC

Overview

To prevent oversubscription of VoIP networks, the number of voice calls allowed on the network must be limited. This lesson describes the reasons for Call Admission Control (CAC) and its configuration parameters.

Relevance

CAC must be implemented in a VoIP network to limit the use of bandwidth. Network administrators should understand the need for CAC and how to configure it.

Objectives

Upon completing this lesson, you will be able to configure call control on the network. This includes being able to meet these objectives:

- State the reasons for CAC
- Describe the effects of bandwidth oversubscription on overall voice quality
- State the function of CAC as it relates to overall call control services
- Describe the operation of RSVP
- Describe the three distinct groups of CAC mechanisms
- Describe how H.323 implements CAC
- Describe how SIP implements CAC
- Describe how MGCP implements CAC
- State the two types of call admission that Cisco CallManager allows

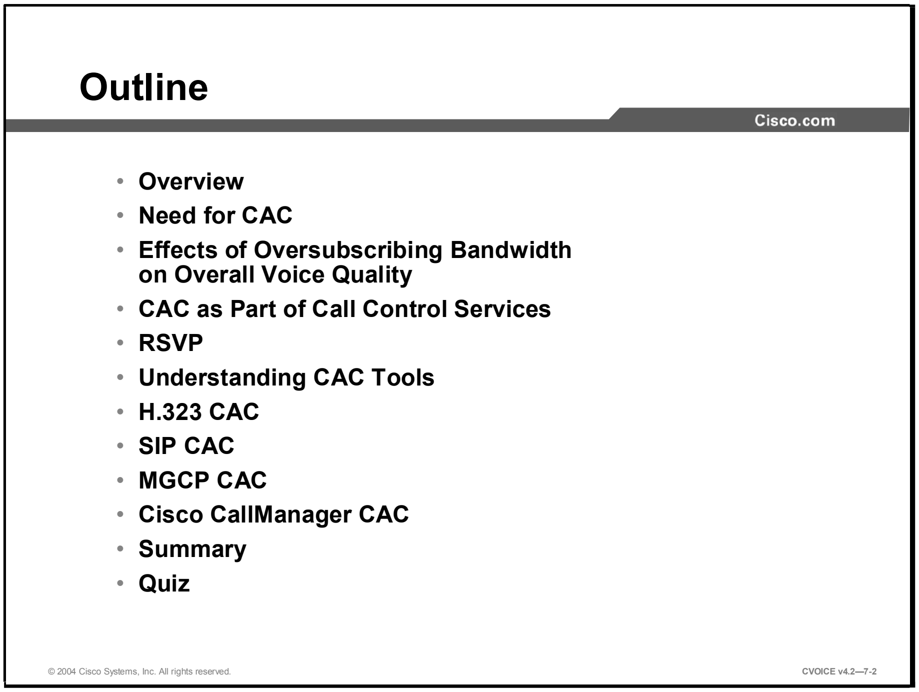
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Knowledge of the basics of VoIP
- Knowledge of TCP/IP networks

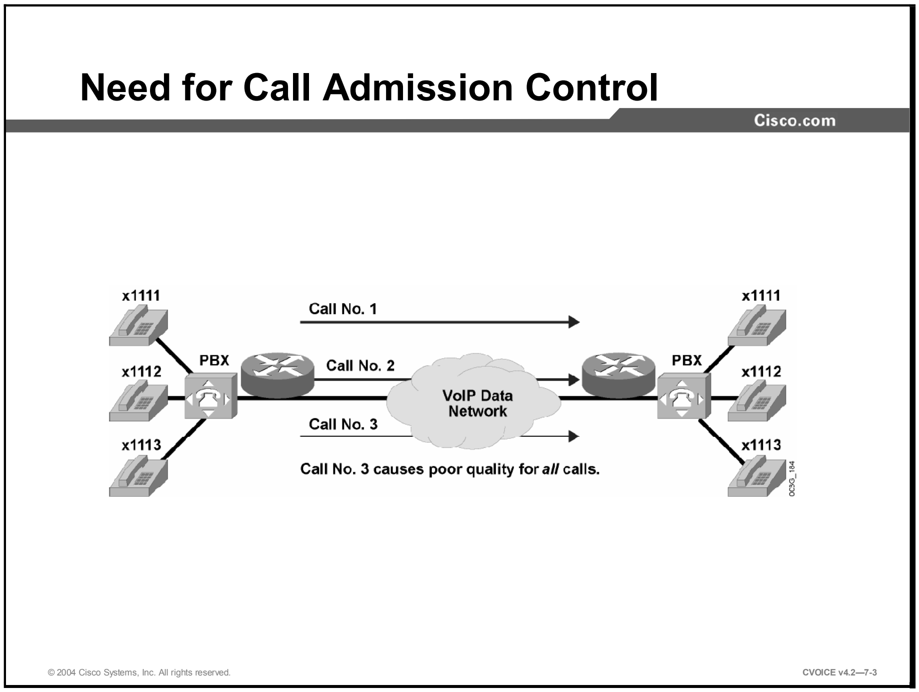
Outline

The outline lists the topics included in this lesson.



Need for CAC

This topic explains why CAC is needed.



CAC is a concept that applies to voice traffic only, not data traffic. If an influx of data traffic oversubscribes a particular link in the network, queuing, buffering, and packet drop decisions resolve the congestion. The extra traffic is simply delayed until the interface becomes available to send the traffic, or, if traffic is dropped, the protocol or the end user initiates a timeout and requests a retransmission of the information.

Because real-time traffic is sensitive to latency and packet loss, resolving network congestion when real-time traffic is present will jeopardize the QoS. For real-time delay-sensitive traffic such as voice, it is better to deny network access under congestion conditions than to allow traffic on the network to be dropped and delayed. Dropped or delayed network traffic causes intermittent impaired QoS and results in customer dissatisfaction.

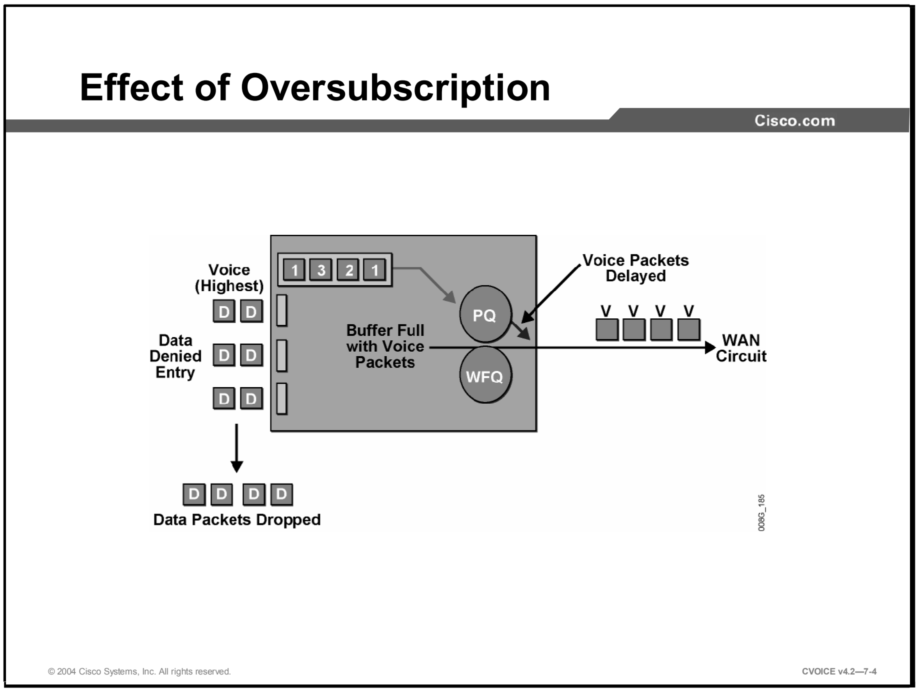
CAC is a determining and informed decision that is made before a voice call is established. CAC is based on whether the required network resources are available to provide suitable QoS for the new call.

Example: CAC Applied

CAC mechanisms extend the capabilities of QoS tools to protect voice traffic from the negative effects of other voice traffic and to keep excess voice traffic off the network. The figure illustrates the need for CAC. If the WAN access link between the two PBXs has the bandwidth to carry only two VoIP calls, admitting the third call impairs the voice quality of all three calls.

Effects of Oversubscribing Bandwidth on Overall Voice Quality

This topic describes the effects of network link oversubscription.



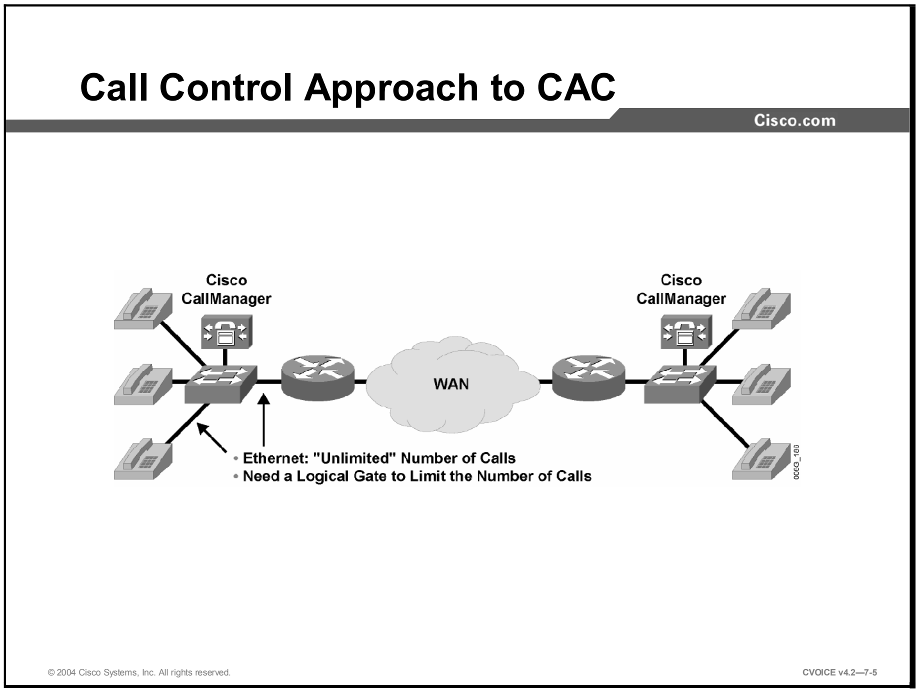
QoS tools such as queuing ensure that voice traffic receives priority over data traffic. If a network link is oversubscribed with too much voice traffic, data packets are dropped and the remaining voice calls suffer because they must compete for bandwidth from the low latency queue.

Example: Oversubscription

The figure illustrates the effect of oversubscription. Note that Priority Queue is allowed to forward packets while the data packets, destined for the Weighted Fair Queue, are denied entry to the queue and are dropped. In this case, the PQ buffer is full and so the voice packets are competing with other voice packets for access to the network link. This results in a degradation of all voice calls on this link.

CAC as Part of Call Control Services

This topic describes CAC as a function of call control services.



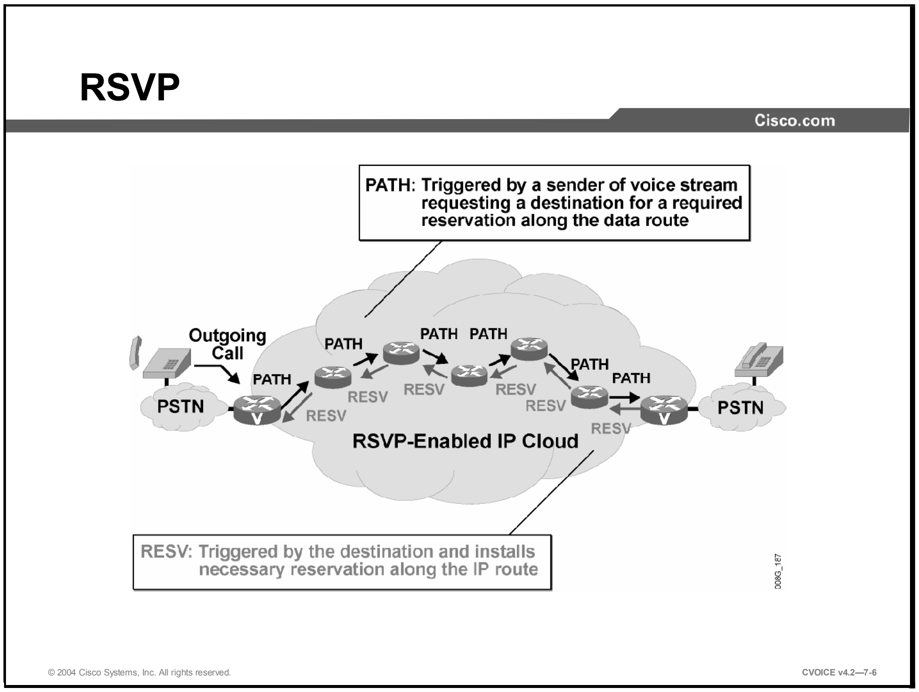
CAC, as part of call control services, functions on the outgoing gateway. CAC bases its decision on nodal information, such as the state of the outgoing LAN or WAN link. If the local packet network link is down, there is no point in executing complex decision logic based on the state of the rest of the network, because that network is unreachable. Local mechanisms include configuration items that disallow all calls that exceed a specified number.

Example: Call Control CAC

If the network designer already knows that bandwidth limitations allow no more than five calls across the outgoing WAN link, then the local node can be configured to allow no more than five calls. You can configure this type of CAC on outgoing dial peers.

RSVP

This topic describes Resource Reservation Protocol (RSVP).



RSVP is the only CAC mechanism that makes a bandwidth reservation and does not make a call admission decision based on a best guess before the call is set up. This gives RSVP the unique advantage of not only providing CAC for voice, but also guaranteeing the QoS against changing network conditions for the duration of the call. The RSVP reservation is made in both directions because a voice call requires a two-way speech path and bandwidth in both directions.

The terminating gateway ultimately makes the CAC decision based on whether both reservations succeed. At that point, the H.323 state machine continues with either an H.225 Alerting/Connect (the call is allowed and proceeds), or with an H.225 Reject/Release (the call is denied). The RSVP reservation is in place by the time the destination phone starts ringing and the caller hears ringback.

RSVP has the following important differences from other CAC methods discussed in this lesson:

- The ability to maintain QoS for the duration of the call.
- An awareness of topology. In concept, the RSVP reservation installs on every interface that the call will traverse through the network. RSVP ensures bandwidth over every segment without any requirement to know the actual bandwidth provisioning on each interface or the path on which the routing protocols direct the packets. RSVP, therefore, adjusts automatically to network configuration changes, and no manual calculations are necessary to keep different aspects of the configuration synchronized.
- To function correctly, RSVP is dependent on the correct configuration for all devices in the network. (It can have a scaling issue depending on how the network is designed.)

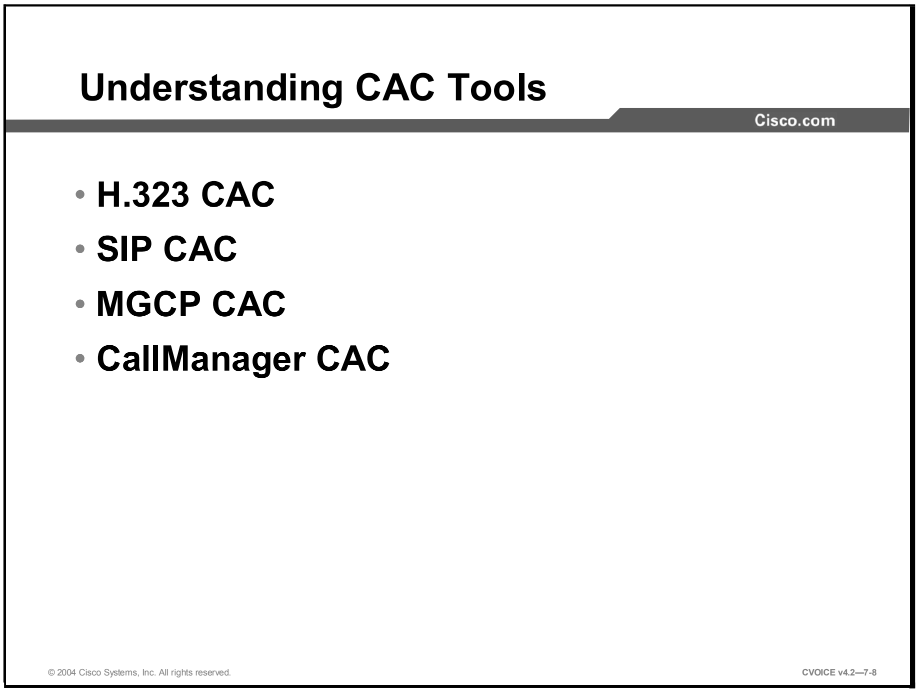
- RSVP provides end-to-end reservation per call and has visibility for that call only. RSVP is unaware of how many other calls are active from a site or across an interface, or the source or destination of any other call.

Example: RSVP

Configuring RSVP in Cisco routers allows the administrator to limit the amount of bandwidth requested per call and the total amount of bandwidth allowed for all calls. This configuration is entered directly against the interface that will permit or deny the calls. The configuration also requires RSVP to be configured on the dial peers for the calls that will be managed by RSVP.

Understanding CAC Tools

This topic describes CAC tools available for various protocols and systems.



As the many interesting aspects of CAC on packet networks have been considered, several different solutions have come into prominence. None of them solves the entire problem, but they all are useful to address a particular aspect of CAC. Unlike circuit-based networks, which reserve a free DS0 time slot on every leg of the path the call will take, determining whether a packet network has the resources to carry a voice call is not a simple undertaking.

There are four areas in which CAC may be implemented. These areas are:

- H.323 CAC
- SIP CAC
- MGCP CAC
- CallManager CAC

Each area is associated with a specific protocol or system. Each of these areas will be explored in the following figures.

H.323 CAC

This topic describes the configuration options available for H.323 CAC.

H.323 CAC

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- **call threshold** {**global** *trigger-name* | **interface** *interface-name interface-number* **int-calls**} **low** *value* **high** *value* [**busyout** | **treatment**]
- **call spike** *call-number* [**steps** *number-of-steps* **size** *milliseconds*]
- **call treatment** {**on** | **action** *action* [*value*] | **cause-code** *cause-code* | **isdn-reject** *value*}

CVOICE v4.2—7-9

The CAC for the H.323 VoIP gateways feature allows you to configure thresholds for local resources, memory, and CPU resources.

With the **call threshold** command, you can configure two thresholds, high and low, for each resource. Call treatment is triggered when the current value of a resource exceeds the configured high. The call treatment remains in effect until the current resource value falls below the configured low. Having high and low thresholds prevents call admission flapping and provides hysteresis in call admission decision making.

With the **call spike** command, you can configure the limit for incoming calls during a specified time period. A call spike is the term for when a large number of incoming calls arrive from the PSTN in a very short period of time; for example, 100 incoming calls in 10 milliseconds.

With the **call treatment** command, you can select how the call should be treated when local resources are not available to handle the call. For example, when the current resource value for any one of the configured triggers for call threshold has exceeded the configured threshold, the call treatment choices are as follows:

- **Time-division multiplexing (TDM) hairpinning:** Hairpins the calls through the POTS dial peer
- **Reject:** Disconnects the call
- **Play message or tone:** Plays a configured message or tone to the user

To enable the global resources of this gateway, use the **call threshold** command in global configuration mode. To disable this command, use the **no** form of this command.

> **call threshold** {**global** *trigger-name* | **interface** *interface-name interface-number* **int-calls**} **low** *value* **high** *value* [**busyout** | **treatment**]
>
> **no call threshold** {**global** *trigger-name* | **interface interface-name int-calls**}

Call Threshold Commands

Command	Description
global *trigger-name*	Specifies the global resources on the gateway The *trigger-name* arguments are as follows: ■ **cpu-5sec:** CPU utilization in the last 5 seconds ■ **cpu-avg:** Average CPU utilization ■ **io-mem:** IO memory utilization ■ **proc-mem:** Processor memory utilization ■ **total-calls:** Total number of calls. The valid range is from 1 to 10,000. ■ **total-mem:** Total memory utilization
interface *interface-name interface-number*	Specifies the gateway. The types of interfaces and their numbers will depend upon the configured interfaces.
int-calls	Number of calls through the interface. The valid range is from 1 to 10,000 calls**.**
low *value*	Value of low threshold. The valid range is from 1 to 100 percent for the utilization triggers.
high *value*	Value of high threshold. The valid range is from 1 to 100 percent for the utilization triggers.
busyout	(Optional—global only) Automatically busies out the T1/E1 channels if the resource is not available.
treatment	(Optional—global only) Applies call treatment from session application if the resource is not available.

To configure the limit of incoming calls in a short period of time, use the **call spike** command in global configuration mode. To disable this command, use the **no** form of this command. The **call spike** command uses a sliding window to determine the period in which the spike is limited. The sliding window period is defined using the **size** command, with valid ranges from 100 to 250 ms. If a longer spike period is desired, the **steps** command is used as a multiplier for the **size** command. For example, if the **steps** were set to 2 and the **size** was set to 250, the spike period would be 500 ms.

> **call spike** *call-number* [**steps** *number-of-steps* **size** *milliseconds*]
>
> **no call spike**

Call Spike Commands

Command	Description
call-number	Incoming call numbers for spiking threshold; valid range is from 1 to 2,147,483,647
steps *number-of-steps*	(Optional) Number of steps; valid range is from 3 to 10
size *milliseconds*	(Optional) Step size in milliseconds; valid range is from 100 to 2000

To configure how calls should be processed when local resources are unavailable, use the **call treatment** command in global configuration mode. To disable the call treatment triggers, use the **no** form of this command.

call treatment {**on** | **action** *action* [*value*] | **cause-code** *cause-code* | **isdn-reject** *value*}

no call treatment {**on** | **action** *action* [*value*] | **cause-code** *cause-code* | **isdn-reject** *value*}

Call Treatment Commands

Command	Description
on	Enables call treatment from default session application
action *action*	Action to take when call treatment is triggered. The *action* argument has the following possible values: ■ **hairpin:** Hairpin ■ **playmsg:** Specifies the audio file to play (URL) ■ **reject:** Disconnect the call and pass down cause code
value	(Optional) (For the *action* playmsg argument only) Specifies the audio file to play; URL format
cause-code *cause-code*	Specifies reason for disconnect to caller The *cause-code* argument can have the following values: ■ **busy:** Indicates that gateway is busy ■ **no-QoS:** Indicates that the gateway cannot provide QoS ■ **no-resource:** Indicates that the gateway has no resources available
isdn-reject *value*	Selects the ISDN reject cause-code. The *value* argument has the following: ■ 34–47 (ISDN cause code for rejection)

ISDN Cause Codes

Cause No.	Description	Function
34	No circuit available (circuit/channel congestion)	Indicates that there is no channel available to handle the call.
38	Net out of order	Indicates that the network is not functioning properly and it is likely to last a long time. Re-attempting the call is not likely to be successful.
41	Net problem, redial (temporary failure)	Indicates that the network is not functioning properly and it is not going to last a long time. Re-attempting the call is likely to be successful.
42	Net busy, redial (switching equipment congestion)	Indicates that the switching equipment is experiencing high traffic load.
43	Access/user information discarded	Indicates that the network is unable to deliver user information to the remote users as was requested.
44	No channel available (requested circuit/channel not available)	Indicates that the circuit or channel indicated by the requesting side cannot be used by the other side of the interface.
47	Resource unavailable/new destination	Indicates a resource unavailable event only when no other cause in the resource unavailable class applies.

Example: H.323 CAC Configuration

The following example will busyout the total-calls resource if 5 (low) or 5000 (high) is reached:

```
call threshold global total-calls low 5 high 5000 busyout
```

The following example enables thresholds of 5 (low) and 2500 (high) for interface calls on interface Ethernet 0:

```
call threshold interface Ethernet 0 int-calls low 5 high 2500
```

The following example will busyout the average CPU utilization if 5 percent (low) or 65 percent (high) is reached:

```
call threshold global cpu-avg low 5 high 65 busyout
```

The following configuration of the **call spike** command has a call number of 30, 10 steps, and a step size of 2000 milliseconds:

```
call spike 30 steps 10 size 2000
```

The following example enables the call treatment feature with a hairpin action:

```
call treatment on
call treatment action hairpin
```

The following example displays proper formatting of the **playmsg action** keyword:

```
call treatment action playmsg tftp://keyer/prompts/conjestion.
au
```

Note The **congestion.au** file plays when local resources are not available to handle the call.

The following example configures a call treatment cause-code to display no-QoS when local resources are unavailable to process a call:

```
call treatment cause-code no-qos
```

SIP CAC

This topic describes the configuration options available for session initiation protocol (SIP) CAC.

SIP CAC

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- **SAA RTR Responder**
  – rtr responder
- **PSTN Fallback**
  – call fallback active
- **Resource Availability Check**
  – call threshold global *trigger-name* low *value* high *value* [busyout][treatment]
  – call treatment {on | action *action* [*value*]| cause-code *cause-code* | isdn-reject *value*}
  – call threshold interface *interface-name interface number* int-calls low *value* high *value*

 CVOICE v4.2—7-10

The measurement-based CAC for SIP features support within SIP to monitor IP network capacity and reject or redirect calls based on congestion detection. This feature does the following:

- Verifies that adequate resources are available to carry a successful VoIP session
- Implements a mechanism to prevent calls arriving from the IP network from entering the gateway when required resources are not available to process the call
- Supports measurement-based CAC processes

Configuring SAA RTR Responder

Service Assurance Agent (SAA) is a generic network management feature that provides a mechanism for network congestion analysis. SAA determines latency, delay, and jitter and provides real-time Calculated Planning Impairment Factor (ICPIF) calculations before establishing a call across an IP infrastructure. The SAA Responder feature uses SAA probes to traverse the network to a given IP destination and measure the loss and delay characteristics of the network along the path traveled. These values are returned to the outgoing gateway to use in making a decision on the condition of the network and its ability to carry a call. Threshold values for rejecting a call are configured at the outgoing gateway.

Each probe consists of multiple packets, a configurable parameter of this feature. SAA packets emulate voice packets and receive the same priority as voice throughout the entire network. The delay, loss, and ICPIF values entered into the cache for the IP destination are averaged from all the responses. If the call uses G.729 and G.711 codecs, the probe packet sizes mimic those of a voice packet for that codec. Other codecs use G.711-like probes. In Cisco IOS software releases later than Release 12.1(3)T, other codec choices may also be supported with their own specific probes.

The IP precedence of the probe packets can also be configured to simulate the priority of a voice packet more closely. This parameter should be set equal to the IP precedence used for other voice media packets in the network.

SAA probes used for CAC go out randomly on ports selected from within the top end of the audio UDP-defined port range (16384 to 32767). Probes use a packet size based on the codec the call will use. IP precedence can be set if desired, and a full RTP, UDP, or IP header is used, just as a real voice packet would carry. The SAA Responder feature was called Response Time Reporter (RTR) in earlier releases of Cisco IOS software.

Use the **rtr responder** command to enable SAA Responder functionality on the destination node.

Configuring PSTN Fallback

The measurement-based CAC for SIP feature supports PSTN Fallback, which monitors congestion in the IP network and either redirects calls to the PSTN or rejects calls based on network congestion. Calls can be rerouted to an alternate IP destination or to the PSTN if the IP network is found unsuitable for voice traffic at that time. You can define congestion thresholds based on the configured network. This functionality allows the service provider to give a reasonable guarantee about the quality of the conversation to VoIP users at the time of call admission.

Note PSTN Fallback does not provide assurances that a VoIP call that proceeds over the IP network is protected from the effects of congestion. This is the function of the other QoS mechanisms, such as IP RTP or LLQ.

PSTN Fallback includes the following capabilities:

- Provides the ability to define the congestion thresholds based on the network
  - Defines a threshold based on ICPIF, which is derived as part of ITU G.113
  - Defines a threshold based solely on packet delay and loss measurements
- Uses SAA probes to provide packet delay, jitter, and loss information for the relevant IP addresses. Based on the packet loss, delay, and jitter encountered by these probes, an ICPIF or delay or loss value is calculated.
- Supports calls of any codec. Only G.729 and G.711 have accurately simulated probes. Calls of all other codecs are emulated by a G.711 probe.

The call fallback subsystem has a network traffic cache that maintains the ICPIF or delay or loss values for various destinations. This capability helps performance, because each new call to a well-known destination need not wait for a probe to be admitted because the value is usually cached from a previous call.

Once the ICPIF or delay or loss value is calculated, they are stored in a fallback cache where they remain until the cache ages out or overflows. Until an entry ages out, probes are sent periodically for that particular destination. This time interval is configurable.

To configure PSTN Fallback, use the following command:

```
call fallback active
```

This command enables a call request to fall back to alternate dial peers in case of network congestion. The **active** keyword enables a call request to fall back to alternate dial peers in case of network congestion.

Configuring Resource Availability Check

User-selected thresholds allow you to configure call admission thresholds for local resources and end-to-end memory and CPU resources. You can configure two thresholds, high and low, for each global or interface-related resource. The specified call treatment is triggered when the current value of a resource goes beyond the configured high, and remains in effect until the current resource value falls below the configured low.

You can select how the call should be treated when local resources are not available to handle the call. For example, when the current resource value for any one of the configured triggers for call threshold exceeds the configured threshold, you have the following call treatment choices:

- **TDM hairpinning:** Hairpins the calls through the POTS dial peer
- **Reject:** Disconnects the call
- **Play message or tone:** Plays a configured message or tone to the user

To configure Resource Availability Check, use the following command:

```
call threshold global trigger-name low value high value
   [busyout] [treatment]
```

This command enables a trigger and defines associated parameters to allow or disallow new calls on the router. Action is enabled when the trigger value exceeds the value specified by the **high** keyword and is disabled when the trigger value drops below the value specified by the **low** keyword.

Call Threshold Global Commands

Command	Description
trigger-name	The *trigger-name* argument can be one of the following: ■ **cpu-5sec:** CPU utilization in the last 5 seconds ■ **cpu-avg:** Average CPU utilization ■ **io-mem:** IO memory utilization ■ **proc-mem:** Processor memory utilization ■ **total-calls:** Total number of calls ■ **total-mem:** Total memory utilization
low *value*	Value of low threshold; range is from 1 to 100 percent for utilization triggers and from 1 to 10,000 for total calls
high *value*	Value of high threshold; range is from 1 to 100 percent for utilization triggers and from 1 to 10,000 for total-calls
busyout	(Optional) Busies out the T1 or E1 channels if the resource is not available
treatment	(Optional) Applies call treatment from the session application if the resource is not available

To configure call treatment, use the following command:

> **call treatment** {**on** | **action** *action* [*value*]| **cause-code** *cause-code* | **isdn-reject** *value*}

This command configures how calls should be processed when local resources are unavailable.

Call Treatment Commands

Command	Description
on	Enables call treatment from the default session application
action *action*	Specifies the action to be taken when call treatment is triggered The *action* argument can be one of the following: ■ **hairpin:** Specifies hairpinning action ■ **playmsg:** Specifies that the gateway play the selected message. The optional *value* argument specifies the audio file to play in URL format. ■ **reject:** Specifies whether the call should be disconnected and the ISDN cause code passed
cause-code *cause-code*	Specifies the reason for disconnection to the caller The *cause-code* argument can be one of the following: ■ **busy:** Indicates that the gateway is busy ■ **no-QoS:** Indicates that the gateway cannot provide QoS ■ **no-resource:** Indicates that the gateway has no resources available
isdn-reject *value*	Applies to ISDN interfaces only and specifies the ISDN reject cause code. Range for the *value* argument is 34 to 47 (ISDN cause code for rejection)

To configure Resource Availability Checking for interface resources, perform the following task.

> **call threshold interface** *interface-name interface number* **int-calls low** *value* **high** *value*

This command allows threshold values to be configured for total numbers of voice calls placed through a particular interface. This command is used to allow or disallow admission for new calls on the router.

Call Threshold Interface Commands

Command	Description
interface-name	Specifies the interface used in making call admission decisions. Types of interfaces and their numbers will depend upon the configured interfaces.
interface-number	Specifies the number of calls through the interface that triggers a call admission decision
int-calls	Configures the gateway to use the number of calls through the interface as a threshold
low	Enables the specified call treatment until the number of calls through the interface drops below the configured low *value*. The *value* argument specifies the number of calls used to make call admission decisions. The range is from 1 to 10,000 calls.
high	Enables the specified call treatment until the number of calls through the interface exceeds the configured high *value*. The *value* argument specifies the number of calls used to make call admission decisions. The range is from 1 to 10,000 calls.

Example: SIP CAC Configuration

The following are examples of uses of the SIP CAC commands:

- SAA RTR Responder

  Router(config)# rtr responder

- PSTN Fallback

  Router(config) # call fallback active

- Resource Availability Check

  Router(config)# call threshold global total-calls low 5 high 1000 busyout

  Router(config)# call treatment action cause-code 17

  Router(config)# call threshold interface ethernet 0 int-calls low 5 high 2500

MGCP CAC

This topic describes the configuration options available for MGCP CAC.

MGCP CAC

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- **System Resource Check (SRC) CAC**
  - **call threshold global** *trigger-name* **low** *value* **high** *value* **treatment**
- **Resource Reservation Protocol (RSVP) CAC**
  - **ip rsvp bandwidth** [*interface-kbps* [*single-flow-kbps*]]
- **Cisco Service Assurance Agent (SAA) CAC**
  - **call fallback active**
  - **mgcp rtrcac**
  - **rtr responder**

 CVOICE v4.2—7-11

The Media Gateway Control Protocol (MGCP) VoIP CAC feature enables certain Cisco CAC capabilities on VoIP networks that are managed by MGCP call agents. These capabilities permit the gateway to identify and gracefully refuse calls that are susceptible to poor voice quality.

Poor voice quality on an MGCP voice network can result from transmission artifacts such as echo, from the use of low-quality codecs, from network congestion and delay, or from overloaded gateways. The first two causes can be overcome by using echo cancellation and better codec selection. The last two causes are addressed by MGCP VoIP CAC.

Before the release of MGCP VoIP CAC, MGCP voice calls were often established regardless of the availability of resources for those calls in the gateway and the network. MGCP VoIP CAC ensures resource availability by disallowing calls when gateway and network resources are below configured thresholds and by reserving guaranteed bandwidth throughout the network for each completed call.

MGCP VoIP CAC has three components for improving voice quality and reliability:

- System Resource Check (SRC) CAC evaluates memory and call resources local to the gateway. It is supported in MGCP 1.0 and MGCP 0.1.
- RSVP CAC surveys bandwidth availability on the network. It is supported in MGCP 1.0 and MGCP 0.1.
- Cisco Service Assurance Agent (SAA) CAC appraises network congestion conditions on the network. It is supported only in MGCP 1.0.

Configuring SRC CAC

To set thresholds and enable MGCP SRC CAC, use the following command in global configuration mode:

call threshold global *trigger-name* **low** *value* **high** *value* **treatment**

This command enables a resource and defines its parameters. Treatment of attempted calls is enabled when the resource cost goes beyond the high value. Treatment is not disabled until the resource cost drops below the low value. The arguments and keywords are as follows:

Call Threshold Global Commands

Command	Description
trigger-name	The *trigger-name* argument can be one of the following: ■ **cpu-5sec:** CPU utilization in the last 5 seconds ■ **cpu-avg:** Average CPU utilization ■ **io-mem:** IO memory utilization ■ **proc-mem:** Processor memory utilization ■ **total-calls:** Total number of calls ■ **total-mem:** Total memory utilization
low *value*	Value of low threshold; range is from 1 to 100 percent for utilization triggers and from 1 to 10,000 for total calls
high *value*	Value of high threshold; range is from 1 to 100 percent for utilization triggers and from 1 to 10,000 for total-calls
treatment	(Optional) Applies call treatment from the session application if the resource is not available

If network conditions rise above the high threshold value, SRC rejects the call by sending the call agent an MGCP error message with the return code 403. The call agent applies a treatment to the rejected call.

Configuring RSVP CAC

To configure MGCP RSVP CAC on a media gateway, use the following command in global configuration mode:

ip rsvp bandwidth (*interface-kbps* [*single-flow-kbps*])

This command enables RSVP for IP on an interface. RSVP is disabled by default. It should be noted that, in order for RSVP to operate correctly end-to-end, it must be configured on all routers in the network. The arguments are as follows:

IP RSVP Bandwidth Commands

Command	Description
interface-kbps	(Optional) Maximum amount of bandwidth, in kbps, that may be allocated by RSVP flows. The range is from 1 to 10,000,000. This should be configured for the maximum amount of voice bandwidth that this interface is limited to for all total calls.
single-flow-kbps	(Optional) Maximum amount of bandwidth, in kbps, that may be allocated to a single flow. The range is from 1 to 10,000,000. This should be configured for the amount of bandwidth for one call.

Configuring Cisco SAA CAC

The Cisco SAA is an application-aware synthetic operation agent that monitors network performance by measuring response time, network resource availability, application performance, jitter (interpacket delay variance), connect time, throughput, and packet loss. Performance can be measured between any Cisco device that supports this feature and any remote IP host (server), Cisco routing device, or mainframe host. Performance measurement statistics provided by this feature can be used for troubleshooting, problem analysis, and designing network topologies.

The SAA Responder that is enabled using the **rtr responder** command is a component embedded in the target Cisco routing device that allows the system to anticipate and respond to SAA request packets. The responder can listen on any user-defined port for UDP and TCP protocol messages. In client-server terminology, the SAA Responder is a concurrent multiservice server.

The commands to configure Cisco SAA CAC are as follows:

- **call fallback active:** Enables a call request to fall back to alternate dial peers in case of network congestion
- **mgcp rtrcac:** Enables MGCP SAA CAC
- **rtr responder:** Enables the SAA responder functionality on a Cisco device

Example: MGCP VoIP CAC on a Trunking Gateway

This configuration enables all three types of MGCP VoIP CAC: SRC, RSVP, and SAA. Comment lines are provided above the CAC commands to help you identify the commands needed for a particular CAC type.

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname eastcoast
!
!
voice-card 2
!
```



```
voice-card 3
!
ip subnet-zero
ip dhcp smart-relay
!
! The following command is used in MGCP SA Agent CAC.
call fallback active
! The following command is used in MGCP RSVP CAC.
call rsvp-sync
! The following six commands are used in MGCP SRC CAC.
call threshold global cpu-5sec low 55 high 70 treatment
call threshold global cpu-avg low 70 high 80 treatment
call threshold global total-mem low 70 high 80 treatment
call threshold global io-mem low 70 high 80 treatment
call threshold global proc-mem low 70 high 80 treatment
call threshold global total-calls low 10 high 12 treatment
!
controller T1 2/0
!
controller T1 2/1
!
controller T1 3/0
 framing esf
 clock source internal
 ds0-group 1 timeslots 1-5 type none service mgcp
 ds0-group 2 timeslots 6-24 type none service mgcp
!
controller T1 3/1
 framing esf
 ds0-group 1 timeslots 1-10 type none service mgcp
 ds0-group 2 timeslots 11-24 type none service mgcp
!
!
!
interface FastEthernet0/0
 ip address 192.168.1.61 255.255.255.0
 duplex auto
 speed auto
! The following command is used in MGCP RSVP CAC to configure
the bandwidth allocated.
! for VoIP calls through the interface.
```

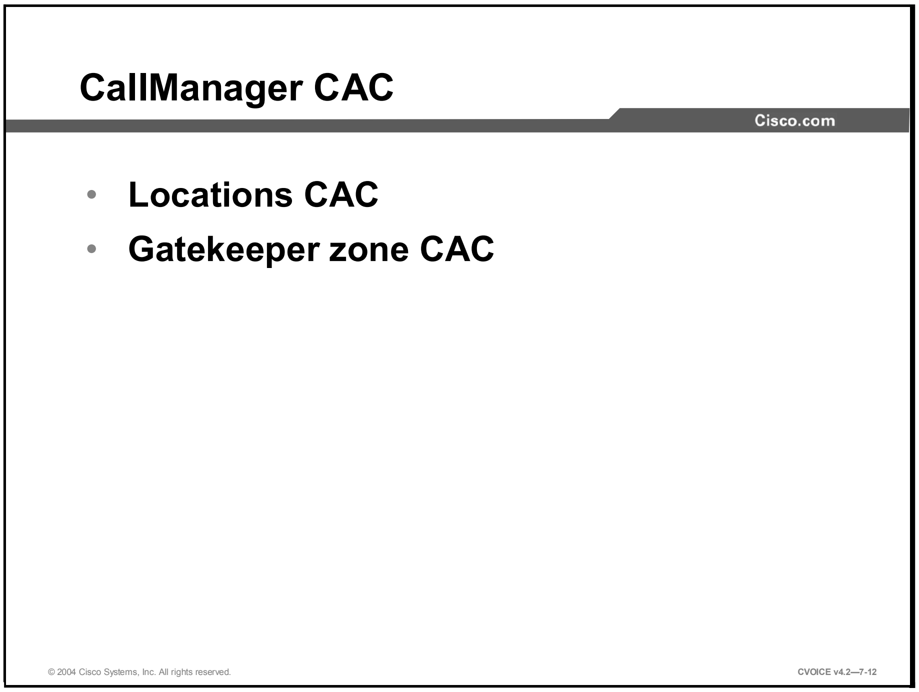
```
 ip rsvp bandwidth 512 512
!
interface FastEthernet0/1
 ip address 172.20.1.1 255.255.0.0
 duplex auto
 speed auto
!
ip kerberos source-interface any
ip classless
ip route 10.0.0.0 10.0.0.0 192.168.1.10
no ip http server
!
snmp-server engineID local 0000000902000002B95D89F0
no snmp-server ifindex persist
snmp-server manager
!
voice-port 3/0:1
!
voice-port 3/0:2
!
voice-port 3/1:1
!
voice-port 3/1:2
!
mgcp
mgcp call-agent 10.13.57.88 service-type mgcp version 1.0
mgcp modem passthrough voip mode nse
mgcp modem passthrough voaal2 mode
mgcp package-capability trunk-package
! The following command is used for MGCP SA Agent CAC.
 mgcp rtrcac
! The following command is used in MGCP SRC CAC.
mgcp src-cac
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
!
dial-peer voice 1 pots
```



```
 application mgcpapp
 port 3/0:1
!
dial-peer voice 2 pots
 application mgcpapp
 port 3/0:2
!
dial-peer voice 3 pots
 application mgcpapp
 port 3/1:1
!
dial-peer voice 4 pots
 application mgcpapp
 port 3/1:2
!
! The following command is used in MGCP SA Agent CAC.
rtr responder.
!
line con 0
 exec-timeout 0 0
 privilege level 15
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Cisco CallManager CAC

This topic describes the types of call admission that are possible with Cisco CallManager CAC.



Using Cisco CallManager, the following two types of call admission are possible:

- **Locations CAC:** The locations feature provides CAC for centralized call-processing systems. A centralized system uses a single Cisco CallManager cluster to control all of the locations. The locations feature in Cisco CallManager allows you to specify the maximum amount of bandwidth available for calls *to* and *from* each location, thereby limiting the number of active calls and preventing oversubscription of the bandwidth on the IP WAN links.
- **Gatekeeper zone CAC:** A gatekeeper device provides CAC for distributed call-processing systems. In a distributed system, each site contains its own call-processing capability. Calls are limited between zones in this configuration.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- CAC is a determining and informed decision that provides QoS.
- If a network link is oversubscribed with too much voice traffic, data packets are dropped and voice calls suffer.
- CAC functions as part of call control services.
- RSVP is a CAC mechanism.
- RSVP can be configured in a Frame Relay network.
- RSVP can be configured with PPP.
- Distinct CAC mechanisms consist of local, measurement-based, and resource-based.

 CVOICE v4.2—7-13

Summary (Cont.)

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- Various CAC tools are available for different protocols and systems.
- Cisco provides for H.323 Call Admission Control.
- Cisco provides for SIP Call Admission Control.
- Cisco provides for MGCP Call Admission Control.
- Cisco CallManager allows for two types of call admission.

 CVOICE v4.2—7-14

References

For additional information, refer to this resource:

- *Resource-Based CAC Mechanisms*
  http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/cac.htm#77341
- *Call Admission Control for H.323 Gateways*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a00800e0d4b.html#76206
- *Call Admission Control for MGCP*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a0080087d37.html#1048015
- *Measurement-Based Call Admission Control for SIP*
  http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a00801541b2.html

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) CAC mechanisms extend the capabilities of QoS tools to protect voice from what? (Choose two.)

- A) the negative effects of excessive data traffic
- B) to keep excessive voice traffic off the network
- C) bursty data traffic
- D) the negative effects of other voice traffic

Q2) What is the function of CAC?

- A) to deny traffic that does not meet overhead requirements
- B) to deny traffic that has not been authenticated
- C) to deny voice traffic if the required network resources are not available
- D) to deny data traffic until higher priority voice traffic has been transmitted

Q3) How is traffic affected if a link is oversubscribed? (Choose two.)

- A) Data packets continue to be sent but at a slower rate.
- B) Data packets are denied entry to the queue.
- C) Voice packets compete for bandwidth with other packets from the priority queue.
- D) Voice packets are sent as priority traffic and dropped packets are retransmitted.

Q4) If CAC and queuing are enabled on a link, which traffic will be transmitted first?

- A) small data packets
- B) large data packets
- C) voice packets
- D) packets that arrived first

Q5) At what location in the network are CAC call control services configured?

- A) the incoming gateway
- B) the outgoing gateway
- C) the incoming gatekeeper
- D) the outgoing gatekeeper
- E) the entire network

Q6) Which statement describes CAC call control?

A) The local node is configured to allow no more than a specified number of calls.

B) The local node routes the calls on links that have enough bandwidth.

C) The network does not allow calls that are not negotiated.

D) The network holds the link open until all priority calls have gone through.

Q7) Which characteristic of RSVP is different from other CAC mechanisms?

A) requires manual configuration at each interface in the path

B) denies a call based on a best guess

C) guarantees QoS against changing network conditions

D) knows how many other calls are active on the link

Q8) Which device makes the CAC decision when RSVP is being used?

A) the originating dial peer

B) the terminating dial peer

C) the originating gateway

D) the terminating gateway

Q9) In a network that is supporting voice, in which areas can CAC be implemented?

A) H.323

B) SGCP

C) gatekeeper

D) SIP

E) H.245

F) H.225

Q10) Each area of CAC is associated with a specific _____ or system.

A) server

B) gateway

C) endpoint

D) protocol

Q11) With the _____ command, you can configure two thresholds, high and low, for each resource.

A) **call spike**

B) **call threshold**

C) **call treatment**

D) **rtr responder**

Q12) With the _____ command, you can select how the call should be treated when local resources are not available to handle the call.

A) **call spike**

B) **call threshold**

C) **call treatment**

D) **rtr responder**

Q13) Which three of the following do Service Assurance Agents use to monitor network performance? (Choose three.)

A) latency

B) codec availability

C) delay

D) jitter

E) bandwidth

Q14) Which CAC tool monitors congestion in the IP network and either redirects calls to the PSTN or rejects calls based on network congestion?

A) PSTN Fallback

B) Service Assurance Agents

C) Resource Availability Check

D) RSVP

Q15) Which three does MGCP VoIP CAC use for improving voice quality and reliability? (Choose three.)

A) SRC

B) RSVP

C) DSCP

D) SA Agent

E) IP precedence

Q16) To set thresholds and enable MGCP SRC CAC, which command is used in global configuration mode?

A) **configure terminal**

B) **call-rsvp sync**

C) **call threshold global *trigger-name* low *value* high *value* treatment**

D) **call global threshold low *value* high *value* *trigger-name* treatment**

Q17) Which feature of Cisco CallManager allows you to specify the maximum bandwidth available for calls to and from each location?

A) the set bandwidth feature

B) the locations feature

C) the gatekeeper zone feature

D) the active calls feature

Q18) Which device provides CAC for distributed call processing using Cisco CallManager?

A) DSP

B) dial peer

C) gatekeeper

D) gateway

Q19) Which two commands are necessary for SAA to monitor network performance with MGCP? (Choose two.)

A) **rtr responder**

B) **mgcp rtrcac**

C) **call fallback active**

D) **mgcp src-cac**

E) **interface-kbps**

F) **single-flow-kbps**

Quiz Answer Key

Q1) D

Relates to: Need for CAC

Q2) C

Relates to: Need for CAC

Q3) B, C

Relates to: Effects of Oversubscribing Bandwidth on Overall Voice Quality

Q4) C

Relates to: Effects of Oversubscribing Bandwidth on Overall Voice Quality

Q5) B

Relates to: CAC as Part of Call Control Services

Q6) A

Relates to: CAC as Part of Call Control Services

Q7) C

Relates to: RSVP

Q8) D

Relates to: RSVP

Q9) A, C, D

Relates to: Understanding CAC Tools

Q10) D

Relates to: Understanding CAC Tools

Q11) B

Relates to: H.323 CAC

Q12) C

Relates to: H.323 CAC

Q13) A, C, D

Relates to: SIP CAC

Q14) A

Relates to: SIP CAC

Q15) A, B, D

Relates to: MGCP CAC

Q16) C

Relates to: MGCP CAC

Q17) B

Relates to: Cisco CallManager CAC

Q18) C

Relates to: Cisco CallManager CAC

Q19) A, B

Relates to: MGCP CAC

Understanding Voice Bandwidth Engineering

Overview

This lesson describes the importance of proper bandwidth engineering. Simply adding voice to an existing IP network is not acceptable. You must take proper precautions to ensure enough bandwidth for existing data applications and the added voice.

Relevance

Network administrators must be able to calculate existing bandwidth and added voice bandwidth to implement VoIP.

Objectives

Upon completing this lesson, you will be able to allocate bandwidth for voice and data traffic. This includes being able to meet these objectives:

- Describe the tools that are used to examine and collect traffic statistics
- Describe the network objectives for voice and data to ensure proper performance of the network
- Describe how to meet the network objectives within the current network
- Calculate the required number of trunks that are necessary to support voice traffic, considering busy hour and dropped calls
- Calculate busy hour traffic
- Describe erlangs as they relate to trunks
- Explain three traffic probability assumptions when determining the number of trunks required to meet a grade of service
- Explain the purpose of traffic calculations
- Describe the creation of a call density matrix for calculating the trunk requirements between two points in a network
- Describe how to calculate the required bandwidth allocation for voice and data traffic

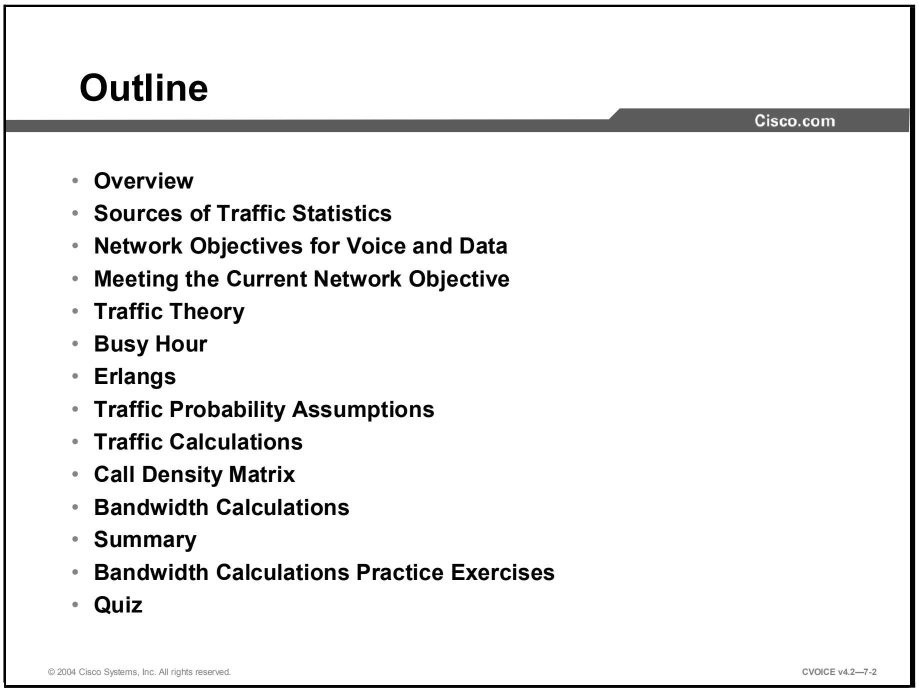
Learner Skills and Knowledge

To benefit fully from this lesson, you must have these prerequisite skills and knowledge:

- Basic knowledge of VoIP
- Knowledge of TCP/IP networks

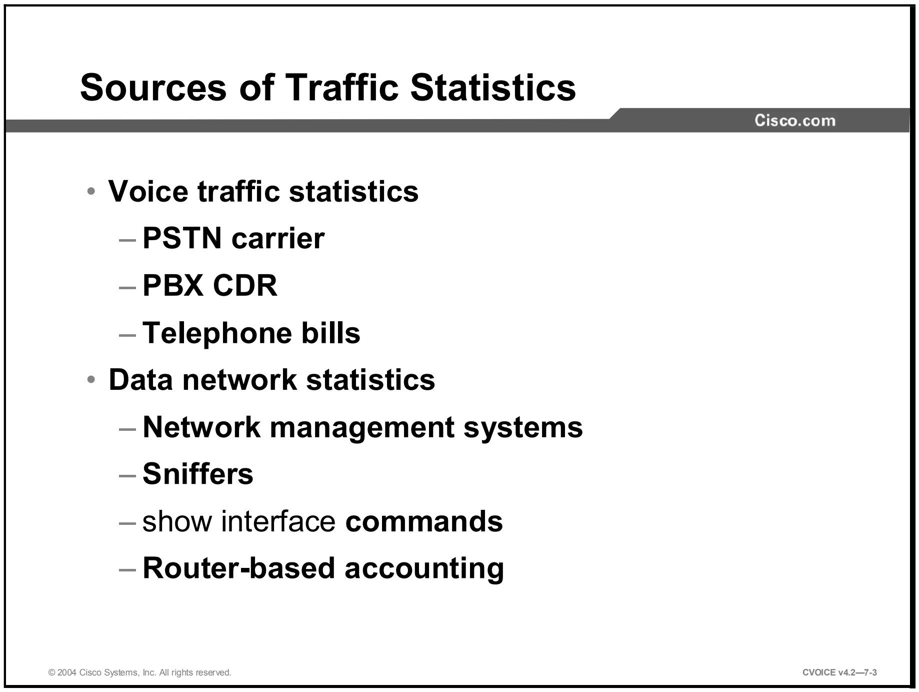
Outline

The outline lists the topics included in this lesson.



Sources of Traffic Statistics

This topic describes the sources of traffic statistics for voice and data networks.



Traffic engineering, as it applies to traditional voice networks, is determining the number of trunks that are necessary to carry a required number of voice calls during a specific time period. For designers of a Voice over *x* network, the goal is to properly size the number of trunks and provision the appropriate amount of bandwidth that is necessary to carry the data equivalent of the number of trunks determined.

To determine the number of trunks, you must have statistics showing the current voice traffic.

Example: Gathering Statistics

You can gather voice traffic statistics from several sources, including:

- PSTN carriers
- Call Detail Records (CDRs) in PBXs
- Telephone bills

From the PSTN carrier, you can often gather the following information:

- Peg counts for calls offered, calls abandoned, and all trunks busy. A peg count is a telephony term that dates back to the days of mechanical switches. A mechanical counter was attached to a peg to measure the number of events on that peg. The peg might be energized (sent a signal) for any one of several reasons, including call overflow and trunk seized. Today, electronic switches record peg counts by way of the software programs in their common control components.
- Total traffic carried per trunk group

In the absence of this detailed information, you could use a telephone bill to approximate the total traffic, but telephone bills do not show you the lost calls or the grade of service.

The internal telecommunications department provides CDRs for PBXs. This information typically records calls that are offered, but may not provide information on calls that were blocked because all trunks were busy.

Ideally, all call statistics are provided on a time-of-day basis. The number of trunks required to carry voice traffic is based on the *peak* daily traffic, not the *average* daily traffic.

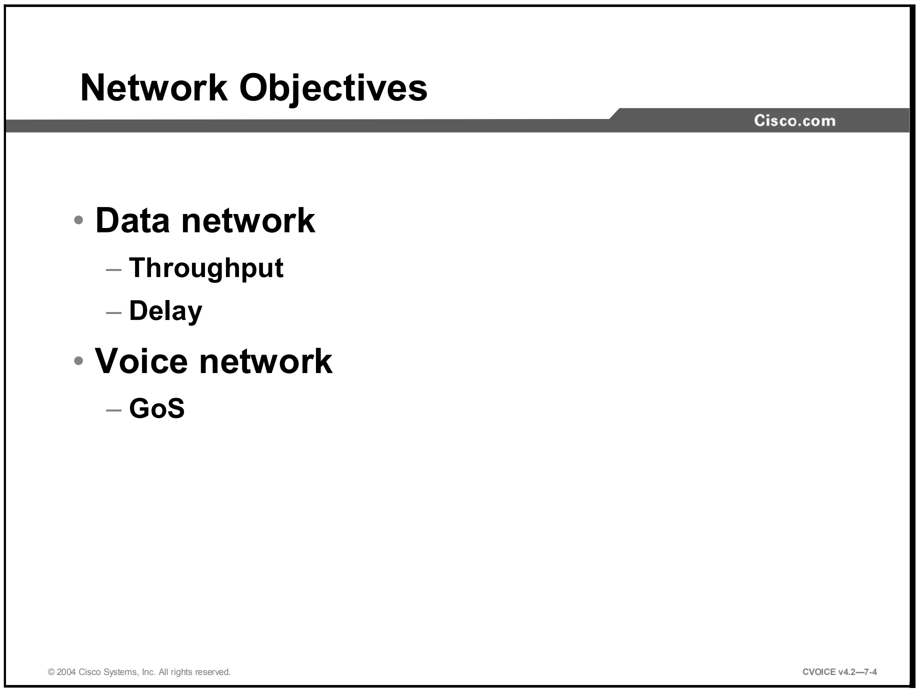
Unless the data network is accommodating voice and data on separate facilities, the resulting data traffic after migrating voice to the data network is the sum of the data traffic and the new voice traffic. Therefore, you must know how much bandwidth is required for data application.

You can gather data traffic statistics from the following:

- Network management systems (NMSs)
- Sniffers
- **show interface** commands
- Router-based accounting

Network Objectives for Voice and Data

This topic discusses the network objectives for voice and data to ensure proper performance of the network.



To provide an acceptable level of access to telephone and data services in a combined voice and data network, you must establish guidelines for the acceptable performance of each.

In a data network, users can reasonably expect to achieve a level of throughput in bits per second (bps) or a network transit delay in milliseconds (ms). Unfortunately, few networks have stated objectives for throughput and delay. In planning a combined voice and data network, voice is sharing the same paths as data. Because of its real-time requirements, voice is given first access to the network resources. Consequently, service to the data users can be affected. Without a target for throughput and delay, you are unable to judge the suitability of the combined voice and data network.

Traffic engineering for voice is based on a target Grade of Service (GoS). GoS is a unit that measures the chance that a call is blocked. The GoS is usually defined for the peak or busiest period in the business day, thereby representing service at its worst. During off-busy times, access to the service is better.

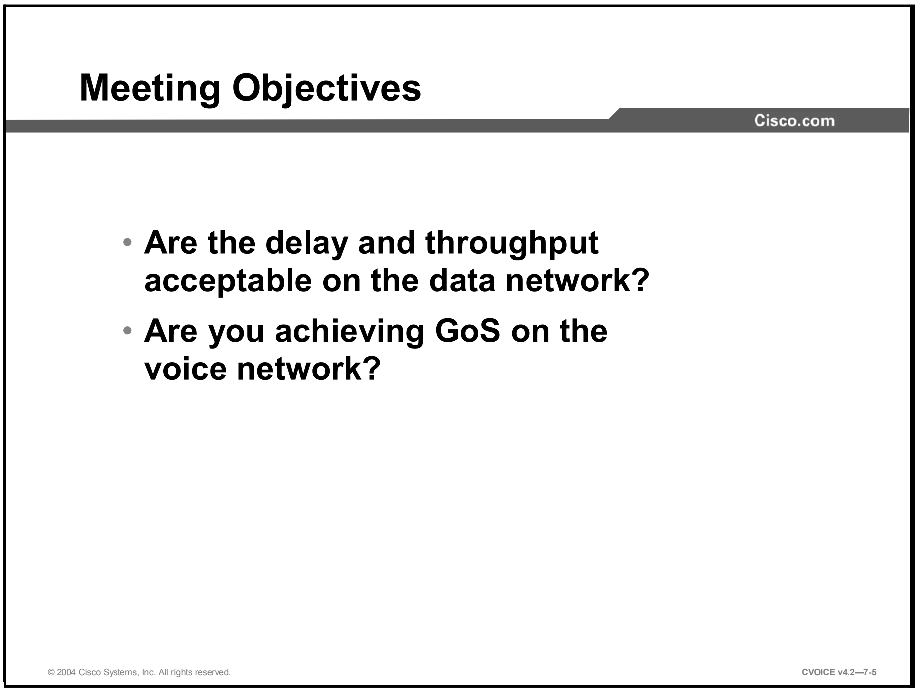
The GoS is an important parameter when calculating the number of trunks required to carry an amount of voice traffic.

Example: GoS Value

For example, a GoS of P .01 means that one call is blocked in 100 call attempts, and a GoS of P .001 results in one blocked call per 1000 attempts.

Meeting the Current Network Objective

This topic describes meeting the network objectives within the current network.



When the current level of voice and data traffic in the networks has been determined and objectives for throughput, delay (data traffic), and GoS (voice traffic) have been set, you should consider whether you are meeting the objectives in the current networks.

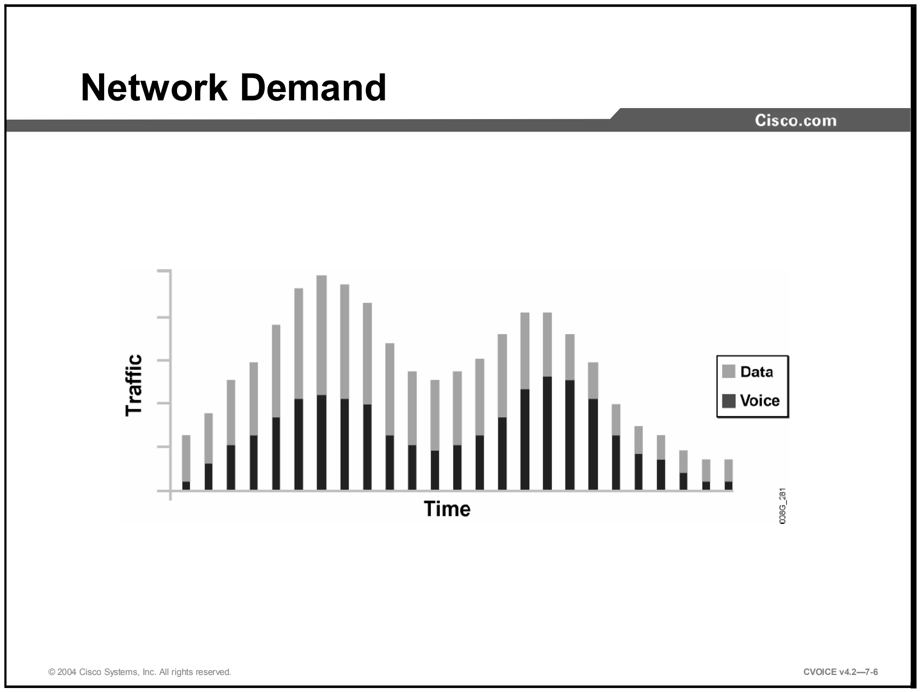
You might discover through your analysis of the voice and data networks that you are providing a poor GoS to your voice users, or the throughput and delay are below the standards for your data users. Without recognizing these shortcomings, you might be inclined to plan a combined network on the assumption of business as usual, which is a mistake. Even worse, you could create an integrated voice and data network that behaves even more poorly.

You must ask two questions to determine if you are meeting current network objectives:

1. Are the delay and throughput acceptable on the data network?

2. Are you achieving GoS on the voice network?

If you conclude that your network performance is below objectives, add a factor to your current traffic analysis for excessive demand.



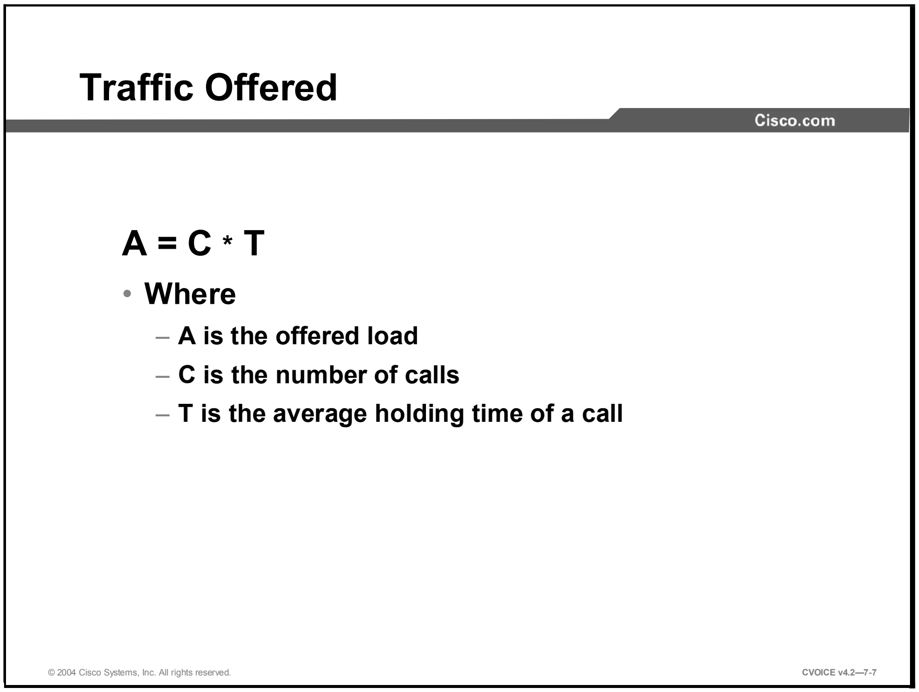
You must understand how voice and data network demands relate to each other. If available, you should look at the relationship between the peak demands on each of the networks. This gives you some idea of what to expect later in this process, when you convert the number of voice trunks to bandwidth and add the voice bandwidth requirement to the data bandwidth for the same period. Clearly, the peak bandwidth demand is less if the two network demands are out of phase with each other than if both peaks coincide.

Example: Network Demand

Usually, networks will exhibit peak demands early in the morning and just after the noon hour. To best calculate the required bandwidth necessary to support demands, peak usage times must be understood.

Traffic Theory

This topic describes how to calculate the number of trunks necessary to support voice traffic.



If you know the amount of traffic generated and the GoS required, then you can calculate the number of trunks required to meet your needs. Use the following simple equation to calculate traffic flow:

$A = C * T$

In the equation, A is the offered traffic, C is the number of calls originating during a period of one hour, and T is the average holding time of a call.

It is important to note that C is the number of calls originated, not carried. Typically, the information received from the carrier or from the internal CDRs of the company is in terms of *carried* traffic, not *offered* traffic, as is usually provided by PBXs.

The holding time of a call (T) must account for the average time a trunk is occupied and must factor in variables other than the length of a conversation. This includes the time required for dialing and ringing (call establishment), time to terminate the call, and a method of amortizing busy signals and noncompleted calls. Adding 10 to 16 percent to the length of an average call helps account for these miscellaneous time segments.

Hold times based on call billing records might have to be adjusted, based on the increment of billing. Billing records based on one-minute increments are usually rounded up to the *next* minute, not rounded to the *nearest* minute. Consequently, billing records overstate calls by 30 seconds on average; for example, a bill showing 404 calls (C) totaling 1834 minutes of traffic (A) should be adjusted as follows:

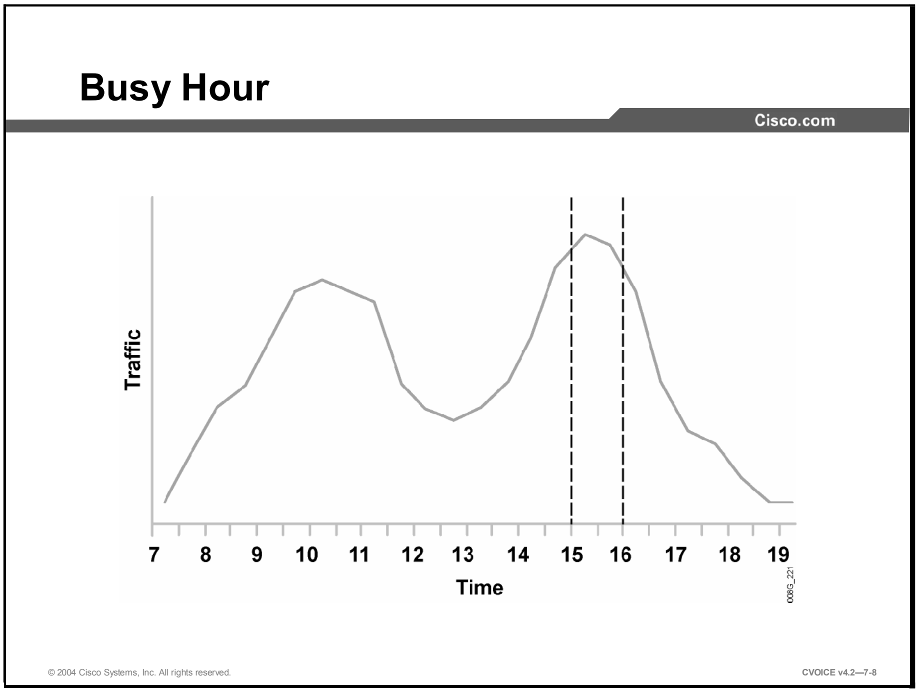
- 404 calls * 0.5 minutes (overstated call length) = 202 excess call minutes
- Adjusted traffic (A): 1834 – 202 = 1632 actual call minutes

Another way to calculate this would be to use the formula A = C * T to derive the average holding time (T), reduce T by 0.5 minutes (the overstated amount), and recalculate the traffic offered (A).

- **Average holding time (T):**
  - T = 1834 minutes (A) / 404 calls (C)
  - T = 4.54 minutes
- **Corrected holding time (T):**
  - T = 4.54 – 0.50
  - T = 4.04 minutes
- **Adjusted traffic offered (A):**
  - A = 404 calls (C) * 4.04 minutes (T)
  - A = 1632 call minutes

Busy Hour

This topic explains how to calculate busy hour during any given day.

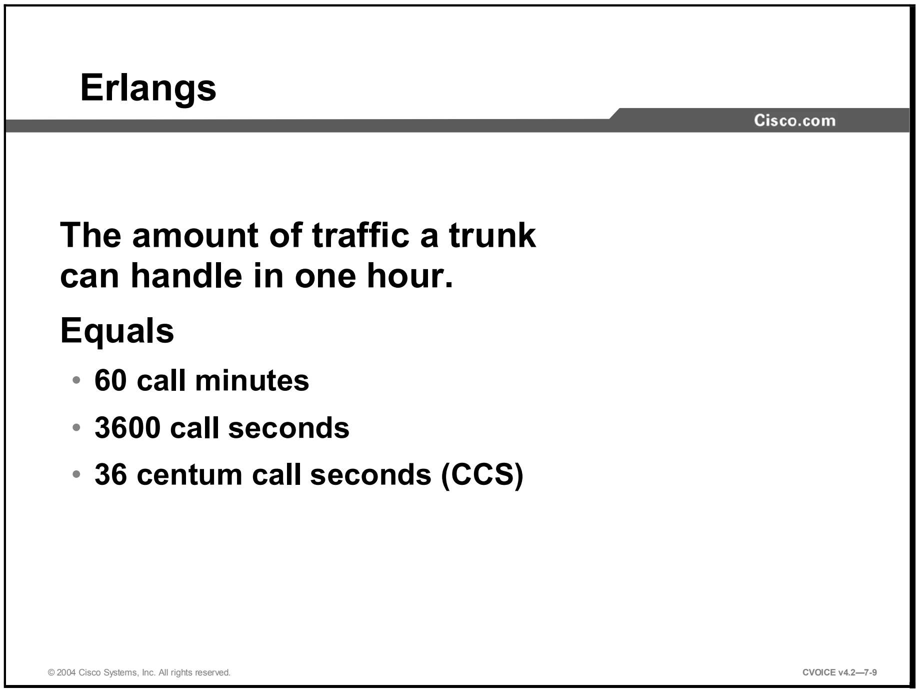


It is important to look at call attempts during the busiest hour of a day. The most accurate method of finding the busiest hour is to take the 10 busiest days in a year, sum the traffic on an hourly basis, find the busiest hour, and then derive the average amount of time.

In the absence of a traffic profile from which you can derive a precise value of the busy hour traffic, a simple way to calculate the busy hour is to collect one business month of traffic. Determine the amount of traffic that occurs in a day based on 22 business days in a month and then multiply that number by 15 to 17 percent. As a rule, the busy hour traffic represents 15 to 17 percent of the total traffic that occurs in one day. So, for example, if you have a trunk group that carries 66,000 minutes in one month or 3000 minutes per day on average (66,000/22), you could estimate the busy hour traffic by calculating 15 percent of the average daily traffic, or 3000 * 15% = 450 minutes.

Erlangs

This topic describes erlangs.



The traffic volume in telephone engineering is measured in units called erlangs. An erlang is the amount of traffic one trunk can handle in one hour. It is a nondimensional unit that has many functions.

Other equivalent measurements that you might encounter include the following:

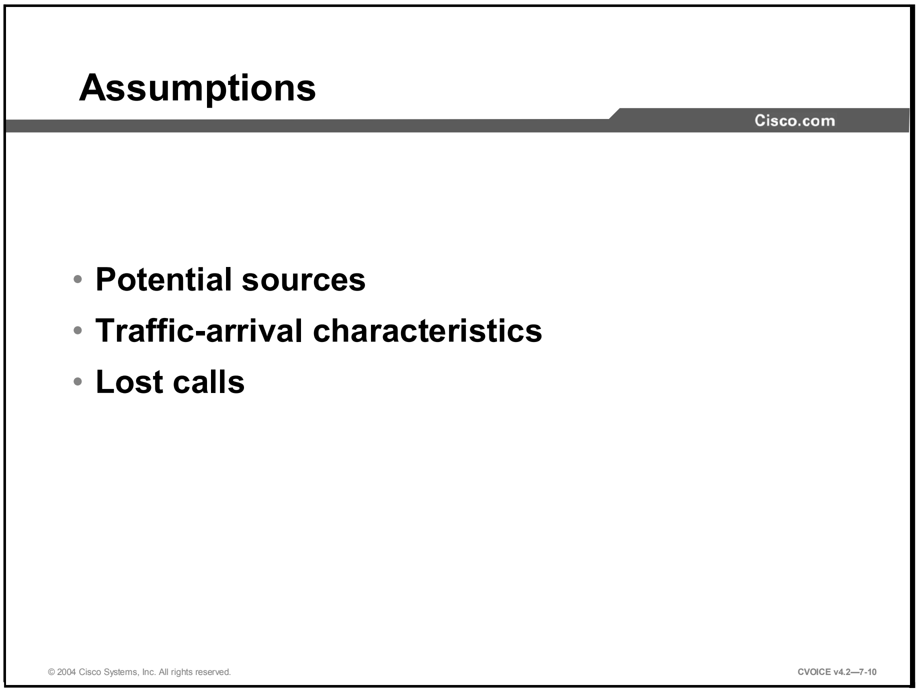
1 erlang = 60 call minutes = 3600 call seconds = 36 centum call seconds (CCS)

Example: Erlang Calculation

Assume that, on average, each user in a branch makes 10 calls with an average duration of 5 minutes during the busy hour. If the branch has 25 employees, the total number of minutes of call time during the busy hour is 10 calls per busy hour multiplied by 5 minutes per call multiplied by 25 employees per branch, or 1250 call minutes. However, erlangs is a measurement based on hours, so the number of erlangs is 1250 divided by 60, or 20.83 erlangs.

Traffic Probability Assumptions

This topic explains three traffic probability assumptions to consider when determining the number of trunks required to meet a GoS.



When you have determined—in erlangs—the amount of traffic that occurs during the busy hour, you can determine the number of trunks required to meet a particular GoS. The number of trunks required differs, depending on the following traffic probability assumptions:

- **Assumption 1—Number of potential sources:** There can be a major difference between planning for an infinite versus a small number of sources. As the number of sources increases, the probability of a wider distribution in the arrival times and holding times of calls increases. As the number of sources decreases, the ability to carry traffic increases.
- **Assumption 2—Traffic arrival characteristics:** Usually, this assumption is based on a Poisson traffic distribution (named after the mathematician who studied this extensively) in which call arrivals follow a classic bell-shaped curve. You commonly use Poisson distribution for infinite traffic sources. The arrival characteristics of traffic (calls) may be classified as random, smooth, or bursty.
- **Assumption 3—Treatment of lost calls:** What do you do when the station you are calling does not answer or all trunks are busy? Traffic theory considers three possibilities:
  - **Lost Calls Cleared (LCC):** LCC assumes that once a call is placed and the server (network) is busy or not available, the call disappears from the system. In essence, you give up and do something different.
  - **Lost Calls Held (LCH):** LCH assumes that a call is in the system for the duration of the hold time, regardless of whether the call is placed. In essence, you continue to redial for as long as the hold time before giving up. Recalling, or redialing, is an important traffic consideration. Suppose 200 calls are attempted. Forty receive busy signals and attempt to redial. That results in 240 call attempts, a 20 percent increase. The trunk group is now providing an even poorer GoS than initially thought.

— **Lost Calls Delayed (LCD):** LCD means that when a call is placed, it remains in a queue until a server is ready to handle the call. Then it uses the server for the full holding time. This assumption is most commonly used for automatic call distribution (ACD) systems.

LCC tends to understate the number of trunks that are required; on the other hand, LCH overstates the number of trunks that are required.

Example: Traffic Arrival Assumption

Random arrivals are common with a large (tending to infinite) source of users whose calls are independent of each other. Assuming random arrivals, the probability of calls arriving during any particular time interval (such as the busy hour) is modeled by a Poisson distribution. Given the average number of calls per busy hour and the distribution of call-holding times, the Poisson distribution predicts the probability of zero calls, one call, two calls, and so on, up to the probability of a large number of calls arriving during the hour.

The characteristic bell shape of the distribution suggests that the probability of few calls is low, as is the probability of a high number of calls. The peak probability represents the average; for example, if you have calculated the average number of calls during the busy hour to be 250, the Poisson distribution estimates the probability that you will receive 0 calls (very low probability), 100 calls (modest probability), 250 calls (the average, so the peak probability), 900 calls (very low probability), or any other number of calls.

Smooth traffic arrivals are common in applications in which the traffic is dependent on other traffic, as in a telemarketing scenario. Smooth traffic arrivals are not modeled on the Poisson distribution due to their nonrandom nature.

Bursty traffic arrivals are common in trunk overflow scenarios in which excess overflow tends to occur for a short time and then disappears for an extended period of time. Bursty traffic is not modeled on the Poisson distribution because of its nonrandom nature.

Traffic Calculations

This topic explains the purpose of traffic calculations.

Traffic Calculations

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Trunks	Probability of Lost Call					
	0.003	0.005	0.01	0.02	0.03	0.05
1	0.003	0.005	0.011	0.021	0.031	0.053
2	0.081	0.106	0.153	0.224	0.282	0.382
3	0.289	0.349	0.456	0.603	0.716	0.9
4	0.602	0.702	0.87	1.093	1.259	1.525
5	0.995	1.132	1.361	1.658	1.876	2.219
6	1.447	1.622	1.909	2.276	2.543	2.961
7	1.947	2.158	2.501	2.936	3.25	3.738
8	2.484	2.73	3.128	3.627	3.987	4.543
9	3.053	3.333	3.783	4.345	4.748	5.371
10	3.648	3.961	4.462	5.084	5.53	6.216
11	4.267	4.611	5.16	5.842	6.328	7.077
12	4.904	5.279	5.876	6.615	7.141	7.95
13	5.559	5.964	6.608	7.402	7.967	8.835
14	6.229	6.664	7.352	8.201	8.804	9.73
15	6.913	7.376	8.108	9.01	9.65	10.63

 CVOICE v4.2—7-11

The purpose of traffic calculations is to determine the number of physical trunks that are required. After you have determined the amount of offered traffic during the busy hour, established the target GoS, and recognized the three basic assumptions, you can calculate the number of trunks that are required by using formulas or tables.

Traffic theory consists of many queuing methods and associated formulas. Anyone who has taken a queuing theory class can testify to the complexity of the many queuing models that are derived for various situations. As such, tables dealing with the most commonly encountered model are used.

The most commonly used model and table is Erlang B. Erlang B is based on infinite sources, LCC, and Poisson distribution that is appropriate for either exponential or constant holding times. In general, Erlang B understates the number of trunks because of the LCC assumption, but it is generally the most commonly used algorithm.

Example: Traffic Calculations

A trunk group is a hunt group of parallel trunks. The following example determines the number of trunks in a trunk group carrying the following traffic:

- 352 hours of offered call traffic in a month
- 22 business days per month
- 10 percent call-processing overhead
- 15 percent of the traffic occurs in the busy hour
- GoS = P .01
- Busy hour = 352 / 22 * 15% * 1.10 (call-processing overhead) = 2.64 erlangs

The traffic assumptions are:

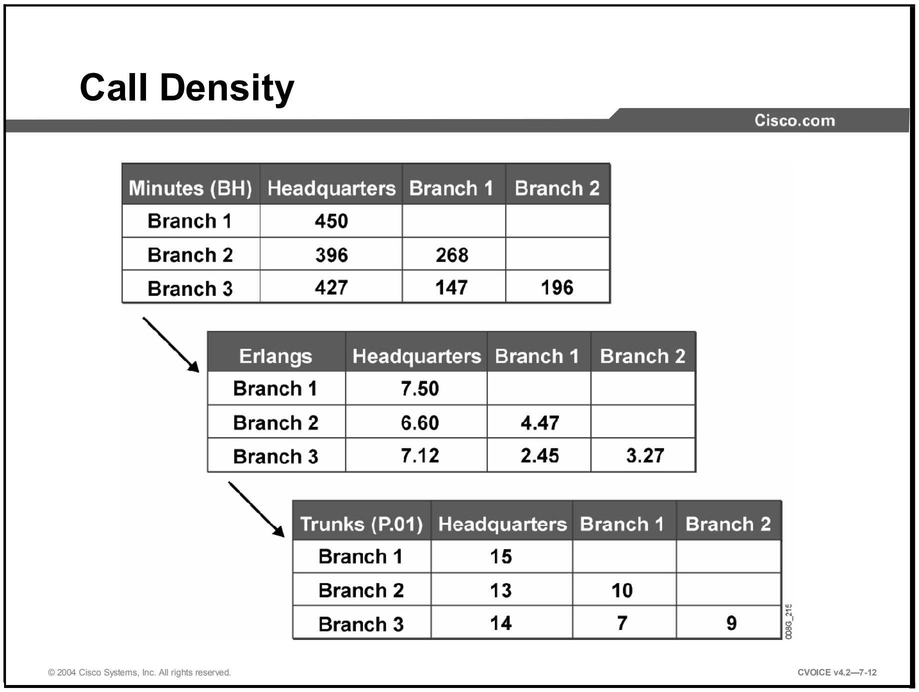
- Infinite sources
- Random or Poisson traffic distribution
- Lost calls are cleared

Based on these assumptions, the appropriate algorithm to use is Erlang B. You use the table in the preceding figure to determine the appropriate number of trunks for a GoS of P .01.

Because a GoS of P .01 is required, you use the column that is designated as P .01 only. The calculations indicate a busy hour traffic amount of 2.64 erlangs, which is between 2.501 and 3.128 in the P .01 column. This corresponds to between seven and eight trunks. Because you cannot use a fractional trunk, use the next larger value of eight trunks to carry the traffic.

Call Density Matrix

This topic describes a call density matrix.



Minutes (BH)	Headquarters	Branch 1	Branch 2
Branch 1	450		
Branch 2	396	268	
Branch 3	427	147	196

Erlangs	Headquarters	Branch 1	Branch 2
Branch 1	7.50		
Branch 2	6.60	4.47	
Branch 3	7.12	2.45	3.27

Trunks (P.01)	Headquarters	Branch 1	Branch 2
Branch 1	15		
Branch 2	13	10	
Branch 3	14	7	9

Unless your voice network is extremely small (two points, for example), calculating the trunk requirements between any two points in a network is a tedious task. One way to manage this more effectively is to design a *to/from* traffic matrix.

As you determine the call minutes between any two points, you enter the amount into the matrix. If you do this on a spreadsheet, you can convert the minutes to busy hour minutes, to erlangs, and then calculate the number of trunks from the erlangs. The number of trunks represents the number of concurrent calls that you should plan to support during the busy hour.

Example: Call Density Matrix

The figure shows this progression of spreadsheets for a network with one headquarters and three branches.

Bandwidth Calculations

This topic describes how to calculate the required bandwidth allocation for voice and data traffic.

VoIP Bandwidth

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Codec	Codec Speed	Sample Size	Frame Relay	Frame Relay with CRTP	Ethernet	Ethernet with CRTP
G.711	64000	240	76267	66133	78933	68800
G.711	64000	160	82400	67200	86400	71200
G.726r32	32000	120	44267	34133	46933	36800
G.726r32	32000	80	50400	35200	54400	39200
G.726r24	24000	80	37800	26400	40800	29400
G.726r24	24000	60	42400	27200	46400	31200
G.726r16	16000	80	25200	17600	27200	19600
G.726r16	16000	40	34400	19200	38400	23200
G.728	16000	80	25200	17600	27200	19600
G.728	16000	40	34400	19200	38400	23200
G.729	8000	40	17200	9600	19200	11600
G.729	8000	20	26400	11200	30400	15200
G.723r63	6300	48	12338	7350	13650	8663
G.723r63	6300	24	18375	8400	21000	11025
G.723r53	5300	40	11395	6360	12720	7685
G.723r53	5300	20	17490	7420	20140	10070

CVOICE v4.2—7-13

If you are provisioning a circuit-switched voice network, you must calculate how much traffic each of your trunks is expected to carry (according to theory), and investigate the most cost-effective way to provision each of the trunks.

Although you might still choose to perform the calculations used for VoIP, for the purposes of this example, assume that all voice traffic is transported over the IP network. Consequently, you do not need to go through the rigors of choosing the most cost-effective solution for each individual trunk.

Determining IP Bandwidth

At this stage, the goal is to determine the IP bandwidth:

1. When you estimate the VoIP bandwidth required for different codecs and over different data links, you can also calculate the benefit of CRTP. These estimates of bandwidth for VoIP are shown in the figure.

2. When you identify the number of concurrent calls that you expect during the busy hour between points in the network, as a conservative (worst-case) first approximation to the bandwidth required for voice, you can simply multiply the number of concurrent calls by the bandwidth per call.

Bandwidth

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Voice Bandwidth (kbps)	Headquarters	Branch 1	Branch 2
Branch 1	168		
Branch 2	145.6	112	
Branch 3	156.8	78.4	100.8

+

Data Bandwidth (kbps)	Headquarters	Branch 1	Branch 2
Branch 1	248		
Branch 2	216	63	
Branch 3	187	28	46

=

Total Bandwidth (kbps)	Headquarters	Branch 1	Branch 2
Branch 1	416		
Branch 2	361.6	175	
Branch 3	343.8	106.4	146.8

 CVOICE v4.2—7-14

The spreadsheet shows the results for VoIP over a Frame Relay network using G.729 with CRTP. Each call requires 11.2 kbps.

You may want to consider some refinements to this simple multiplier by considering the net benefits of bandwidth-reduction strategies, such as voice activity detection (VAD). VAD commonly reduces the bandwidth to between 60 and 70 percent of the original bandwidth. However, two animated talkers may not allow VAD to have any effect at all.

Finally, remember to add the budget for data applications to the bandwidth budget for voice.

Now that you have the total bandwidth budget, you must ensure that you do not overload the data links. Although it might seem most cost-effective to match the access rate on an interface to the total bandwidth budget, this is not correct.

On any network, dropped traffic and delays are proportional to the load on the network. As load approaches middle percentages, drops and delay exponentially increase. When designing networks to carry voice and data, peak bandwidth calculations should *not* equate to total bandwidth required for a given network link. For most business environments, you can use any of the following load levels as general rules for determining when a network is approaching excessive load:

- 20 percent of full capacity averaged over an 8-hour work day
- 30 percent averaged over the worst hour of the day
- 50 percent averaged over the worst 15 minutes of the day

Capacity planning should take these factors into account, and link speed should be chosen to accommodate proper load factors. An ideal goal is to have demand equal to about 35 percent of total link speed.

To put these rules into the context of the example, the spreadsheet shows a demand for 416 kbps between headquarters and branch 1 during the busy (worst) hour of the day. Using the rules, the average demand during the worst hour should represent only 30 percent of the link speed. If 30 percent of the link speed is 416 kbps, then the link speed must be 1387 kbps or greater.

Based on the traffic in your network, you may be justified in aiming for a higher utilization of the links, but expecting full utilization is unrealistic. Setting too high a value results in low throughput and high delay for data traffic. Voice is prioritized so that it is not delayed. You need to determine, through experience, a utilization factor that balances throughput and delay with cost.

Summary

This topic summarizes the key points discussed in this lesson.

Summary

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- **Voice traffic statistics and data network statistics are among the sources of traffic statistics.**
- **The network objective for voice is measured in GoS.**
- **To meet the current network objective, delay and throughput must be acceptable on the data network and GoS must be achieved on the voice network.**
- **The equation A = C * T is used to calculate traffic flow.**
- **Call volume should be measured during the busiest hour.**
- **Erlangs help determine trunk requirements.**

CVOICE v4.2—7-15

Summary (Cont.)

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- **Three traffic probability assumptions to consider when determining the number of required trunks involve potential sources, traffic arrival characteristics, and lost calls.**
- **Traffic calculations can be used to determine the number of physical trunks required to meet GoS.**
- **A call density matrix helps determine the number of trunks.**
- **Bandwidth required does not equal link speed.**

CVOICE v4.2—7-16

References

For additional information, refer to these resources:

- *Cisco Voice Design and Implementation Guide*
- *Cisco Traffic Analysis for Voice over IP*

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) How many erlangs would be required with these assumptions:

- 733 hours of offered call traffic in a month
- 22 business days per month
- 5 percent call-processing overhead
- 10 percent of the traffic occurs in the busy hour
- GoS = P .02

A) 34.98

B) 3.498

C) .3498

D) 4.389

Q2) How many trunks would be required with these assumptions:

- 450 hours of offered call traffic in a month
- 22 business days per month
- 10 percent call-processing overhead
- 9 percent of the traffic occurs in the busy hour
- GoS = P .01

A) 4

B) 5

C) 6

D) 7

Q3) How many trunks would be required with these assumptions:

- 400 hours of offered call traffic in a month
- 28 business days per month
- 7 percent call-processing overhead
- 2 percent of the traffic occurs in the busy hour
- GoS = P .01

A) 3

B) 4

C) 5

D) 6

Quiz Answer Key

Q1) B

Relates to: Traffic Calculations

Q2) D

Relates to: Traffic Calculations

Q3) A

Relates to: Traffic Calculations

Quiz

Use the practice items here to review what you learned in this lesson. The correct answers are found in the Quiz Answer Key.

Q1) What are two goals of traffic engineering? (Choose two.)

A) reduce the number of erlangs supported

B) properly size the number of trunks

C) provision the appropriate amount of bandwidth

D) double the peak demand

E) deny calls that exceed acceptable bandwidth

Q2) Which source of traffic statistics can provide you with the most information?

A) CDRs in PBXs

B) PSTN carriers

C) telephone bills

D) NMSs

Q3) Which parameter is important for calculating the number of trunks required to carry voice traffic?

A) throughput

B) delay

C) GoS

D) packet loss

Q4) Which type of statistic is measured by the GoS?

A) the delay variation of a call

B) the chance of a call being denied

C) the voice quality of a call

D) the chance of a call being blocked

Q5) Which condition means that more bandwidth is required?

A) The data network demand is less than the voice network demand.

B) The voice network demand is less than the data network demand.

C) The peak bandwidth demands of the voice and data networks coincide.

D) The peak bandwidth demands of the voice and data networks are out of phase.

Q6) Which two factors determine whether data network requirements are being met? (Choose two.)

A) acceptable GoS

B) acceptable QoS

C) acceptable delay

D) acceptable packet loss

E) acceptable throughput

Q7) In the equation A = C * T used to calculate traffic flow, what does the C represent?

A) number of calls originating during a one-hour period

B) number of calls carried during a one-hour period

C) average holding time for a call

D) length of an average call

Q8) How much time must be added to the length of the actual call to get the holding time?

A) half the length of the call

B) double the length of the call

C) 10 to 16 percent the length of the call

D) 10 to 16 times the length of the call

Q9) What will be the busy hour traffic if the traffic for one month is 15,400 minutes?

A) 77 minutes

B) 105 minutes

C) 116 minutes

D) 119 minutes

Q10) The most accurate method of finding the busiest hour is to take _____ in a year, sum the traffic on an hourly basis, find the busiest hour, and derive the average amount of time.

A) the total number of days

B) the 10 busiest days

C) 22 business days

D) the 12 busiest days

Q11) What is the equivalent of one erlang?

A) 60 call seconds

B) 36 centum call seconds

C) 3600 centum call seconds

D) 1 minute

Q12) What is the traffic volume in erlangs for a company with 30 employees who make an average of five 10-minute calls during the busy hour?

A) 25 erlangs

B) 150 erlangs

C) 300 erlangs

D) 1500 erlangs

Q13) Which three types of arrival traffic can be modeled on a Poisson distribution? (Choose three.)

A) random arrivals

B) arrivals from a small number of sources

C) bursty arrivals

D) arrivals from an infinite number of sources

E) smooth arrivals

Q14) Regarding treatment of lost calls, which possibility overstates the number of trunks required?

A) LCC

B) LCD

C) LCH

D) LCL

Q15) What is the most commonly used table for calculating the number of physical trunks required?

A) Poisson distribution

B) Erlang B

C) GoS algorithm

D) LCC table

Q16) What are three traffic assumptions for the Erlang B algorithm? (Choose three.)

A) fixed number of sources

B) random distribution

C) lost calls delayed

D) lost calls cleared

E) infinite sources

F) lost calls held

Q17) What is an easy way to calculate the trunk requirements for a network?

A) Calculate the requirements between any two points and average them.

B) Calculate the requirements separately for each link.

C) Design a to/from traffic matrix.

D) Use the A = C * T formula.

Q18) The call density matrix can be used to calculate the _____.

A) number of concurrent calls you should plan to support during the busy hour

B) number of calls you should plan to support during the day

C) total bandwidth of the calls that are on the network at any given time

D) number of calls that can be made without affecting voice quality of other calls

Q19) What happens to drops and delays as traffic load approaches middle percentages?

A) They increase for all traffic.

B) They decrease for all traffic.

C) They increase for data traffic only.

D) They decrease for voice traffic only.

Q20) When doing capacity planning, what is the ideal goal for demand?

A) average demand of 20 percent

B) average demand of 25 percent

C) average demand of 30 percent

D) average demand of 35 percent

Quiz Answer Key

Q1) B, C

Relates to: Sources of Traffic Statistics

Q2) B

Relates to: Sources of Traffic Statistics

Q3) C

Relates to: Network Objectives for Voice and Data

Q4) D

Relates to: Network Objectives for Voice and Data

Q5) C

Relates to: Meeting the Current Network Objective

Q6) C, E

Relates to: Meeting the Current Network Objective

Q7) A

Relates to: Traffic Theory

Q8) C

Relates to: Traffic Theory

Q9) B

Relates to: Busy Hour

Q10) B

Relates to: Busy Hour

Q11) B

Relates to: Erlangs

Q12) A

Relates to: Erlangs

Q13) A, C, E

Relates to: Traffic Probability Assumptions

Q14) C

Relates to: Traffic Probability Assumptions

Q15) B

Relates to: Traffic Calculations

Q16) B, D, E

Relates to: Traffic Calculations

Q17) C

Relates to: Call Density Matrix

Q18) A

Relates to: Call Density Matrix

Q19) A

Relates to: Bandwidth Calculations

Q20) D

Relates to: Bandwidth Calculations

Course Glossary

The Course Glossary for *Cisco Voice over IP* (CVOICE) v4.2 highlights and defines key terms and acronyms used throughout this course. Many of these terms are also described in the Cisco Internetworking Terms and Acronyms resource, available via http://www.cisco.com.

Acronym or Term	Definition
3DES	Triple Data Encryption Standard. A stronger form of the Data Encryption Standard (DES), 3DES follows a pattern of encryption/decryption/encryption. 3DES has many different variations.
AAL1	ATM adaptation layer 1. One of four AALs recommended by the ITU-T. AAL1 is used for connection-oriented, delay-sensitive services requiring constant bit rates, such as uncompressed video and other isochronous traffic.
ABR	available bit rate. QoS class defined by the ATM Forum for ATM networks. ABR is used for connections that do not require timing relationships between source and destination. ABR provides no guarantees in terms of cell loss or delay, providing only best-effort service. Traffic sources adjust their transmission rate in response to information they receive describing the status of the network and its capability to successfully deliver data.
access rate	*See* AR.
adaptive differential pulse code modulation	*See* ADPCM.
adaptive predictive coding	*See* APC.
Ad-Hoc conference	A conference call feature where a conference is started by an initiator and only the initiator of the conference can add people into the conference.
admission request	*See* ARQ.
ADPCM	adaptive differential pulse code modulation. A waveform process by which analog voice samples are encoded into digital signals.
advanced integration module	*See* AIM.
Advanced Research Projects Agency	*See* ARPA.
AF	Assured Forwarding. A means of providing different levels of forwarding assurances for IP packets. This method is used by providers who offer differentiated services to their customers.
AIM	advanced integration module. A module in some Cisco routers that provides enhanced processing capabilities to the routers.
alternate mark inversion	*See* AMI.
American National Standards Institute	*See* ANSI.
AMI	alternate mark inversion. Line-code modulation type used on T1 and E1 circuits. In AMI, marks (or ones) cause a pulse in alternating positive and negative directions, while zeros never pulse. Two pulses of the same polarity are not allowed. AMI requires that the sending device maintain ones density. Ones density is not maintained independently of the data stream. Sometimes called *binary coded alternate mark inversion*.
ANI	automatic number identification. SS7 feature in which a series of digits, either analog or digital, are included in the call, identifying the telephone number of the calling device. In other words, ANI identifies the number of the calling party. *See also* CLID.
ANSI	American National Standards Institute. A voluntary organization composed of corporate, government, and other members that coordinates standards-related activities, approves U.S. national standards, and develops positions for the United States in international standards organizations. ANSI helps develop international and U.S. standards relating to, among other things, communications and networking. ANSI is a member of the International Electrotechnical Commission (IEC) and the International Organization for Standardization (ISO).

Acronym or Term	Definition
APC	adaptive predictive coding. A narrowband analog-to-digital conversion technique employing a one-level or multilevel sampling system in which the value of the signal at each sample time is adaptively predicted to be a linear function of the past values of the quantized signals. APC is related to LPC in that both use adaptive predictors. However, APC uses fewer prediction coefficients, thus requiring a higher bit-rate than LPC.
API	application programming interface. The means by which an application program talks to communications software. Standardized APIs allow application programs to be developed independently of the underlying method of communication. A set of standard software interrupts, calls, and data formats that computer application programs use to initiate contact with other devices (for example, network services, mainframe communications programs, or other program-to-program communications). Typically, APIs make it easier for software developers to create the links that an application needs to communicate with the operating system or with the network.
application programming interface	*See* API.
AR	access rate. (1) The maximum data rate of the access channel, typically referring to access to broadband networks and network services. (2) A Frame Relay term that addresses the maximum transmission rate supported by the access link into the network, and the port speed of the device (switch or router) at the edge of the carrier network. The AR defines the maximum rate for data transmission or receipt. *See also* CIR.
ARPA	Advanced Research Projects Agency. Research and development organization that is part of Department of Defense (DoD). ARPA is responsible for numerous technological advances in communications and networking. ARPA evolved into Defense Advanced Research Projects Agency (DARPA), and then back into ARPA again (in 1994).
ARQ	admission request. An RAS admission message defined as an attempt by an endpoint to initiate a call.
AS5300	A series of Cisco gateways that provide reliable, scalable, and feature-rich data and voice gateway functionality. The Cisco AS5300 Series Universal Gateways include the Cisco AS5300 Access Server/Voice Gateway and the Cisco AS5350 Universal Gateway.
Assured Forwarding	*See* AF.
Asynchronous Transfer Mode	*See* ATM.
ATM	Asynchronous Transfer Mode. The international standard for cell relay in which multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells. Fixed-length cells allow cell processing to occur in hardware, thereby reducing transit delays. ATM is designed to take advantage of high-speed transmission media, such as E3, SONET, and T3.
ATM adaptation layer 1	*See* AAL1.
automatic number identification	*See* ANI.
available bit rate	*See* ABR.
B8ZS	binary 8-zero substitution. Line-code modulation type used on T1 circuits. In B8ZS, marks (or ones) cause a pulse in alternating positive and negative directions, while zeros never pulse. Two pulses of the same polarity are not allowed, except when inserting a code to represent eight zeros. B8ZS maintains ones density by inserting a special code in place of eight consecutive zeros. The special code contains intentional violations of the bipolar pattern.
bandwidth change request	*See* BRQ.
basic call	*See* BC.
BC	basic call. A call between two users that does not require Advanced Intelligent Network Release 1 features (e.g., a POTS call).

Acronym or Term	Definition
Bc	committed burst. Negotiated tariff metric in Frame Relay internetworks. The maximum amount of data (in bits) that a Frame Relay internetwork is committed to accept and transmit above the CIR. *See also* Be *and* CIR.
Be	excess burst. Negotiated tariff metric in Frame Relay internetworks. The number of bits that a Frame Relay internetwork attempts to transmit after Bc is accommodated. Be data, in general, is delivered with a lower probability than Bc data because Be data can be marked as DE by the network. *See also* Bc.
Bell operating company	*See* BOC.
BHCA	busy hour call attempts. A traffic engineering term that refers to the number of call attempts made during the busiest hour of the day.
binary 8-zero substitution	*See* B8ZS.
BLF	busy lampfield. A visual display of the status of all or some of your phones. Your BLF tells you if a phone is busy or on hold. Your BLF is typically attached to or part of your operator phone.
BOC	Bell operating company. BOC is a term for any of the 22 original companies (or their successors) that were created when AT&T was broken up in 1983 and given the right to provide local telephone service in a given geographic area. The companies had previously existed as subsidiaries of AT&T and were called the "Bell System." The purpose of the breakup was to create competition at both the local and long-distance service levels. BOCs compete with other, independent companies to sell local phone service. In certain areas, long-distance companies, including AT&T, can now compete for local service. Collectively, companies offering local phone service are referred to legally as local exchange carriers (LECs). BOCs are not allowed to manufacture equipment and were initially not allowed to provide long-distance service. The Telecommunications Act of 1996 now permits them to engage in long-distance business under certain circumstances. As of 1996, the BOCs consisted of original and successor companies to: Bell Telephone Company of Nevada, Illinois Bell, Indiana Bell, Michigan Bell, New England Telephone and Telegraph Company, New Jersey Bell, New York Telephone Company, U S West Communications Company, South Central Bell, Southern Bell, Southwestern Bell, Bell Telephone of Pennsylvania, The Chesapeake and Potomac Telephone Company, The Chesapeake and Potomac Telephone Company of Maryland, The Chesapeake and Potomac Telephone Company of Virginia, The Diamond State Telephone Company, The Ohio Bell Telephone Company, The Pacific Telephone and Telegraph Company, and the Wisconsin Telephone Company.
BRI voice module	*See* BVM.
BRQ	bandwidth change request. RAS bandwidth control message sent by endpoint to gatekeeper requesting an increase/decrease in call bandwidth.
busy hour call attempts	*See* BHCA.
busy lampfield	*See* BLF.
BVM	BRI voice module. An optional device for Cisco modular routers providing four ISDN BRI ports for connection to ISDN PBXs or PINXs. The BVM has four ISDN BRI ports for voice traffic. Each BRI port supports two voice channels (ISDN B channels) and one signaling channel (ISDN D channel).
calling line ID	*See* CLID.
CAS	channel associated signaling. The transmission of signaling information in association with the voice channel. In T1 networks, CAS signaling often is referred to as "robbed-bit" signaling because the network is robbing user bandwidth for other purposes.
CBR	constant bit rate. QoS class defined by the ATM Forum for ATM networks. CBR is used for connections that depend on precise clocking to ensure undistorted delivery.
CBWFQ	class-based weighted fair queuing. Congestion management mechanism that extends the standard WFQ functionality to provide support for user-defined traffic classes.

Acronym or Term	Definition
CCIS	common channel interoffice signaling. A technology that uses a common link to carry signaling information for a number of trunks. CCIS is similar to ITU-T SS6 protocol that operated at low bit rates (2.4, 4.8, and 9.6 kbps) and transmitted messages that were only 28 bits in length.
CCITT	Consultative Committee for International Telegraph and Telephone. Former name for the International organization responsible for the development of communications standards. Now called the ITU-T. *See also* ITU-T.
CCS	common channel signaling. Signaling system used in telephone networks that utilizes a statistical multiplexing protocol for signaling. A specified channel is exclusively designated to carry signaling information for all channels in the system. An example is ISDN or SS7. *See also* SS7.
CDVT	cell delay variation tolerance. In ATM, a QoS parameter for managing traffic that is specified when a connection is set up. In CBR transmissions, CDVT determines the level of jitter that is tolerable for the data samples taken by the PCR. *See also* CRB.
cell delay variation tolerance	*See* CDVT.
CELP	code excited linear prediction. Compression algorithm used in low bit-rate voice encoding. Used in ITU-T Recommendations G.728, G.729, G.723.1.
central office	*See* CO.
centum call seconds	Units used to measure traffic load. A CCS is 1/36th of an erlang. The formula for a centum call second is the number of calls per hour multiplied by their average duration in seconds, all divided by 100.
CES	circuit emulation service. Enables users to multiplex or to concentrate multiple circuit emulation streams for voice and video with packet data on a single high-speed ATM link without a separate ATM access multiplexer.
channel associated signaling	*See* CAS.
channel ID	*See* CID.
CID	channel ID. Designates the Frame Relay subchannel ID for Voice over Frame Relay.
CIR	committed information rate. The rate at which a Frame Relay network agrees to transfer information under normal conditions, averaged over a minimum increment of time. CIR, measured in bits per second, is one of the key negotiated tariff metrics. *See also* Bc.
circuit emulation service	*See* CES.
Cisco Architecture for Voice, Video and Integrated Data	*See* Cisco AVVID.
Cisco AVVID	Cisco Architecture for Voice, Video and Integrated Data. Cisco AVVID is the architecture for Voice, Video and Integrated Data. Cisco AVVID includes three components: infrastructure, such as switches and routers; clients, such as IP Phones, H.323 videoconferencing equipment, and PCs; and applications, such as call control, that use a common IP network.
Cisco CallManager	Software-based call-processing agent. It is a component of the Cisco IP telephony solution, part of Cisco AVVID. The software extends enterprise telephony features and functions to packet telephony network devices such as IP Phones, media processing devices, VoIP gateways, and multimedia applications.
Cisco.com	The name of the Cisco Systems external website.
Cisco ICM software	Cisco Intelligent Call Management software. Software, which delivers an integrated suite of contact center capabilities. Cisco ICM software provides intelligent queue management in a contact center environment. It enables improved queue management across a variety of ACDs from different vendors as well as integrating IVRs, database and desktop applications, and CTI solutions

Acronym or Term	Definition
Cisco Intelligent Call Management software	*See* Cisco ICM software.
Cisco IOS	Cisco Systems software that provides common functionality, scalability, and security for all products under the CiscoFusion architecture. Cisco IOS software allows centralized, integrated, and automated installation and management of internetworks while ensuring support for a wide variety of protocols, media, services, and platforms.
Cisco IPCC	Cisco IP Contact Center. An integrated suite of products that enables contact center agents using Cisco IP Phones to receive both TDM and VoIP calls. IPCC provides ACD and IVR capabilities in a single-vender IP suite. The IPCC can be implemented in a single-site environment or integrated into an enterprise-wide multisite contact center.
Cisco IP Contact Center	*See* Cisco IPCC.
Cisco IP Phone	The Cisco family of IP Phones provides a complete range of intelligent communication systems that use the data network while providing the convenience and ease of use of a business telephone.
Cisco IP SoftPhone	A Windows-based application for the PC. Used as a standalone end station or in conjunction with the Cisco IP Phone, it provides mobility, directory integration, user interface, and a virtual conference room.
class-based weighted fair queuing	*See* CBWFQ.
CLEC	competitive local exchange carrier. A company that builds and operates communication networks in metropolitan areas and provides its customers with an alternative to the local telephone company. In the United States, a CLEC is a company that competes with the already established local telephone business by providing its own network and switching. The term distinguishes new or potential competitors from established local exchange carriers (LECs) and arises from the Telecommunications Act of 1996, which was intended to promote competition among both long-distance and local phone service providers. North American Telecom and Winstar Communications are examples of CLECs, which are generally listed as simply “local exchange carriers.”
CLI	command-line interface. An interface that allows the user to interact with the operating system by entering commands and optional arguments. The UNIX operating system and Microsoft MS-DOS provide CLIs.
CLID	calling line ID. Information about the billing telephone number from which a call originated. The CLID value might be the entire telephone number, the area code, or the area code plus the local exchange. Also known as Caller ID.
CNG	comfort noise generation. While using VAD, the DSP at the destination emulates background noise from the source side, preventing the perception that a call is disconnected.
CO	central office. The local telephone company office to which all local loops in a given area connect and in which circuit switching of subscriber lines occurs.
“Codebook” excitation index	Used by the receiver to look up a set of excitation values. A codebook is a set of rules that helps to determine what conditions indicate a Cisco device fault.
code excited linear prediction	*See* CELP.
comfort noise generation	*See* CNG.
command-line interface	*See* CLI.
committed burst	*See* Bc.
committed information rate	*See* CIR.

Acronym or Term	Definition
common channel interoffice signaling	*See* CCIS.
common channel signaling	*See* CCS.
competitive local exchange carrier	*See* CLEC.
Compressed Real-Time Transport Protocol	*See* CRTP.
computer telephony integration	*See* CTI.
Conjugate Structure Algebraic Code Excited Linear Prediction	*See* CS-ACELP.
constant bit rate	*See* CBR.
Consultative Committee for International Telegraph and Telephone	*See* CCITT.
CPE	customer premises equipment. (1) Terminating equipment, such as terminals, telephones, and modems, installed at customer sites, and connected to the telephone company network. (2) Any telephone equipment residing on the customer site.
CRC	cyclic redundancy check. Error-checking technique in which the frame recipient calculates a remainder by dividing frame contents by a prime binary divisor and compares the calculated remainder to a value stored in the frame by the sending node.
cross-connect (adj.) cross connect (n, v)	Cross connect is a connection scheme between cabling runs, subsystems, and equipment, using patch cords or jumpers that attach to connecting hardware on each end. Cross-connection is the attachment of one wire to another, usually by anchoring each wire to a connecting block and then placing a third wire between them so that an electrical connection is made. The TIA/EIA-568-A standard specifies that cross-connect cables (also called patch cords) are to be made out of stranded cable.
CRTP	Compressed Real-Time Transport Protocol. A type of header compression designed to reduce the IP/UDP/RTP headers to two bytes for most packets in the case where no UDP checksums are being sent, or four bytes with checksums.
CS-ACELP	Conjugate Structure Algebraic Code Excited Linear Prediction. CELP voice compression algorithm providing 8 kbps, or 8:1 compression, standardized in ITU-T Recommendation G.729 or G.729A.
CTI	computer telephony integration. The name given to the merger of traditional telecommunications (PBX) equipment with computers and computer applications. The use of caller ID to retrieve customer information automatically from a database is an example of a CTI application.
customer premises equipment	*See* CPE.
cyclic redundancy check	*See* CRC.
DACS	digital access and crossconnect system. A digital cross-connect system that provides grooming, switching, and aggregation.
data carrier detect	*See* DCD.
data circuit-terminating equipment (ITU-T expansion)	*See* DCE.

Acronym or Term	Definition
data communications equipment (EIA expansion)	*See* DCE.
data-link connection identifier	*See* DLCI.
data terminal equipment	*See* DTE.
data terminal ready	*See* DTR.
dB	decibel. Unit for measuring relative power ratios in terms of gain or loss. The rule of thumb to remember is that 10 dB indicates an increase (or a loss) by a factor of 10; 20 dB indicates an increase (or a loss) by a factor of 100; 30 dB indicates an increase (or a loss) by a factor of 1000.
DCD	data carrier detect. Signal from the DCE (modem or printer) to the DTE (typically your PC), indicating that the modem is receiving a carrier signal from the DCE (modem) at the other end of the telephone circuit.
DCE	data communications equipment (EIA expansion). data circuit-terminating equipment (ITU-T expansion). Devices and connections of a communications network that comprise the network end of the user-to-network interface. The DCE provides a physical connection to the network, forwards traffic, and typically provides a clocking signal used to synchronize data transmission between DCE and DTE devices. Modems and interface cards are examples of DCE.
DDS	digital data service. A class of service that is offered by telecommunications companies to transport data rather than voice. Originally called Dataphone Digital Service by AT&T in the late 1970s.
DE bits	discard eligible bits. Bits that are used to tag Frame Relay frames that are eligible to be discarded if the network gets congested.
decibel	*See* dB.
delay budget	The maximum amount of delay in data, voice, and video applications. The total end-to-end delay when engineering a VoIP implementation should not exceed the 150- to 200-ms delay budget.
Delay Dial	A signaling method in which the terminating side remains off hook until it is ready to receive address information. The off-hook interval is the delay dial signal.
DHCP	Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.
Dialed Number Identification Service	*See* DNIS.
dial plan mapper	Provides the mapping of IP addresses to telephone numbers. After enough digits are accumulated to match a configured destination pattern, the dial plan mapper maps the IP host to a telephone number.
dial pulse	*See* DP.
dialup (adj, n) dial up (v)	Modem access to a data network. The use of a dial or push-button telephone to create a telephone or data call. Dialup calls are usually billed by time of day, duration of call, and distance traveled. It is a connection to the Internet, or any network, where a modem and a standard telephone are used to make a connection between computers.

Acronym or Term	Definition
dialup remote access server	A remote access server is computer hardware that resides on a corporate LAN and into which employees dial on the PSTN to get access to their e-mail and to software and data on the corporate LAN (for example, status on customer orders). Remote access servers are also used by commercial service providers, such as ISPs, to allow their customers access into their networks. Remote access servers are typically measured by how many simultaneous dial-in users (on analog or digital lines) they can handle and whether they can work with cheaper digital circuits, such as T 1 and E 1 connections.
digital access and crossconnect system	*See* DACS.
digital data service	*See* DDS.
Digital Private Network Signaling System	*See* DPNSS.
digital service level zero	*See* DS0.
digital signal processor	*See* DSP.
digital speech interpolation	*See* DSI.
digital subscriber line	*See* DSL.
Digital T1/E1 Packet Voice Trunk Network Module	A flexible and scalable T1/E1 voice solution for Cisco 2600 and 3600 series Modular Access routers that supports up to 60 voice channels in a single network module.
Digital T1/E1 voice port adapter	A single-width port adapter, which incorporates one or two universal ports configurable for either T1 or E1 connection with high-performance DSP support for up to 24 to120 channels of compressed voice.
discard eligible bits	*See* DE bits.
disengage request	*See* DRQ.
DLCI	data-link connection identifier. Value that specifies a PVC or an SVC in a Frame Relay network. In the basic Frame Relay specification, DLCIs are locally significant (connected devices might use different values to specify the same connection at different ends of the network).
DNIS	Dialed Number Identification Service. Feature of trunk lines where the called number is identified; this called number information is used to route the call to the appropriate service. DNIS is a service used with toll-free dedicated services whereby calls placed to specific toll-free numbers are routed to the appropriate area within the company.
DP	dial pulse. A means of signaling that consists of regular momentary interruptions of a direct or alternating current at the sending end in which the number of interruptions corresponds to the value of the digit or character. In short, the old style of rotary dialing. Dial the number "5" and you will hear five "clicks."
DPNSS	Digital Private Network Signaling System. A common-channel, message-oriented signaling protocol commonly used by PBXs.
drop and insert	Allows DS-0 channels from one T1 or E1 facility to be cross-connected digitally to DS-0 channels on another T1 or E1. By using this method, channel traffic is sent between a PBX and a CO PSTN switch or other telephony device, so that some PBX channels are directed for long-distance service through the PSTN while the router compresses others for interoffice VoIP calls. In addition, drop and insert can cross connect a telephony switch (from the CO or PSTN) to a channel bank for external analog connectivity. Also called TDM Cross-Connect. *See* DACS.
DRQ	disengage request. RAS message sent by the gateway to the gatekeeper during the process of a call. The gateway waits for the DCF message before it sends the setup message to the new destination gatekeeper.

Acronym or Term	Definition
DS0	digital service level zero. Single timeslot on a DS1 (also known as T1) digital interface—that is, a 64-kbps, synchronous, full-duplex data channel, typically used for a single voice connection on a PBX. Also, a single timeslot on an E1.
DSI	digital speech interpolation. An algorithm that analyzes voice channels for silence. It suppresses the voice bits to conserve packet-line bandwidth and inserts a code to indicate to the far end that these bits have been removed. Also referred to as VAD.
DSL	digital subscriber line. Public network technology that delivers high bandwidth over conventional copper wiring at limited distances. There are four types of DSL: ADSL, HDSL, SDSL, and VDSL. All are provisioned via modem pairs, with one modem located at a central office and the other at the customer site. Because most DSL technologies do not use the whole bandwidth of the twisted pair, there is room remaining for a voice channel.
DSP	digital signal processor. An electronic circuit that compresses voice signals, generates tones, and decodes received compressions. DSPs can also emulate modems for purposes of fax relay.
DTE	data terminal equipment. Device at the user end of a user-network interface that serves as a data source, destination, or both. DTE connects to a data network through a DCE device (for example, a modem) and typically uses clocking signals generated by the DCE. DTE includes such devices as computers, protocol translators, and multiplexers.
DTMF	dual tone multifrequency. Tones generated when a button is pressed on a telephone to convey address signaling.
DTR	data terminal ready. EIA/TIA-232 circuit that is activated to let the DCE know when the DTE is powered up and not in test mode.
dual tone multifrequency	*See* DTMF.
Dynamic Host Configuration Protocol	*See* DHCP.
E&M	ear and mouth. Earth and Magneto. recEive and transMit. (1) Trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco analog E&M interface is an 8-pin modular connector that allows connections to PBX trunk lines (tie-lines). E&M also is emulated on E1 and T1 digital interfaces. (2) A type of signaling traditionally used in the telecommunications industry. Indicates the use of a handset that corresponds to the ear (receiving) and mouth (transmitting) component of a telephone.
ear and mouth	*See* E&M.
Earth and Magneto	*See* E&M.
ECMA	European Computer Manufacturers Association. Group of European computer vendors who have done substantial OSI standardization work.
E-lead	The wiring arrangement on an E&M circuit in which the signal side sends its signaling information.
ESF	Extended Superframe. Framing type used on T1 circuits that consists of 24 frames of 193 bits each, with the 193rd bit providing framing information and other functions. ESF is an enhanced version of SF. *See also* SF.
ETSI	European Telecommunication Standards Institute. ETSI is a nonprofit organization producing voluntary telecommunications standards used throughout Europe.
European Computer Manufacturers Association	*See* ECMA.
European Telecommunication Standards Institute	*See* ETSI.
excess burst	*See* Be.

Acronym or Term	Definition
Extended Superframe	*See* ESF.
FDM	frequency-division multiplexing. Technique whereby information from multiple channels can be allocated bandwidth on a single wire based on frequency. An example is DSL.
FIFO	first-in/first-out. Refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queuing scheme where the first calls received are the first calls processed.
first-in/first-out	*See* FIFO.
flash memory	A special type of electrically erasable programmable read-only memory (EEPROM) that can be erased and reprogrammed in blocks instead of one byte at a time. Many modern PCs have their basic input/output system (BIOS) stored on a flash memory chip so that it can be updated easily if necessary. Such a BIOS is sometimes called a flash BIOS. Flash memory is also popular in modems because it enables the modem manufacturer to support new protocols as they become standardized.
Foreign Exchange Office	*See* FXO.
Foreign Exchange Station	*See* FXS.
four-wire	One of two distinct types of audio interfaces (two-wire or four-wire). The four-wire implementation provides separate paths for receiving and sending audio signals, consisting of T, R, and T1, R1 leads.
frame forwarding	Mechanism by which frame-based traffic, such as HDLC and SDLC, traverses an ATM network.
Frame Relay traffic shaping	*See* FRTS.
frequency-division multiplexing	*See* FDM.
FRTS	Frame Relay traffic shaping. Queuing method that uses queues on a Frame Relay network to limit surges that can cause congestion. Data is buffered and sent into the network in regulated amounts to ensure that the traffic can fit within the promised traffic envelope for the particular connection.
FXO	Foreign Exchange Office. An FXO interface connects to the PSTN central office. Cisco FXO interface is an RJ-11 connector that allows an analog connection at the PSTN's central office or to a station interface on a PBX.
FXS	Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, key sets, and PBXs.
gatekeeper	(1) The component of an H.323 telephony system that performs call address resolution, admission control, and subnet bandwidth management. (2) Telecommunications: H.323 entity on a LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper can provide other services to the H.323 terminals and gateways, such as bandwidth management and locating gateways. A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup and request admission to a call from the gatekeeper.
gatekeeper discovery request	*See* GRQ.
gateway	An H.323 term that describes the component of a H.323 telephony network that translates between one technology and another, typically between traditional telephony and TCP/IP.
generic traffic shaping	Shapes traffic by reducing outbound traffic flow to avoid congestion by constraining traffic to a particular bit rate using the token bucket mechanism.
GRQ	gatekeeper discovery request. RAS gatekeeper discovery message sent by endpoint to gatekeeper.

Acronym or Term	Definition
HDB3	high density binary 3. A line coding method used to maintain synchronization by ensuring a sufficient number of binary ones. HDB3 is used on E1 circuits.
HDLC	High-Level Data Link Control. Bit-oriented synchronous data-link-layer protocol developed by International Organization for Standardization (ISO). *See also* SDLC.
high density binary 3	*See* HDB3.
High-Level Data Link Control	*See* HDLC.
Hoot and Holler	A broadcast audio network used extensively by the brokerage industry for market updates and trading. Similar networks are used in publishing, transportation, power plants, and manufacturing.
Hot Standby Router Protocol	*See* HSRP.
HSRP	Hot Standby Router Protocol. Provides high network availability and transparent network topology changes. HSRP creates a hot standby router group with a lead router that services all packets sent to the hot standby address. Other routers in the group monitor the lead router, and if it fails, one of these standby routers inherits the lead position and the hot standby group address.
HTTP	Hypertext Transfer Protocol. The protocol used by web browsers and web servers to transfer files, such as text and graphic files.
Hyperterm software	Terminal emulation software.
Hypertext Transfer Protocol	*See* HTTP.
IC	*See* IXC.
IETF	Internet Engineering Task Force. Task force consisting of over 80 working groups responsible for developing Internet standards
ILEC	incumbent local exchange carrier. An ILEC is a telephone company in the United States that was providing local service when the Telecommunications Act of 1996 was enacted. ILECs include the former Bell operating companies (BOCs), which were grouped into holding companies known collectively as the regional Bell operating companies (RBOCs) when the Bell System was broken up by a 1983 consent decree. ILECs are in contradistinction to competitive local exchange carriers (CLECs). A “local exchange” is the local “central office” of an LEC. Lines from homes and businesses terminate at a local exchange. Local exchanges connect to other local exchanges within a local access and transport area (LATA) or to inter-exchange carriers (IXCs), such as long-distance carriers AT&T, MCI, and Sprint.
IMAP	Internet Message Access Protocol. Method of accessing e-mail or bulletin board messages kept on a mail server that can be shared. IMAP permits client e-mail applications to access remote message stores as if they were local without actually transferring the message.
IMT	Inter-Machine Trunk. A means to give service providers access to more favorable tariffs and rates. In SS7 environments, IMTs terminate bearer traffic on the voice gateways.
IN	Intelligent Network. A network that provides IP routing, QoS, network access and control, and network management services.
incumbent local exchange carrier	*See* ILEC.
Integrated Services Digital Network	*See* ISDN.
Intelligent Network	*See* IN.
interactive voice response	*See* IVR.

Acronym or Term	Definition
inter-exchange carrier	*See* IXC.
Inter-Machine Trunk	*See* IMT.
International Telecommunication Union	*See* ITU.
International Telecommunication Union Telecommunication Standardization Sector	*See* ITU-T.
Internet Engineering Task Force	*See* IETF.
Internet Message Access Protocol	*See* IMAP.
Internet Protocol	*See* IP.
IP	Internet Protocol. Network layer protocol in the TCP/IP stack offering a connectionless internetwork service. IP provides features for addressing, type-of-service specification, fragmentation and reassembly, and security. Defined in RFC 791.
IP cloud	The area in which data travels through an IP network. Illustrated in diagrams as a cloud.
IP precedence	A 3-bit value in the ToS byte used for assigning precedence to IP packets.
IP RTP priority	A Frame Relay feature that provides a strict priority queuing scheme on a Frame Relay PVC for delay-sensitive data, such as voice.
ISDN	Integrated Services Digital Network. Communication architecture offered by telephone companies that permits customers to access digital networks to carry data, voice, and other source traffic.
ISDN User Part	*See* ISUP.
ISUP	ISDN User Part. SS7 protocol layer that defines the protocol used to prepare, manage, and release trunks that carry voice and data between calling and called parties under the auspice of ISDN.
ITU	International Telecommunication Union. An organization established by the United Nations to set international telecommunications standards and to allocate frequencies for specific uses.
ITU-T	International Telecommunication Union Telecommunication Standardization Sector. International body that develops worldwide standards for telecommunications technologies. The ITU-T carries out the functions of the former CCITT. *See also* CCITT.
IVR	interactive voice response. Term used to describe systems that provide information in the form of recorded messages over telephone lines in response to user input in the form of spoken words, or, more commonly, DTMF signaling. Examples include banks that allow you to check your balance from any telephone, and automated stock quote systems.
IXC	inter-exchange carrier. Common carrier providing long-distance connectivity between local access and transport areas (LATAs). The three major IXCs are AT&T, MCI, and Sprint, but several hundred IXCs offer long-distance service in the United States.
Java Telephony Application Programming Interface	*See* JTAPI.
JTAPI	Java Telephony Application Programming Interface. A Java API for call control developed by Sun Microsystems.
LCC	Lost Calls Cleared
LCD	Lost Calls Delayed

Acronym or Term	Definition
LCH	Lost Calls Held
LDAP	Lightweight Directory Access Protocol. Protocol that provides read/write interactive access to X.500 Directories for uniform application security and access levels.
LDCELP	low-delay CELP. CELP voice compression algorithm providing 16-kbps, or 4:1, compression. Standardized in ITU-T Recommendation G.728.
least significant bit	*See* LSB.
LEC	local exchange carrier. LEC is the term for a public telephone company in the U.S. that provides local service. Some of the largest LECs are the Bell operating companies (BOCs), which were grouped into holding companies known collectively as the regional Bell operating companies (RBOCs) when the Bell System was broken up by a 1983 consent decree. In addition to the Bell companies, there are a number of independent LECs, such as GTE. LEC companies are also sometimes referred to as “telcos.” A “local exchange” is the local “central office” of an LEC. Lines from homes and businesses terminate at a local exchange. Local exchanges connect to other local exchanges within a local access and transport area (LATA) or to interexchange carriers (IXCs) such as long-distance carriers AT&T, MCI, and Sprint.
LFI	link fragmentation and interleaving. A Cisco IOS feature that reduces delay on slower-speed links by breaking up large datagrams and interleaving low-delay traffic packets with the smaller packets resulting from the fragmented datagram.
Lightweight Directory Access Protocol	*See* LDAP.
linear predictive coding	*See* LPC.
line code	Electrical modulation scheme used by digital carrier systems. In North America, T1 uses AMI or B8ZS line coding. In other countries, E1 uses AMI or HDB3 line coding.
link fragmentation and interleaving	*See* LFI.
LLQ	Low Latency Queuing. Enables use of a single priority queue in conjunction with CBWFQ. Typically, the priority queue only carries VoIP traffic. All other traffic is carried in the user-defined queues of CBWFQ.
local exchange carrier	*See* LEC.
location request	*See* LRQ.
low-delay CELP	*See* LDCELP.
Low Latency Queuing	*See* LLQ.
LPC	linear predictive coding. Voice coding that uses a special algorithm that models the way human speech works. Because LPC can take advantage of an understanding of the speech process, it can be efficient without sacrificing voice quality.
LRQ	location request. RAS location request message sent to request the gatekeeper contact information for one or more E.164 addresses.
LSB	least significant bit. The bit of a binary expression having the least value; or, representing the number of ones.
Main Distribution Frame	*See* MDF.
MC	multipoint controller. A required part of an MCU. The MC is the conference controller. The MC handles negotiation between all terminals to determine common capabilities and controls conference resources such as multicasting. The MC does not deal directly with any of the media streams.
MCSs	media convergence servers. An integral component of the Cisco IP Communications system. A high availability server platform for Cisco AVVID.

Acronym or Term	Definition
MCU	multipoint control unit. A component that manages videoconferences of three or more participants.
MDF	Main Distribution Frame. The point where all network-related external services, IP equipment, and wiring converge within a building.
mean opinion score	*See* MOS.
media convergence servers	*See* MCSs.
Media Gateway Control Protocol	*See* MGCP.
media termination point	*See* MTP.
Meet-Me conference	A conference feature where everyone who dials the same Meet-Me number will join the conference together.
MEL CAS	Mercury Exchange Limited Channel Associated Signaling. A voice signaling protocol used primarily in the United Kingdom.
Mercury Exchange Limited Channel Associated Signaling	*See* MEL CAS.
MGCP	Media Gateway Control Protocol. Protocol that helps bridge the gap between circuit-switched and IP networks. A combination of IPDC and SGCP, MGCP allows external control and management of data communications devices, or "media gateways" at the edge of multiservice packet networks by software programs.
MICA	Modem ISDN Channel Aggregation. Modem module and card used in the Cisco AS5300 universal access servers. A MICA modem provides an interface between an incoming or outgoing digital call and an ISDN telephone line; the call does not have to be converted to analog as it does with a conventional modem and an analog telephone line. Each line can accommodate, or aggregate, up to 24 (T1) or 30 (E1) calls.
Microsoft NetMeeting	Complete H.323 desktop Internet multimedia solution for all Windows users with multipoint data conferencing, text chat, whiteboard, and file transfer, as well as point-to-point audio and video.
M-lead	The wiring arrangement on an E&M circuit in which the trunking side sends its signaling information.
MLP	Multilink Point-to-Point Protocol. Method of splitting, recombining, and sequencing datagrams across multiple logical data links under the PPP protocol.
Modem ISDN Channel Aggregation	*See* MICA.
MOS	mean opinion score. A common benchmark used to determine the perceived quality of sound produced by specific codecs.
MP	multipoint processor. Part of an MCU. The MP processes the media streams. It receives audio, video, or data bits from the endpoints for which it does the required mixing, switching, and other processing before distributing the stream to the videoconference participants.
MTP	media termination point. A Cisco software application. An MTP software device allows the Cisco CallManager to extend supplementary services, such as hold and transfer, to calls routed through an H.323 endpoint or an H.323 gateway.
multicast backbone	Multicast backbone of the Internet. It is a virtual multicast network composed of multicast LANs and the point-to-point tunnels that interconnect them.
Multilink Point-to-Point Protocol	*See* MLP.
multipoint controller	*See* MC.

Acronym or Term	Definition
multipoint control unit	*See* MCU.
multipoint processor	*See* MP.
network service access point	*See* NSAP.
NSAP	network service access point. Network address, as specified by ISO. An NSAP is the point at which OSI network service is made available to a transport layer (Layer 4) entity.
ODBC	Open Database Connectivity. Abstracts data USING applications from database management systems. Standard API for accessing data in both relational and nonrelational database management systems. Using this API, common database applications can be written to access data stored in a variety of database management systems on a variety of computers regardless of DBMS or programming interface.
off hook	Call condition in which transmission facilities are already in use. Also known as *busy*.
Off-Premises eXtension	*See* OPX.
OMAP	operations, maintenance, administration, and provisioning. Telephony operations functions include monitoring and discovery of problems before they negatively impact service. Administration deals with billing, department cross-charges, accounting, and capacity management. The telephony maintenance function is quite similar to the data networking processes of fault isolation and correction. The final element, provisioning, is used to define services for individual subscribers.
on hook	(1) Condition that exists when a receiver or a handset is resting on the switch hook, or is not in use. (2) Idle state (open loop) of a single telephone or PBX line loop.
OOS	Out-of-Service. State of the call or trunk.
Open Database Connectivity	*See* ODBC.
Open System Interconnection	*See* OSI.
operations, maintenance, administration, and provisioning	*See* OMAP.
OPX	Off-Premises eXtension. A telephone line from a telephone system that is terminated in a different building than the one in which the telephone system resides.
OSI	Open System Interconnection. International standardization program created by ISO and ITU-T to develop standards for data networking that facilitate multivendor equipment interoperability.
Out-of-Service	*See* OOS.
packet voice digital signal processor module	*See* PVDM.
PAM	pulse amplitude modulation. Modulation scheme where the modulating wave is caused to modulate the amplitude of a pulse stream.
PBX	private branch exchange. Digital or analog telephone switches located on the customer premises and used to connect private and public telephone networks.
PCI	protocol control information. Control information added to user data to compose an OSI packet. The OSI equivalent of the term “header.”
PCM	pulse code modulation. Technique of encoding analog voice into a 64-kbps data stream by sampling with 8-bit resolution at a rate of 8000 times per second.
PCMCIA	Personal Computer Memory Card Industry Association. A standard interface that connects any type of device to a portable computer. Developed by PCMCIA in the early 1990s.

Acronym or Term	Definition
Perceptual Speech Quality Measurement	*See* PSQM.
permanent virtual circuit	*See* PVC.
Personal Computer Memory Card Industry Association	*See* PCMCIA.
phase-lock loop	*See* PLL.
PINX	private integrated services network exchange. A PBX or key system, which in a BRI voice application uses QSIG signaling.
plain old telephone service	*See* POTS.
PLAR	private line, automatic ringdown. Voice circuit that connects two single endpoints together. When either telephone handset is taken off hook, the remote telephone automatically rings.
PLAR Off-Premises eXtension	*See* PLAR-OPX.
PLAR-OPX	PLAR Off-Premises eXtension. Specifies a PLAR Off-Premises eXtension connection. Using this option, the local voice port provides a local response before the remote voice port receives an answer. On FXO interfaces, the voice port will not answer until the remote side answers.
PLL	phase-lock loop. A circuit on a T1 or E1 module that provides clocking information.
point of presence	*See* POP.
POP	point of presence. In OSS, a physical location where an inter-exchange carrier installed equipment to interconnect with an LEC.
Post, Telephone, and Telegraph	*See* PTT.
POTS	plain old telephone service. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the PSTN. *See also* PSTN.
PQ	priority queuing. PQ ensures that important traffic gets the fastest handling at each point where it is used. It was designed to give strict priority to important traffic.
priority queuing	*See* PQ.
private branch exchange	*See* PBX.
private integrated services network exchange	*See* PINX.
private line, automatic ringdown	*See* PLAR.
protocol control information	*See* PCI.
PSQM	Perceptual Speech Quality Measurement. A technique used for measuring voice quality. It compares the received audio with the transmitted audio.
PSTN	public switched telephone network. General term referring to the variety of telephone networks and services in place worldwide. Sometimes called POTS.
PTT	Post, Telephone, and Telegraph. Government agency that provides telephone services. PTTs exist in most areas outside North America and provide both local and long-distance telephone services.
public switched telephone network	*See* PSTN.

Acronym or Term	Definition
pulse amplitude modulation	*See* PAM.
pulse code modulation	*See* PCM.
PVC	permanent virtual circuit. Connections that are assigned but not connected until data is sent, thereby not using bandwidth when idle. A virtual circuit that is permanently established. PVCs save bandwidth associated with circuit establishment and teardown in situations where certain virtual circuits must exist all the time. In ATM terminology, called a permanent virtual connection.
PVDM	packet voice digital signal processor module. Provides the ability to increase the voice processing capabilities within a single network module.
QoS	quality of service. (1) Measure of performance for a transmission system that reflects its transmission quality and service availability. (2) A set of tools used in networking devices to ensure best-of-class transmission quality and service availability.
QSIG	Q Signaling. An inter-PBX signaling protocol for networking PBX supplementary services in a multi- or single-vendor environment.
Q Signaling	*See* QSIG.
quality of service	*See* QoS.
RAS	registration, admission, and status. Protocol that is used between endpoints and the gatekeeper to perform management functions. RAS signaling function performs registration, admissions, bandwidth changes, status, and disengage procedures between the VoIP gateway and the gatekeeper.
RBOCs	regional Bell operating companies. Seven regional telephone companies formed by the breakup of AT&T. RBOCs differ from regional Bell holding companies (RBHCs) in that RBOCs do not cross boundaries that were set out by the consent decree. Regional Bell operating company (RBOC) is a term describing one of the U.S. regional telephone companies (or their successors) that were created as a result of the breakup of American Telephone and Telegraph Company (AT&T, known also as the Bell System or “Ma Bell”) by a U.S. Federal Court consent decree on December 31, 1983. The seven original regional Bell operating companies were Ameritech, Bell Atlantic, BellSouth, NYNEX, Pacific Bell, Southwestern Bell, and US WEST. Each of these companies owned at least two Bell operating companies (BOCs). The BOCs were given the right to provide local phone service while AT&T was allowed to retain its long-distance service. The RBOCs and their constituent BOCs are part of the class of local exchange carriers (LECs). In addition to the RBOCs, there are more than 100 other franchised local telephone companies classed as local exchange carriers. Competitive local exchange carriers (CLECs) are additional companies allowed to compete with the LECs. These include AT&T, in some localities, and power companies. An interexchange carrier (IXC) is a long-distance carrier that carries traffic between LECs. Under the Telecommunications Act of 1996, RBOCs and LECs are allowed to compete for long-distance telephone traffic under certain circumstances. RBOCs are generally in competition for digital data and Internet traffic with wireless service providers and cable TV companies. RBOCs are gradually making available new telephone carrier technologies such as ISDN and DSL.
RBS	robbed-bit signaling. A technique by which a single bit in every DS0 bearer channel is "stolen" from every sixth frame. The stolen bit is then used to carry signaling information.
Real Time Streaming Protocol	*See* RTSP.
Real-Time Transport Protocol	*See* RTP.
recEive and transmit	*See* E&M.

Acronym or Term	Definition
redirect server	A server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client. It does not initiate its own SIP request nor does it accept calls.
reduced instruction set computing	*See* RISC.
regional Bell operating companies	*See* RBOCs.
registration, admission, and status	*See* RAS.
registration request	*See* RRQ.
Request For Comments	*See* RFC.
request notification	*See* RQNT.
Resource Reservation Protocol	*See* RSVP.
RFC	Request For Comments. Document series generated by the IETF and used as the primary means for communicating information about the Internet. Some RFCs are designated by the Internet Architecture Board (IAB) as Internet standards. Most RFCs document protocol specifications, such as Telnet and FTP, but some are humorous or historical. RFCs are available online from numerous sources.
RISC	reduced instruction set computing. A microprocessor design that provides fewer and simpler instructions burned into the silicon. Fewer instructions let the processor perform at a higher speed. The difference is made up by requiring more work to be done by compilers and greater memory usage.
robbed-bit signaling	*See* RBS.
round robin (n) round-robin (adj)	An algorithm used to schedule processes in a fixed cyclic order. Simply put, it means to "take turns."
RQNT	request notification. RAS message that instructs the gateway to watch for specific events.
RRQ	registration request. RAS message sent as a registration request.
RSVP	Resource Reservation Protocol. Protocol that supports the reservation of resources across an IP network. Applications running on IP end systems can use RSVP to indicate to other nodes the nature (bandwidth, jitter, maximum burst, and so on) of the packet streams they want to receive. Also known as Resource Reservation Setup Protocol.
RTCP	Real-Time Transport Control Protocol. Protocol that monitors the QoS of an IP RTP connection and conveys information about the ongoing session.
RTP	Real-Time Transport Protocol. Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time stamping, and delivery monitoring to real-time applications.
Real-Time Transport Control Protocol	*See* RTCP.
RTSP	Real Time Streaming Protocol. Enables the controlled delivery of real-time data, such as audio and video. Sources of data can include both live data feeds, such as live audio and video, and stored content, such as pre-recorded events. RTSP is designed to work with established protocols, such as RTP and HTTP.
SAP	Session Announcement Protocol. A protocol used to assist in the advertisement of multicast multimedia conferences and other multicast sessions, and to communicate the relevant session setup information to prospective participants.

Acronym or Term	Definition
SCCP	(1) signaling connection control part. Software that provides an OSI network layer service to its users. It does not support intermediate routing. (2) Skinny Client Control Protocol. The Cisco standard for real-time calls and conferencing over IP.
SCP	service control point. An element of an SS7-based Intelligent Network that performs various service functions, such as number translation, call setup and teardown, and so on.
SDLC	Synchronous Data Link Control. IBM Systems Network Architecture (SNA) data-link-layer communications protocol. SDLC is a bit-oriented, full-duplex serial protocol that has spawned numerous similar protocols, including HDLC. *See also* HDLC.
SDP	Session Description Protocol. A protocol used to describe multimedia sessions in order to enable session announcement, session invitation, and other forms of multimedia session initiation.
serial tunnel	*See* STUN.
service control point	*See* SCP.
service level agreement	*See* SLA.
service switching point	*See* SSP.
Session Announcement Protocol	*See* SAP.
Session Description Protocol	*See* SDP.
session initiation protocol	*See* SIP.
SF	Super Frame. Common framing type used on T1 circuits. SF consists of 12 frames of 193 bits each, with the 193rd bit providing framing synchronization. SF is superseded by ESF but is still widely used. Also called D4 framing. *See also* ESF.
signal ground	Refers to the common electrical reference point of a circuit.
signaling connection control part	*See* SCCP.
Signaling System 7	*See* SS7.
signal/noise ratio	*See* SNR.
signal transfer point	*See* STP.
SIMMs	single in-line memory modules. A small circuit board that holds a number of memory chips.
Simple Mail Transfer Protocol	*See* SMTP.
single in-line memory modules	*See* SIMMs.
SIP	session initiation protocol. Protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.
Skinny Client Control Protocol	*See* SCCP.
SLA	service level agreement. An agreement between the ISP and the client that guarantees a certain level of data transmission over the network.
small office/home office	*See* SOHO.

Acronym or Term	Definition
SMDS	Switched Multimegabit Data Service. High-speed, packet-switched, datagram-based WAN networking technology offered by the telephone companies.
SMTP	Simple Mail Transfer Protocol. The standard Internet protocol providing e-mail services.
SNR	signal/noise ratio. A measure of transmission quality. The ratio of good or usable data (signal) to bad or undesired (noise) on a line, expressed in decibels (dB).
SOHO	small office/home office. Networking solutions and access technologies for offices that are not directly connected to large corporate networks.
spanning tree (n) spanning-tree (adj)	Loop-free subset of a network topology.
SQL	Structured Query Language. International standard language for defining and accessing relational databases.
SRST	Survivable Remote Site Telephony. 1. A feature of any VoIP network that provides backup in case of network component failures. 2. A feature on some Cisco routers that uses the Skinny protocol to provide call-handling support for the local IP Phones if the WAN connection to the Cisco CallManager fails.
SS7	Signaling System 7. Standard CCS system used with BISDN and ISDN. Developed by Bellcore. *See also* CCS.
SSP	service switching point. Element of an SS7-based Intelligent Network that performs call origination, termination, or tandem switching.
STP	signal transfer point. Element of an SS7-based Intelligent Network that performs routing of the SS7 signaling.
Structured Query Language	*See* SQL.
STUN	serial tunnel. Router feature allowing two SDLC- or HDLC-compliant devices to connect to one another through an arbitrary multiprotocol topology (using Cisco routers) rather than through a direct serial link.
Super Frame	*See* SF.
Survivable Remote Site Telephony	*See* SRST.
SVC	switched virtual circuit. Virtual circuit that is dynamically established on demand and is torn down when transmission is complete. SVCs are used in situations where data transmission is sporadic.
Switched Multimegabit Data Service	*See* SMDS.
switched virtual circuit	*See* SVC.
Synchronous Data Link Control	*See* SDLC.
T1/E1	T1: The standard digital multiplexed 24-channel voice/data digital span line. Used predominantly in North America. Operates at a data rate of 1.544 Mbps. Digital WAN carrier facility. T1 transmits DS-1 formatted data through the telephone-switching network, using AMI or B8ZS coding. *See also* AMI and B8ZS. E1: Wide-area digital transmission scheme used throughout the world that carries data at a rate of 2.048 Mbps. E1 lines can be leased for private use from common carriers.
tabletop	A conference telephone used on a conference room table.
TAPI	Telephony Application Programming Interface. A call control model developed by Microsoft and Intel.

Acronym or Term	Definition
TCAP	transaction capabilities application part. SS7 protocol layer that helps exchange noncircuit-related data between applications.
T-CCS	Transparent Common Channel Signaling. Feature that allows the connection of two PBXs with digital interfaces that use a proprietary or unsupported CCS protocol without the need for interpretation of CCS signaling for call processing. T1/E1 traffic is transported transparently through the data network, and the feature preserves proprietary signaling. From the PBX standpoint, this is accomplished through a point-to-point connection. Calls from the PBXs are not routed, but follow a preconfigured route to the destination.
TDM	time-division multiplexing. Technique in which information from multiple channels can be allocated bandwidth on a single wire based on preassigned timeslots. Bandwidth is allocated to each channel regardless of whether the station has data to transmit.
Telephony Application Programming Interface	*See* TAPI.
time-division multiplexing	*See* TDM.
time stamp (n) time-stamp (v, adj)	A field in certain FastPacket formats that indicates the amount of time the packet has spent waiting in queues during the transmission between its source and destination nodes. Used to control the delay experienced by the packet.
transaction capabilities application part	*See* TCAP.
Transparent Common Channel Signaling	*See* T-CCS.
Triple Data Encryption Standard	*See* 3DES.
two-wire	One of two distinct types of audio interface (two-wire or four-wire). With the two-wire implementation, full-duplex audio signals are transmitted over a single pair, which consists of tip (T) and ring (R) leads.
UAC	user agent client. A client application that initiates the SIP request.
UAS	user agent server. A server application that contacts the user when a SIP request is received, and then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.
UDP	User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.
U interface	The ISDN interface between the telco and the user, also known as the local loop.
user agent client	*See* UAC.
user agent server	*See* UAS.
User Datagram Protocol	*See* UDP.
VAD	voice activity detection. Used to statistically save bandwidth by not sending packets in the absence of speech. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection uses much less bandwidth.
variable bit rate	*See* VBR.
variable bit rate-nonreal time	*See* VBR-NRT.
variable bit rate real-time	*See* VBR-RT.
VBR	variable bit rate. QoS class defined by the ATM Forum for ATM networks. VBR is subdivided into an RT class and NRT class.

Acronym or Term	Definition
VBR-NRT	variable bit rate-nonreal time. Subclass of VBR. Used for connections in which there is no fixed timing relationship between samples but that still need a guaranteed QoS.
VBR-RT	variable bit rate-real time. Subclass of VBR. Used for connections in which there is a fixed timing relationship between samples.
V card	An electronic business card. V cards carry information such as name, telephone numbers, mail addresses, e-mail addresses, and URLs.
VIC	voice interface card. A Cisco interface card used to connect the system to either the PSTN or to a PBX. *See also* PBX *and* PSTN.
videoconference	A meeting between people in different locations, using audio and video. The simplest type of videoconference can involve transmission of static images between two locations; the most complex videoconferences can use full-motion video and high-quality audio between multiple locations.
video on demand	*See* VoD.
Virtual Private Network	*See* VPN.
VoATM	Voice over ATM. A technology that enables a router to carry voice traffic (for example, telephone calls and faxes) over an ATM network. When sending voice traffic over ATM, the voice traffic is encapsulated using a special AAL5 encapsulation for multiplexed voice.
VoD	video on demand. System using video compression to supply video programs to viewers when requested via ISDN or cable.
VoFR	Voice over Frame Relay. A technology that enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network.
voice activity detection	*See* VAD.
voice interface card	*See* VIC.
Voice over ATM	*See* VoATM.
Voice over Frame Relay	*See* VoFR.
Voice over IP	*See* VoIP.
Voice over X	*See* VoX.
VoIP	Voice over IP. The capability to carry normal telephony-style voice over an IP-based internet with POTS-like functionality, reliability, and voice quality. VoIP enables a router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the DSP segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets. These voice packets are transported using a variety of signaling protocols.
VoIP over Frame Relay	*See* VoIPovFR.
VoIPovFR	VoIP over Frame Relay. Provides VoIP application interworking over an existing Frame Relay network. Can be used over point-to-point leased lines or a Frame Relay circuit; it does not require a full-fledged Frame Relay network or service.
VoX	Voice over X. Term that refers to nontraditional methods of carrying voice.
VPN	Virtual Private Network. Enables IP traffic to travel securely over a public TCP/IP network by encrypting all traffic from one network to another. A VPN uses "tunneling" to encrypt all information at the IP level.
WAN interface card	*See* WIC.
weighted fair queuing	*See* WFQ.

Acronym or Term	Definition
WFQ	weighted fair queuing. Congestion management algorithm that identifies conversations (in the form of traffic streams), separates packets that belong to each conversation, and ensures that capacity is shared fairly between these individual conversations. WFQ is an automatic way of stabilizing network behavior during congestion and results in increased performance and reduced retransmission.
WIC	WAN interface card. A Cisco interface card that connects the system to the WAN link service provider.
Wink Start	A method of E&M signaling. When the signaling leads indicate a change to an off-hook state, the other side must send a momentary *wink* (on-hook to off-hook to on-hook transition) on the correct signaling lead before the call signaling information can be sent by the sending side. After the call signaling information is received, the side that sent the wink goes off hook again when the subscriber answers and stays that way for the duration of the call.

Cisco VoIP Applications

Overview

This appendix describes various implementations and applications of Voice over IP (VoIP) networks.

Relevance

Implementation of creative and cost-saving applications is the cornerstone of VoIP networks. Network administrators who have knowledge of various VoIP applications and who can successfully cut corporate costs bring added value to a company.

Objectives

Upon completing this appendix, you will be able to describe new and evolving applications and identify cost savings ideas that capitalize on the convergence of voice and data in an IP internetwork. This includes being able to meet these objectives:

- Describe the types of networks that implement Hoot and Holler networks and list two reasons businesses may choose this method of networking
- Describe the types of networks that implement Cisco CallManager and list two reasons businesses may choose this method of VoIP
- Explain how enterprise customers can avoid paying toll charges when making interoffice calls
- List the four components a hospitality enterprise can provide and explain the value it can bring to their business
- Explain how an IP Centrex can replace a PBX on a customer site and describe the services it provides
- List two candidates for multitenant applications and explain why building owners are beginning to build this type of network
- Describe the features and benefits of Cisco Voice Infrastructure and Applications, when using prepaid calling card services in a packet telephony network
- Describe the purpose of a computer telephony integration system and describe how it is accessed

- Describe how participants in a collaborative computing environment are connected and the benefit of this type of connection
- Explain how voice-enabled web applications work and why they are an alternative to the Internet
- Explain how contact centers are changing to meet the needs of customers
- Describe how Cisco's brand of unified messaging system is used in an IP environment

Learner Skills and Knowledge

To benefit fully from this appendix, you must have these prerequisite skills and knowledge:

- Knowledge of VoIP networks
- Knowledge of telephony applications

Outline

The outline lists the topics included in this appendix.

Outline

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- Overview
- Hoot and Holler
- Cisco CallManager
- Toll Bypass
- Hospitality
- IP Centrex
- Multitenant
- Prepaid Calling Card
- Computer Telephony Integration
- Collaborative Computing
- Voice-Enabled Web Applications
- Contact Centers
- Unified Messaging
- Summary

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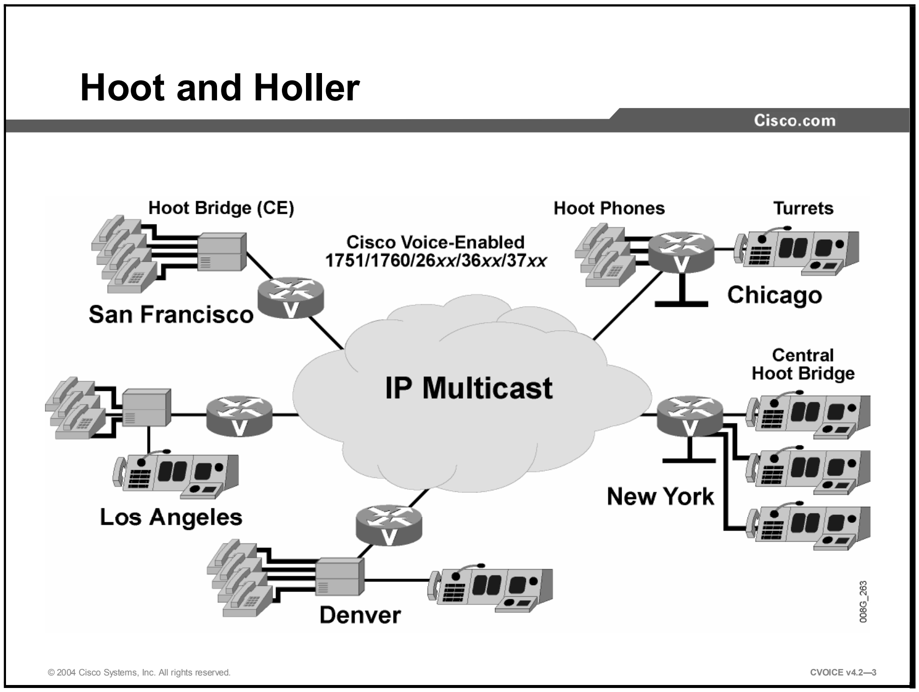
Hoot and Holler

This topic describes the Hoot and Holler implementations and applications.



A Hoot and Holler network (also known as a Junkyard Circuit, Squawk Box System, Holler Down Circuit, or Shout Down Circuit) provides always-on multiuser conferences without requiring users to dial into a conference bridge. This type of network was devised more than 50 years ago when local concentrations of small, specialized businesses needed to communicate common, time-critical information. Junkyard operators up and down the East Coast of the United States were among the first users of these networks. They began to install their own telephone wires, speakers (called "squawk boxes"), and microphones to share information with other locations about parts that their customers needed. These networks functioned as crude, do-it-yourself, business-to-business intercom systems.

Hoot and Holler broadcast audio network systems have evolved into the specialized leased-line networks of today. Financial and brokerage firms use these networks to trade stocks and currency futures and provide time-critical information, such as market updates and morning reports. Users of various forms of Hoot and Holler networks include brokerages, news agencies, publishers, government and municipal emergency response agencies, weather bureaus, transportation providers, utility operators, manufacturers, collectibles dealers, talent agencies, airlines, and nationwide salvage yard organizations.

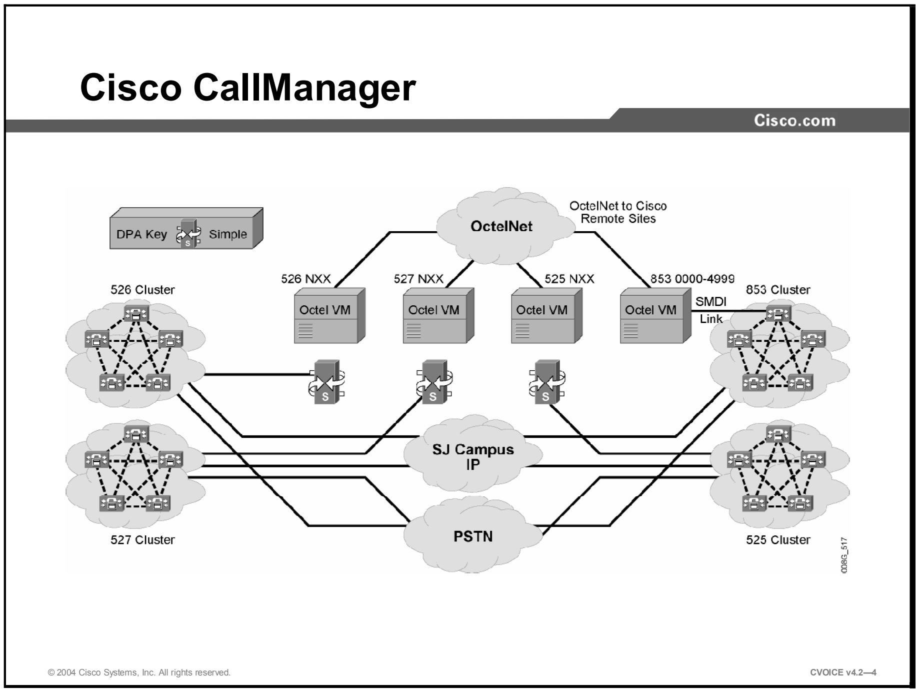
Hoot and Holler over IP transports Hoot and Holler voice traffic over traditional data networking equipment on an existing enterprise multiservice network. Hoot and Holler is another packet-based application that runs on a corporate multiservice network. Hoot and Holler enables businesses to eliminate expensive, dedicated leased lines, while protecting investments in existing Hoot and Holler equipment; for example, turrets, bridges, and four-wire telephones. In addition to eliminating the leased lines, running Hoot and Holler traffic over an IP network allows businesses to utilize bandwidth more efficiently. When bandwidth is not being used for Hoot and Holler traffic, it can be made available for data.

Hoot and Holler requires that IP multicast be active on the routers that support the Hoot and Holler circuit. The connections are configured using special dial peers configured with "session protocol multicast."

Reference	For more information on multicast services, access the Cisco.com website at: http://www.cisco.com/ipmulticast

Cisco CallManager

This topic describes Cisco CallManager implementations and applications.



Cisco IP Communications—a comprehensive system of powerful, enterprise-class solutions including IP telephony, unified communications, IP video and audio conferencing, and customer contact—helps organizations realize business gains by improving operational efficiencies, increasing organizational productivity, and enhancing customer satisfaction. Cisco CallManager, an integral component of the Cisco IP Communications system, is the software-based call-processing component of the Cisco enterprise IP telephony solution; it is enabled by Cisco Architecture for Voice, Video and Integrated Data (AVVID).

Example: Cisco CallManager

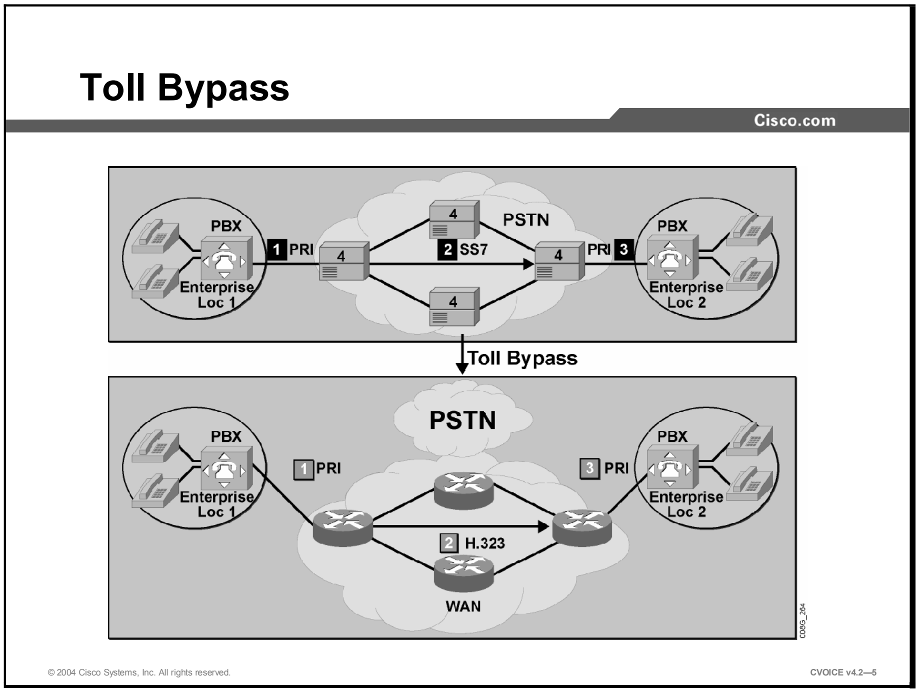
Cisco CallManager software extends enterprise telephony features and capabilities to packet telephony network devices such as IP Phones, media processing devices, VoIP gateways, and multimedia applications. Additional data, voice, and video services such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems interact with the IP telephony solution through Cisco CallManager open telephony application programming interfaces (APIs). Cisco CallManager is installed on the Cisco Media Convergence Servers (MCSs) and selected third-party servers. Cisco CallManager software is shipped with a suite of integrated voice applications and utilities, including the Cisco CallManager Attendant Console—a software-only manual attendant console; a software-only ad-hoc conferencing application; the Bulk Administration Tool (BAT); the CDR Analysis and Reporting (CAR) tool; the Admin Serviceability Tool (AST); a simple, low-density Cisco CallManager Auto Attendant (CM-AA); the Tool for Auto-Registered Phones Support (TAPS); and the IP Manager Assistant (IPMA) application.

Cisco CallManager provides a scalable, distributable, and highly available enterprise IP telephony call-processing solution. Multiple Cisco CallManager servers are clustered and managed as a single entity. Clustering multiple call-processing servers on an IP network is a unique capability in the industry and highlights the leading architecture provided by Cisco AVVID. Cisco CallManager clustering yields scalability of from 1 to 30,000 IP Phones per cluster, load balancing, and call-processing service redundancy. By interlinking multiple clusters, system capacity can be increased up to one million users in a system of 100 or more sites. Clustering aggregates the power of multiple, distributed Cisco CallManagers, enhancing the scalability and accessibility of the servers to phones, gateways, and applications. Triple call-processing server redundancy improves overall system availability.

The benefit of this distributed architecture is improved system availability, load balancing, and scalability. Call Admission Control (CAC) ensures that voice quality of service (QoS) is maintained across constricted WAN links, and automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available. A web-based interface to the configuration database enables remote device and system configuration. HTML-based online help is available for users and administrators.

Toll Bypass

This topic describes the toll-bypass implementations and applications.



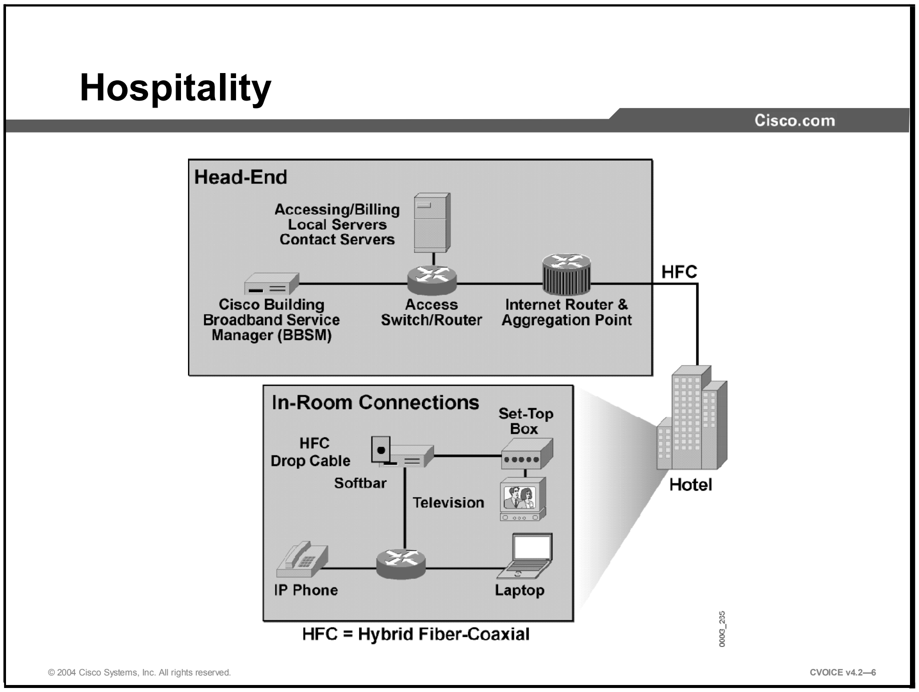
Toll bypass allows customers to bypass the PSTN. The PSTN consists of the tandem time-division multiplexing (TDM)-based switches using the packet network for long-distance (or toll) voice calls. Enterprise customers who typically depend on the PSTN for their interoffice voice traffic avoid toll charges by using the packet network with the Cisco routers that serve as the edge voice gateways. Toll bypass allows some Internet service providers (ISPs) to offer residential customers free or very low-cost long-distance voice calls by routing the calls over the packet network. The figure shows a typical toll-bypass application.

Example: Toll Bypass

In the figure, traffic from the enterprise PBX enters the Cisco routers that serve as edge voice gateways. The edge voice gateways, in turn, route the call over the IP network using the H.323 protocol. As shown, the enterprise customers avoid the TDM-based toll switches for their interoffice voice traffic and rely on the packet network.

Hospitality

This topic describes the hospitality implementations and applications.



Hospitality enterprises (for example, hotels, airports, and convention centers) host guests who demand high-speed connections to the Internet and access to telephony services. Enterprises that are used by a large number of travelers spend money to support LAN-like performance and extend high-speed telecommuting to corporate users.

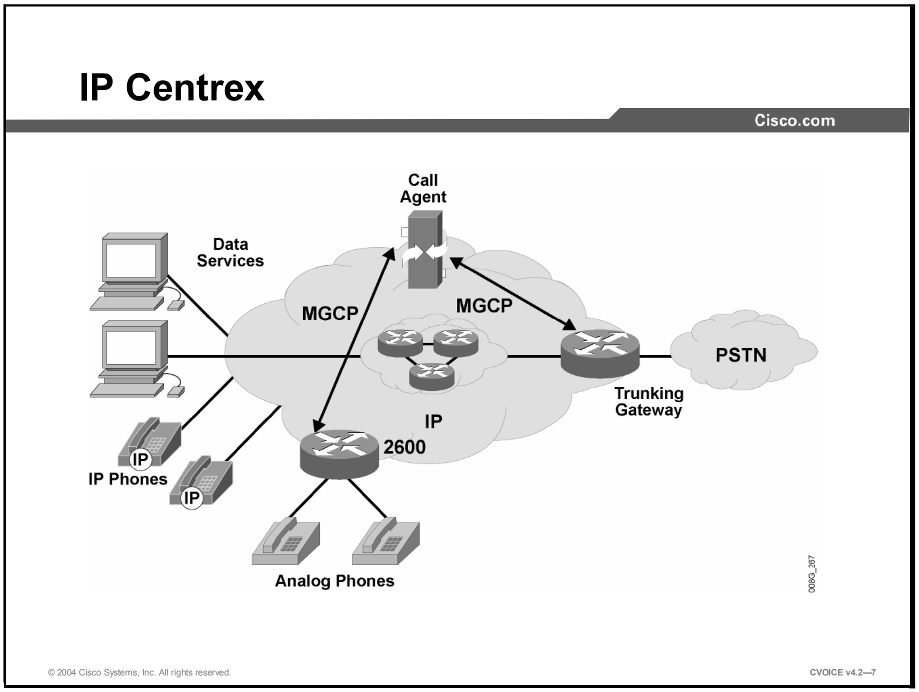
Hospitality providers build networks that offer high-speed Internet services in a flexible, affordable, and transparent manner. Applications include fast Internet access and high-volume VoIP solutions. In hotels, one building or the entire hotel campus is wired with a single broadband access line to supply voice, video, and Internet applications to guest rooms. The hotel can take advantage of high-volume, long-distance discounts from their provider while realizing revenue from direct long-distance dialing by hotel guests. A hospitality environment can deploy the following components:

- **Cisco Building Broadband Service Manager (BBSM):** This service-creation platform enables hotel property owners to create, market, and operate broadband access services. BBSM provides plug-and-play access; customizable portals to restaurants, wineries, and retail shops; and support for multiple authentication and billing options.
- **Cisco IP Phone:** This system provides hotel guests with added conveniences, such as telephone-based applications that may include concierge services, linked directories to local attractions, and automatic speed-dial setup. For groups taking advantage of meeting services, the Cisco IP Phones can provide personalized group directories, meeting agenda and room locations, and broadcast alert capabilities.

- **Cisco Content Transformation Engine (CTE):** The CTE is an appliance-based solution that optimizes the delivery of web content to a variety of wireless and wired devices, such as cell phones, personal digital assistants (PDAs), and IP Phones. Using the CTE with Cisco IP telephones in every room, guests have quick and easy access to hotel information (room service, in-house events) and third-party information (airline schedules, stock quotes, and weather information).

IP Centrex

This topic describes the IP Centrex implementations and applications.



Centrex service is regarded as an "outsourcing" of telephony call services. Centrex does not maintain a PBX on customer premises. Instead, Centrex service removes the PBX function from the customer premises, provides a Centrex trunk to the customer, and provides the telephony services over the trunk. Typically, the Centrex trunk is arranged as a TDM channel associated signaling (CAS) circuit or as an ISDN Q.931 connection. Customer billing for this service is similar to the billing for outsourcing services.

IP Centrex performs the same job as a PBX but delivers the service over a packet network instead of a circuit-switched network. Service is accessed via an IP network and delivered to customers in a private or multitenant installation. The call control functions that Centrex delivers include:

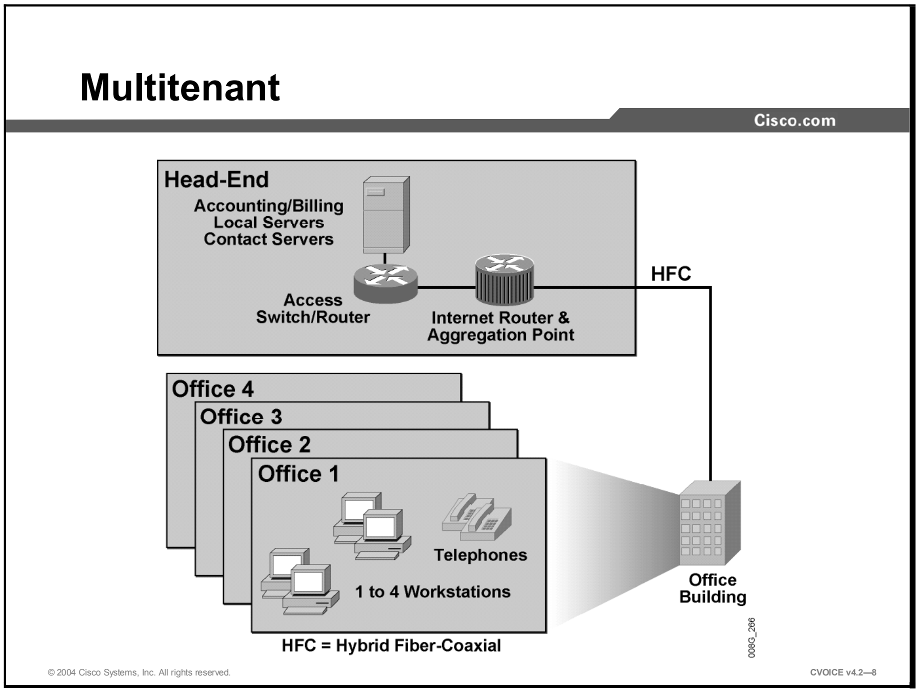
- Dial tone
- Interpretation of dialed digits
- Determination of called party status
- Call status return to caller, such as busy and ringback
- Call to voice-mail reroute, if applicable
- Billing services

Example: IP Centrex

Cisco delivers IP Centrex through the use of a call agent, such as the Cisco BTS 10200 Softswitch. The figure illustrates the use of a call agent for the Centrex services that are delivered to telephones with the trunking gateway serving as a path to the PSTN.

Multitenant

This topic describes multitenant implementations and applications.

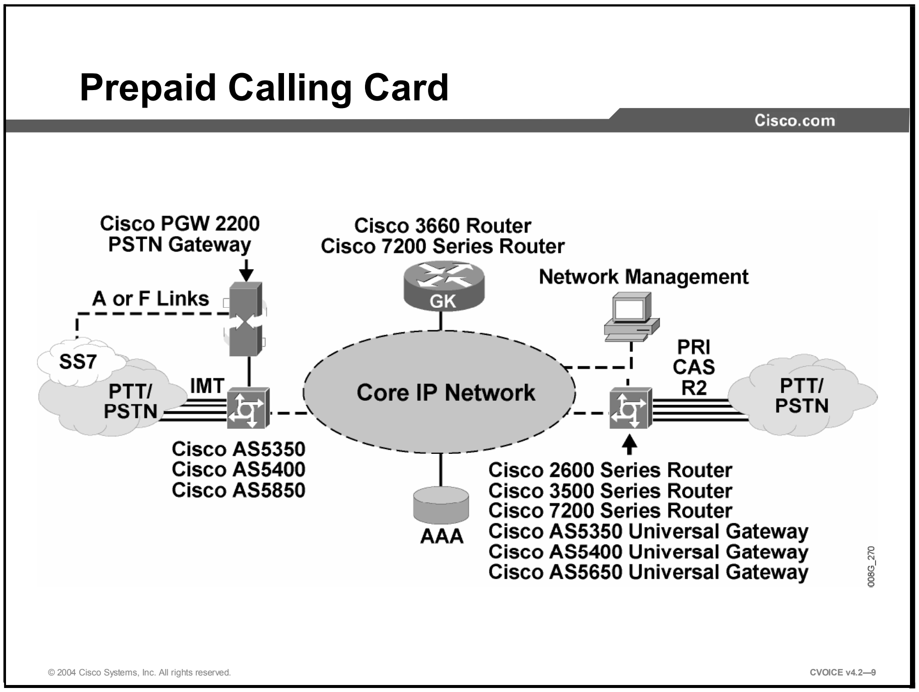


Multitenant applications allow building owners to deploy low-cost services, such as VoIP, cable television, and IP data services, to tenants in a common campus or building. Candidates for multitenant applications include the following:

- **Multi Dwelling Units (MDUs):** MDUs consist of high-rise and garden-style apartments, townhouses, and condominiums. Apartment renters and owners are now demanding high-speed Internet connections to home offices. Owners or MDU associations can attract new buyers or renters when they build an advanced cable IP infrastructure that offers secure, high-speed Internet access, cable television service, and VoIP access services.
- **Multi-Tenant Units (MTUs):** MTUs are commercial properties that house a number of small or medium-sized offices. These users can leverage the existence of a cable IP infrastructure to:
  - Use a high-speed cable broadband medium for improved internal communications, which includes LAN services
  - Develop businesses, attract new opportunities, and increase revenue streams through infrastructure advancements and services that support IP data and VoIP

Prepaid Calling Card

This topic describes the prepaid calling-card implementations and applications.



Prepaid and postpaid calling-card services represent one of the fastest-growing types of enhanced voice services. A variety of consumer segments have propelled the growth of these services, including students, business and leisure travelers, expatriates, and immigrants. They are especially popular among mobile telephone users as an alternative to the costly international rates of mobile operators. For carriers who want to realize more profit from a global long-distance network, prepaid and postpaid calling-card services represent an opportunity to improve margins, direct minutes to the network, and increase customer retention. For service providers that are currently offering prepaid and postpaid calling-card services over a switched-circuit network, Cisco packet telephony networks provide a more cost-effective alternative for network expansions or upgrades.

Packet voice technology offers a compelling alternative to the traditional TDM switched-circuit network. Packet telephony networks reduce the cost and time-to-market requirements associated with launching or expanding voice services, such as national and international transport, voice mail and unified communications, text-to-speech, speech recognition, and calling-card services. TDM-based services use a leased line and typically require a long-term financial commitment to that specific link. A TDM switch also represents a significant initial cash outlay and lengthy time period to achieve investment payback. The need to accelerate investment payback leads some providers to add fees for calling-card activation or connection, diminishing the service marketability in the process.

Cisco offers a feature-rich solution for prepaid and postpaid calling-card services that is deployed via packet voice technology. The Cisco Voice Infrastructure and Applications (VIA) solution includes key features and attributes such as the following:

- A telephony user interface similar to PSTN card services applications
- Cost-efficiency in equipment and bandwidth
- Card recharging
- Balance transfer
- Personal identification number (PIN) change
- Support for multiple languages

The Cisco VIA solution offers the following benefits:

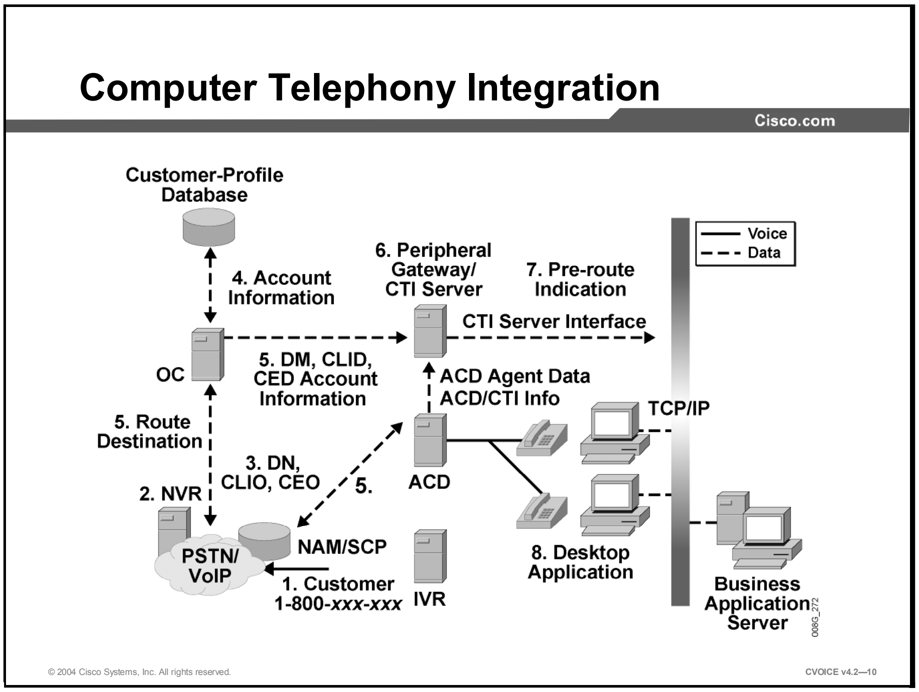
- Lower infrastructure and operating costs compared to other industry offerings
- Industry-leading voice quality, built-in reliability, and scalability
- Architectural and protocol flexibility
- Ability for service providers of any size and location to compete in the calling-card services market

Example: Prepaid Calling Card

In the illustration, Cisco AS5*xxx* Access Servers allow calling-card customers into the network from the PSTN. The authentication, authorization, and accounting (AAA) server verifies the customer account status. When the account status is authorized, the gatekeeper directs the call across the IP core network to the remote router or gateway, and connects to the PSTN. The service provider receives revenue for routing the call across the IP core, long distance.

Computer Telephony Integration

This topic describes implementation and application of computer telephony integration (CTI) systems.



CTI enables access to computer-processing functions while making, receiving, and managing telephone calls. CTI applications allow users to perform tasks such as retrieving customer information from a database provided by the caller ID. CTI applications also enable users to use the information captured by an interactive voice response (IVR) system to route a call to the appropriate customer service representative or to provide information to the individual receiving the call. The figure illustrates a CTI application that routes a customer call to a selected agent based on customer dual-tone multifrequency (DTMF) key input, database lookup, Caller ID, and customer profile information. Such applications are useful to businesses like financial institutions, consumer products, or commercial fulfillment warehouses.

The following is a partial list of Cisco CTI applications:

- **Cisco IP SoftPhone:** Cisco IP SoftPhone, a desktop application, turns your computer into a full-feature telephone with the added advantages of call tracking, desktop collaboration, and one-click dialing from online directories. You can also use Cisco IP SoftPhone in tandem with a Cisco IP Phone to place, receive, and control calls from your desktop PC. All features function in both modes of operation.
- **Cisco IP Auto Attendant:** The Cisco IP Auto Attendant application works with Cisco CallManager to receive calls on specific telephone extensions and allow callers to select extensions.
- **Cisco WebAttendant:** Cisco WebAttendant provides a GUI for controlling a Cisco IP Phone to perform attendant console functions.
- **Cisco Personal Assistant:** Cisco Personal Assistant or a virtual secretary can selectively handle incoming calls and help users place outgoing calls.

Example: CTI

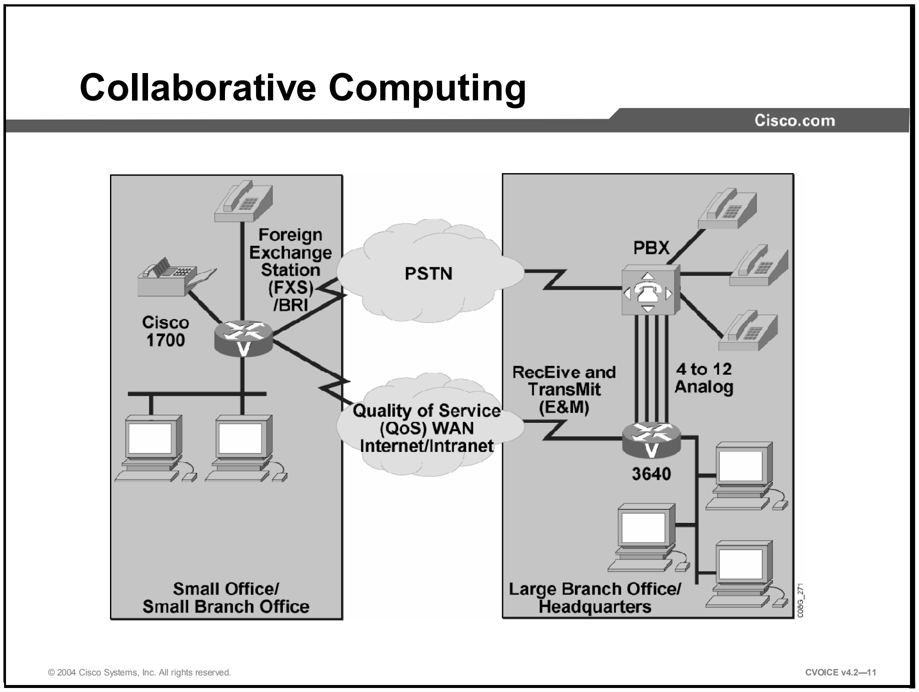
To illustrate how CTI works, consider a customer inquiry to a banking institution. The customer dials a toll-free telephone number from a home telephone. The agent who answers the call is in a pool of agents whose calls are delivered via an automatic call distributor (ACD). The preceding figure follows these steps:

1. The customer dials the toll-free number.
2. A Network Voice Response (NVR) system plays a script that collects caller-entered digits (CED), such as an account number.
3. The network sends a route request through an optical carrier (OC) interface to access the customer profile database.
4. The CED, dialed number (DN), and calling line ID (CLID) are referenced in the customer profile database.
5. A route destination is returned to the network applications management/service control point (NAM/SCP) and the DN, CLID, CED, and account information are forwarded to the automatic call distribution (ACD) system and peripheral gateway/CTI server.
6. The CTI server matches the selected agent from the ACD.
7. The CTI server sends a preroute indication across the CTI server interface to the TCP/IP network for pop-up delivery to the selected agent.
8. The TCP/IP network delivers the caller account information and CED information to the selected agent desktop.

Note The sample call data flow outlined above depicts an IVR in the carrier network. Alternatively, prompting may occur through an IVR at the premises or through a combination of network and premises-based IVRs.

Collaborative Computing

This topic describes collaborative computing implementations and applications.



Collaborative computing allows team members, on a shared project, to share resources and applications in real time, regardless of their physical location. The figure illustrates a simple VoIP internetwork that allows collaborative computing. The PSTN serves as a gateway to non-enterprise-connected participants.

At the heart of the collaborative computing solution is the IP internetwork. Participants connect to each other via a private IP network or the public Internet. Servers and applications are deployed across the network, including the client software on the participant desktops. Collaborative computing applications include the following:

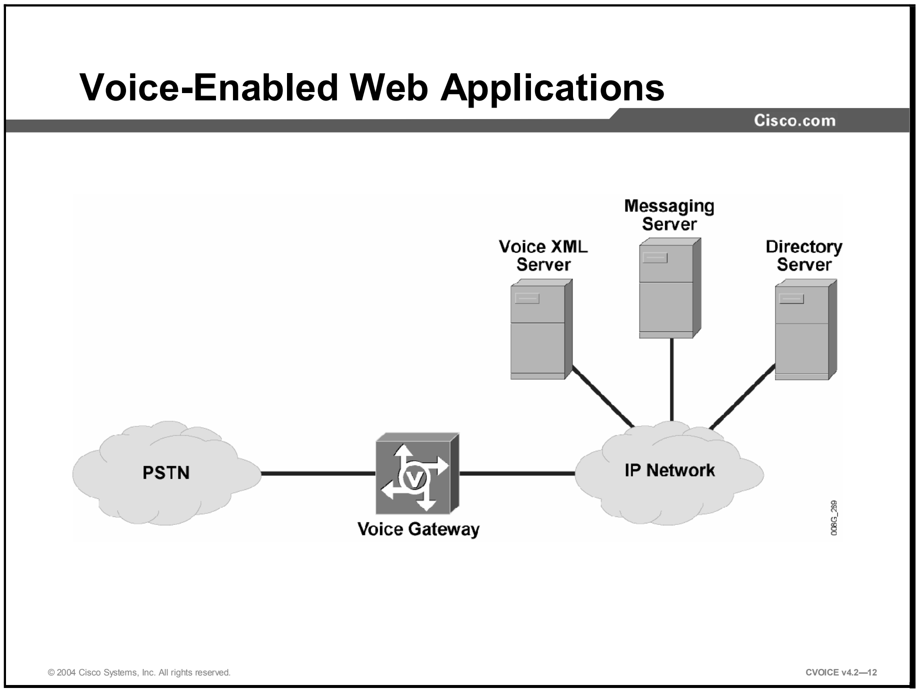
- Telephony and meeting applications, such as Microsoft NetMeeting
- Scheduling and collaboration software, such as IBM Lotus Notes
- Video streaming software
- FTP, TFTP, and peer-to-peer file-sharing applications
- IP Phones
- Whiteboard software

The goals of collaborative computing include:

- Reduced travel expenses
- Network transparency
- Shared calendar
- Real-time document sharing
- Real-time contact, such as telephony or videoconferencing

Voice-Enabled Web Applications

This topic describes the implementation and application of voice-enabled web applications.



Cisco AS5400 Series Universal Gateways can interpret Voice Extensible Markup Language (VoiceXML) documents. VoiceXML is an open-standard markup language that creates voice-enabled web browsers and IVR applications. While HTML enables users to retrieve data with a PC, VoiceXML enables subscribers to retrieve data with a telephone. The universal accessibility of the telephone and its ease of use make VoiceXML applications a powerful alternative to HTML for Internet access. The Cisco VoiceXML solution infrastructure takes advantage of Cisco AS5400 Series Universal Gateway domain specific part (DSP) resources, signaling, and media conversion capabilities to execute VoiceXML application logic at the edge of the network, on offloading servers, and within the network to support unified communications services.

Example: Voice-Enabled Web Applications

Consider the following scenario:

1. A baseball fan dials a number from the PSTN and is connected to a Cisco voice gateway that is configured as a VoiceXML-enabled gateway.

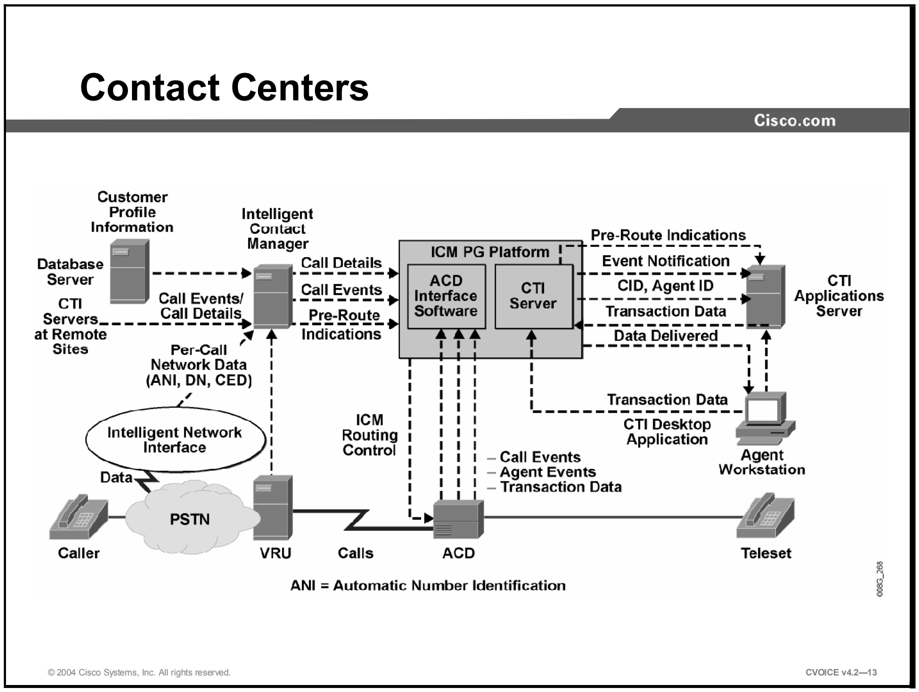
2. The Cisco voice gateway uses the caller ID information and associates it with the appropriate VoiceXML document that resides on a web server. This document provides baseball scores for the caller.

3. The voice gateway runs the VoiceXML document and responds to the caller by playing the appropriate audio content. The application might play a recorded prompt that asks the caller to press a specific DTMF key to hear a sports score, such as, "Press 2 to hear the results of the playoff game between Baltimore and New York."

4. Cisco IOS VoiceXML can transfer the caller to another party, such as customer service. For example, after playing the score, the application might prompt the caller with the message, "If you sign up for a one-year subscription to this service now, you will be entered into a drawing for two tickets to the next World Series. Press 5 to speak to one of our agents."

Contact Centers

This topic describes the contact center implementations and applications for VoIP.



Contact centers are the hubs of the customer service efforts of many growing businesses. Forward-thinking companies are integrating this key function with Internet technology to transform customer care into a powerful business-building force.

Firms such as catalog sellers, telemarketers, and computer helpdesks use traditional contact centers to manage large volumes of telephone calls and customer contacts. Contact center applications route incoming calls to sales and service agents who can respond to customer needs. Integrating this contact center activity with an Internet-based customer-relationship management (CRM) solution gives agents immediate access to customer purchase histories, order tracking capabilities, and other key information and tools. This enhanced information flow enables contact center staff to use customer interaction to build customer loyalty and retention.

Traditional contact center technology recognizes incoming contact requests (calls, e-mails, faxes, web requests, and so on) and routes them to available agents. These contact center technologies include recognition of customer telephone numbers, account numbers, or IDs. Customer data appears on the agent computer screen, ensuring that the agent can access the customer orders, account balance, and other crucial data. However, real-time collaboration between the customer and the agent is limited to the spoken word. Demonstrative services and offers of product are limited to verbal description, usually a script read by the agent.

Cisco IP Contact Center (IPCC) with Internet access allows contact center agents to respond to customer queries over a variety of channels, such as telephone, e-mail, web, and fax.

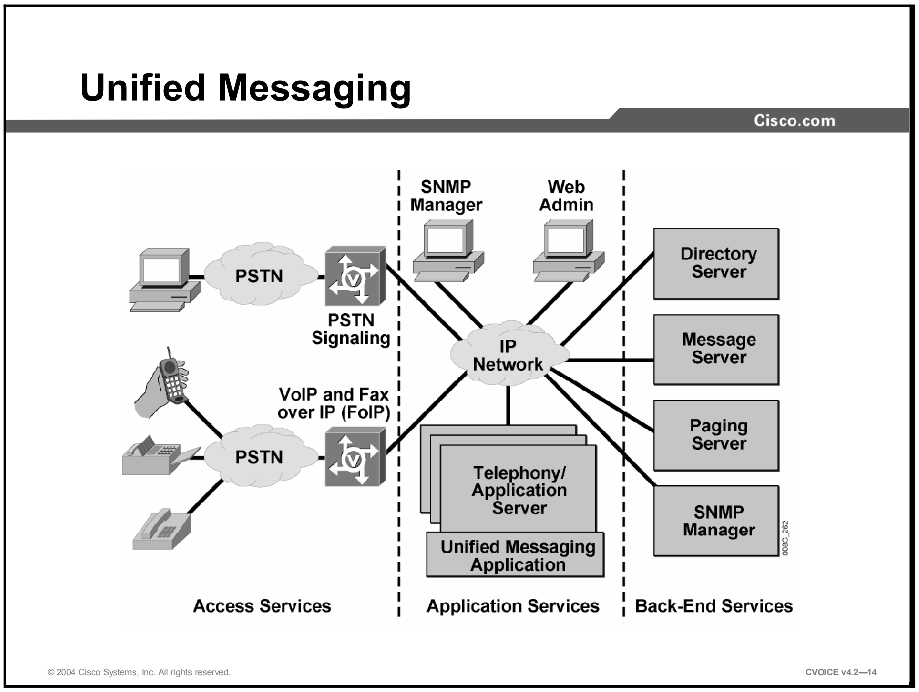
Example: Contact Centers

When a customer calls with questions about a new product, an agent can immediately send an e-mail message that includes product specifications and a link to a downloadable interactive demo. This scenario allows customer-service agents to take on sales and marketing roles, which helps the company roll out new initiatives and promotions quickly to targeted customers. Other ways to enhance real-time customer collaboration include fax-back services and web-page collaboration, where the customer and agent interact on the same web page to ensure, for example, that the color of a sweater is correct.

Contact centers use a range of telephone, computer, and network technologies, including VoIP. In the preceding figure, Cisco Intelligent Call Management (ICM) software is at the heart of the contact center application. The ICM uses CTI technology to deliver caller account information to the agent desktop while the agent receives the VoIP call. The location independence of the agents adds another benefit to this model. “Follow-the-sun” customer support programs allow around-the-clock customer service, regardless of the location of the agents.

Unified Messaging

This topic describes the unified messaging system implementations and applications.



Cisco Unity is designed for an IP environment and complements the full range of IP Communications solutions—Cisco CallManager, Cisco Personal Assistant, and Cisco IP Contact Center. Cisco Unity provides advanced capabilities that unify data and voice. Cisco AVVID enables Cisco Unity to provide a solid foundation to roll out future convergence-based communications services. It is less expensive to use IP for a comprehensive communications solution deployment because it is a single network for both voice and data.

Cisco Unity leverages existing communications infrastructure investments by integrating with leading legacy PBXs and interoperating with legacy voice-mail systems. Cisco Unity supports legacy PBX systems and Cisco CallManager—even simultaneously—paving the way for a cost-effective migration to full IP telephony. Cisco Unity has an optional Audio Messaging Interchange Specification analog (AMIS-a) networking module that allows message interchange between disparate voice messaging systems that support this industry-standard messaging protocol. Cisco Unity Bridge enables advanced message interchange functionality with Avaya and Octel voice messaging systems. With AMIS-a and Cisco Unity Bridge, customers that deploy Cisco Unity can continue to use their legacy messaging systems, to ensure a smooth transition.

Because Cisco Unity shares the same directory as the exchange network, users can make subscriber moves, adds, and changes from one place, eliminating redundant tasks. Studies show that the average cost of a typical system move, addition, or change to a user account is between $75 and $100. Eliminating duplicate administration for separate voice and e-mail systems can quickly pay for the entire system. In addition, because all messages are housed in the same message store, backup costs are cut in half.

Summary

This topic summarizes the key points discussed in this appendix.

Summary

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- Networks that require time-critical information use Hoot and Holler networks to eliminate expensive dedicated leased lines and to use bandwidth more effectively.
- Cisco CallManager is a centralized call agent for managing growing VoIP networks.
- The toll-bypass application allows users to bypass the PSTN and avoid paying toll charges by routing calls over a packet network.
- BBSM, Cisco ICS, Cisco IP Phone, and CTE are components that a hospitality environment can deploy to attract customers and increase revenue.
- IP Centrex replaces the PBX function on a customer site by using a Centrex trunk with TDM CAS to provide telephony services.

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Summary (Cont.)

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- Multitenant applications allow building owners to deploy low-cost technology services to MDUs and MTUs in a common campus or building.
- Cisco VIA provides a more cost-effective and flexible solution to prepaid calling-card services for packet telephony networks.
- CTI applications allow users to perform tasks and use information stored in a database using a caller ID or IVR system.
- Participants in a collaborative computing environment are connected by an IP network or the Internet and have the ability to share resources and applications in real time regardless of their location.

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Summary (Cont.)

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- The universal accessibility of the telephone makes voice-enabled web applications a powerful alternative to HTML and the Internet, because subscribers can retrieve data with a telephone using VoiceXML.
- Businesses are integrating call centers with CRM solutions to give agents immediate access to customer information and roll out new initiatives and promotions to target audiences.
- Cisco Unity leverages existing communication infrastructure investments by integrating with leading legacy PBXs and interoperating legacy voice-mail systems.

 CVOICE v4.2—17

References

For additional information, refer to these resources:

- *Multicast Hoot 'n' Holler*
  http://www.cisco.com/warp/public/cc/so/neso/vvda/hthllr/hhoip_wp.htm
- *MGCP Basic CLASS and Operator Services*
  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t2/ft mgcpgr.htm
- *Service Overview Prepaid and Postpaid Calling Card Services*
  http://www.cisco.com/warp/public/cc/so/neso/voso/ns68/ppccs_ov.htm
- *Cisco VoiceXML Solution Infrastructure*
  http://www.cisco.com/univercd/cc/td/doc/product/access/sc/rel7/soln/vxml_uc/vxml_rn.htm

CVOICE

Cisco Voice over IP

Version 4.2

Lab Guide

Text Part Number: 97-1871-01

Lab Guide

Overview

Use the exercises here to complete the lab exercises for this course. The solutions information is found in the Lab Exercise Answer Key.

Outline

This Lab Guide includes these exercises:

- Lab Exercise 2-1: Lab Familiarity
- Lab Exercise 3-1: Voice Port Configuration
- Lab Exercise 4-1: POTS Dial Peers
- Lab Exercise 4-2: Special-Purpose Connections
- Lab Exercise 5-1: Basic VoIP
- Lab Exercise 6-1: VoIP with H.323
- Lab Exercise 6-2: VoIP with SIP
- Lab Exercise 6-3: VoIP with MGCP
- Lab Exercise 7-1: QoS
- Lab Pull-Out Resource

Lab Exercise 2-1: Lab Familiarity

Complete this lab exercise to practice what you learned in the related lesson.

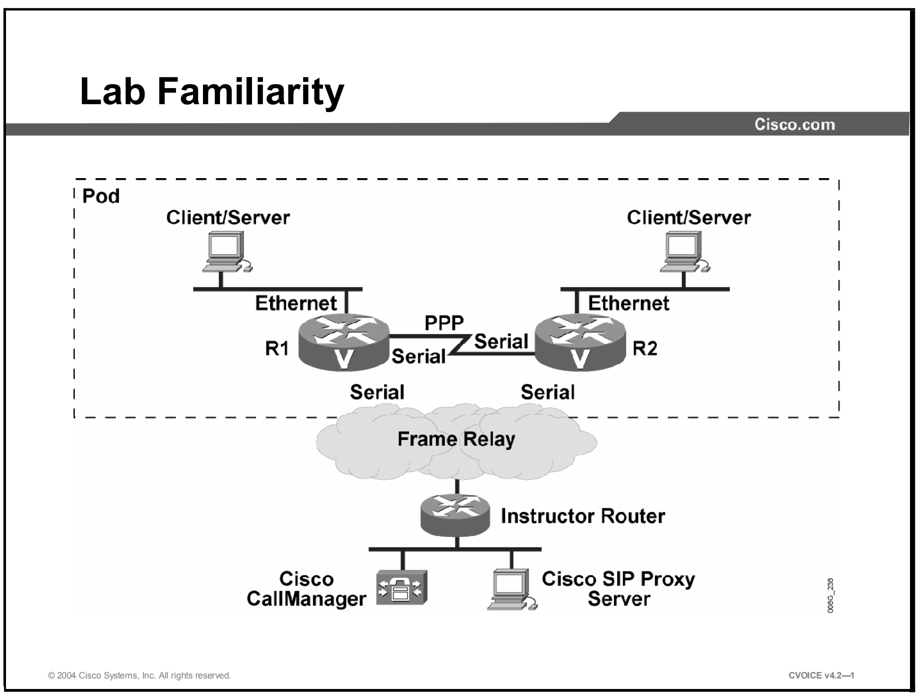
Exercise Objective

In this exercise, you will become familiar with the routers in your pod and your pod's relationship with the other pods. You will also personalize your pod's routers with host names and IP host tables and perform basic IP configuration to allow communication in and around the classroom. You will not be connecting any telephony hardware in this lab. After completing this exercise, you will be able to meet these objectives:

- List the data interfaces and voice ports on your routers
- Refer to the other routers in your pod, and elsewhere, by aliases
- Access the client/servers in all other pods in the classroom
- Access the classroom servers

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

Each pod contains two complete sets of hardware and a shared PSTN simulator. These are the resources and equipment required to complete this exercise:

- Two voice-enabled routers
- Two client/servers (laptop or desktop computers)
- Two Ethernet crossover cables
- Three serial crossover (DTE-DCE) cables

Command List

The commands used in this exercise are described in the table here.

Exercise Commands

Command	Description
clock rate *rate*	Configures the clock rate for the hardware connections on serial interfaces. This command is executed in interface configuration mode.
configure terminal	Enters configuration mode. This command is executed in privileged mode.
copy running-config startup-config	Copies current running configuration into NVRAM. This command is executed in privileged mode.
enable	Enters privileged mode. This command is executed in user mode.
enable secret *password*	Sets password to control access to privilege level. This command is executed in global configuration mode.
encapsulation *encap-type*	Sets the encapsulation type. This command is executed in interface configuration mode.
hostname *name*	Assigns a name to the router. This command is executed in global configuration mode.
interface [Ethernet,serial] *n*	Enters interface configuration mode for the specified interface. This command is executed in global configuration mode.
ip address *ip-address mask*	Configures the IP address and mask. This command is executed in interface configuration mode.
ip host *hostname ip-address*	Creates an IP host table entry assigning a name to an IP address. This command is executed in global configuration mode.
line vty 0 4	Enters vty line configuration mode for lines 0 through 4. This command is executed in global configuration mode.

Command	Description
`logging synchronous`	Redisplays the current line after a logging message has been displayed onscreen. Use for console or vty lines for cleaner display.
`login`	Enables password checking at login. This command is executed in line configuration mode.
`network` ***`network-address`***	Specifies a list of networks for the Enhanced Interior Gateway Routing Protocol (EIGRP) routing process. This command is executed in router configuration mode.
`no auto-summary`	Disables autosummarization and sends subprefix routing information across classful network boundaries. This command is executed in router configuration mode.
`no ip domain-lookup`	Disables DNS lookups. This command is executed in global configuration mode.
`password` ***`password`***	Sets a password. This command is executed in line configuration mode.
`ping (`***`ip-address or host name`*****`)`**	Tests the reachability of a device. When in user mode, it sends five default ping packets to test reachability. When in privileged mode, type in only the command **ping <enter>** to enter extended ping mode. The user can customize ping parameters for testing.
`router eigrp` ***`as-number`***	Configures the EIGRP routing process. This command is executed in global configuration mode.
`show controller t1`	Displays controller status and statistics. This command is executed in user mode.
`show ip interface brief`	Displays a summary of interface status and IP addresses. This command is executed in user mode.
`show ip route`	Displays the IP routing table. This command is executed in user mode.
`show version`	Displays the hardware configuration, software version, names, and sources of configuration files and boot images. This command is executed in user mode.
`show voice port summary`	Displays a summary of all voice ports. This command is executed in user mode.
`traceroute`	Discovers the routes that packets take to a remote destination, as well as where routing breaks down. When in user mode, it sends three packets for each hop. When in privileged mode, type in only the command **traceroute <enter>** to enter extended trace route mode. The user can customize trace route parameters for testing.

Job Aids

These job aids are available to help you complete the lab exercises:

IP Addressing Conventions

An IP addressing strategy has been adopted that allows the student to predict the address of any other device without knowing it beforehand. The table contains evidence of these conventions:

- Where x = pod number, y = host, addresses follow the format 10.x.10.y / 24 for R1 Ethernet; 10.x.20.y / 24 for R2 Ethernet; and 10.x.x.y / 24 for PPP link.
- The Frame Relay network is modeled as a fully meshed, nonbroadcast multiaccess (NBMA) network using network 192.168.1.0.

In addition, the instructor's Ethernet uses addresses in network 192.168.100.0.

IP Address Assignment

Pod Number	Device	Client	Ethernet	PPP	Frame Relay
1	R1	10.1.10.100	10.1.10. 1	10.1.1.1	192.168.1.11
	R2	10.1.20.100	10.1.20.1	10.1.1.2	192.168.1.12
2	R1	10.2.10.100	10.2.10.1	10.2.2.1	192.168.1.21
	R2	10.2.20.100	10.2.20.1	10.2.2.2	192.168.1.22
3	R1	10.3.10.100	10.3.10.1	10.3.3.1	192.168.1.31
	R2	10.3.20.100	10.3.20.1	10.3.3.2	192.168.1.32
4	R1	10.4.10.100	10.4.10.1	10.4.4.1	192.168.1.41
	R2	10.4.20.100	10.4.20.1	10.4.4.2	192.168.1.42

Host Name

Pod Number	Router	Host Name
1	R1	Pod1R1
	R2	Pod1R2
2	R1	Pod2R1
	R2	Pod2R2
3	R1	Pod3R1
	R2	Pod3R2
4	R1	Pod4R1
	R2	Pod4R2
5	R1	Pod5R1
	R2	Pod5R2
6	R1	Pod6R1
	R2	Pod6R2

Task 1: Initial Configuration

In this task, you will discover and configure the interfaces connecting to other equipment.

Exercise Procedure

Complete these steps:

Step 1 Using the client/server, connect to the console port of your router.

Step 2 Identify the Cisco IOS software version and its features.

List the number of available interfaces:

Ethernet ___________________________

Serial______________________________

Channelized T1/PRI __________________

Voice ________ Type ________________

Step 3 Use the appropriate commands to determine which Ethernet and serial interfaces are available and their interface numbers. Note the interface numbers below:

Ethernet ___________, _______________

Serial _____________,________________

T1 ________________, ________________

Step 4 Use the appropriate commands to determine which voice ports are available and their port numbers. Note the type and port numbers below:

Type ________________ Port number ________________

Type ________________ Port number ________________

Type ________________ Port number ________________

Type ________________ Port number ________________

Step 5 To prepare for configuration, fill in the diagram here. The instructor will tell you which interfaces to use when connecting to the Frame Relay switch or router. For the Frame Relay link, connect the DTE end of the cable to your interface and the DCE end to the Frame Relay switch or router. For the PPP connection between the voice-enabled routers in the pod, configure R1 to be the DTE side and R2 to be the DCE side of the connection.

R1	Interface Number	IP Address	DTE/DCE
PPP Serial			
Frame Serial			
Ethernet			N/A
Frame Switch		N/A	

Step 6 Configure your router host name with the appropriate name from the Host Name table. Configure an IP host table for other routers in the classroom. Configure the privileged password to be **san-fran**. Configure vty lines to allow password checking at login and the Telnet password to be **router**.

Step 7 Connect serial interfaces as per the lab diagram using DTE-DCE cables. Remember, clock rate must be configured on the DCE side. Set the link speed to be 72,000 bps.

Step 8 Configure both serial interfaces with correct encapsulations and IP addresses. See the IP Address Assignment table in the Job Aids section for address and encapsulation assignment. Configure one Ethernet interface with the correct IP address.

Step 9 Connect the client/server Ethernet to the router Ethernet port using a crossover cable. Configure the IP address on the client/server, pointing the gateway to the router Ethernet address. Test connectivity by pinging your router.

Step 10 Enable EIGRP routing for Autonomous System 100 and disable autosummarization. Enable EIGRP for all three of your interfaces.

Step 11 View the routing table and compare it to the lab diagram.

Step 12 Test IP reachability throughout the classroom by pinging or tracing from your client/server to other client/servers.

Step 13 Save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- See all other classroom subnets in your IP routing table.
- Trace or ping from your client/server to all other client/servers in the classroom.

Lab Exercise 3-1: Voice Port Configuration

Complete this lab exercise to practice what you learned in the related lesson.

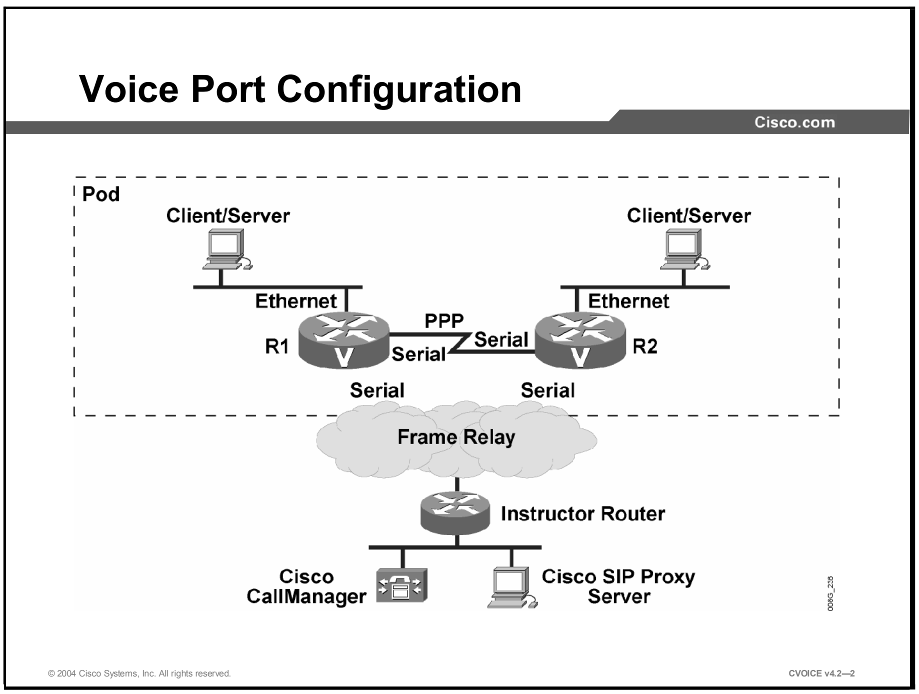
Exercise Objective

In this exercise, you will become familiar with existing analog voice ports. You will learn how to customize your analog ports by configuring various port parameters. You will also create and customize digital voice ports that connect to your PBX. After completing this exercise, you will be able to meet these objectives:

- Identify default voice port settings
- Customize and verify analog port operations
- Create, customize, and verify digital port operations

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- Two T1 crossover cables
- Two PBX devices
- Four telephones
- Four RJ-11 cables

Command List

The commands used in this exercise are described in the table here.

Voice Port Commands

Command	Description
clock source *source*	Specifies the clock source. This command is executed in controller configuration mode.
controller t1 *n*	Enters T1 controller configuration mode for the specified controller. This command is executed in global configuration mode.
cptone *country-code*	Sets the regional analog voice interface-related tone, ring, and cadence setting. This command is executed in voice-port configuration mode.
default *parameter*	Resets the value of the parameter to its default value. This command is executed in various configuration modes.
ds0-group *tag* timeslots *timeslot-list* type *signaling-type*	Specifies the DS0 timeslots that make up a logical voice port on a T1 or E1 controller and the signaling type for the DS0 group. This command is executed in controller configuration mode.
framing *sf|esf*	Specifies the framing type. This command is executed in controller configuration mode.
linecode *ami|b8zs*	Specifies the line code setting. This command is executed in controller configuration mode.
ring cadence define *pulse interval*	Sets the ring cadence for the FXS port. This defines how the telephone will ring. This command is executed in voice-port configuration mode.
show voice port (summary)	Views the voice port status and settings. Displays all default settings for each port. Specify a particular voice port to view only its settings. Use the summary option to view a summary table of the voice ports. This command is executed in user mode.
timeouts initial *secs*	Sets the number of seconds that the system will wait for the caller to input the first digit. This command is executed in voice-port configuration mode.
voice-port *port-number*	Enters voice-port configuration mode. This command is executed in global configuration mode.

Job Aids

These job aids are available to help you complete the lab exercise:

PBX Configuration

PBX parameters are as follows:

- T1 connection
- Timeslots 1–24
- FXO loop-start signaling
- Clock source is line
- ESF framing
- B8ZS line code

Task 1: Analog Voice Port Configuration

In this task, you will examine analog voice ports and configure voice port parameters.

Exercise Procedure

Complete these steps:

Step 1 Connect both phones to your voice-enabled router using RJ-11 cables.

Step 2 Verify that the connections are correct by lifting the handset on both telephones and listening for the dial tone. If the dial tone is not present, troubleshoot the problem. Make sure the router is powered on and the cable is firmly seated. If the problem persists, ask your instructor for help.

Step 3 Using **show** commands, identify available voice ports, their type, and their default settings. Please note that not all settings are applicable to all types of ports; for example, ring frequency and ring cadence only apply to FXS ports. Record the information below:

	Voice Port 1	Voice Port 2	Voice Port 3	Voice Port 4
Port type	__________	__________	__________	__________
Port number	__________	__________	__________	__________
Operational state	__________	__________	__________	__________
Echo cancellation	__________	__________	__________	__________
Echo cancel coverage	__________	__________	__________	__________
Initial timeout	__________	__________	__________	__________
Region tone	__________	__________	__________	__________
Signal type	__________	__________	__________	__________
Ring frequency	__________	__________	__________	__________
Ring cadence	__________	__________	__________	__________

Step 4 Since many companies have international offices, it is important to know how to configure the voice port to match the standard signaling of a country. For this step, assume you are configuring this router for Australia. On the FXS port that your telephone is connected to, configure the call progress tone setting for Australia. Notice that when you change the call progress tone setting, it automatically changes the ring cadence setting to match. Test the change by lifting the handset. You should hear a different dial tone.

Verify the changes with **show** commands and record the new settings below:

Region tone ________________________

Ring cadence ________________________

Once you have tested the tones for Australia, experiment with settings for other countries.

Step 5 What is the default initial timeout setting from Step 3? ___________
On the FXS port that your telephone is connected to, change the initial timeout value to 4 seconds. Lift the handset and listen for more than 4 seconds. Can you dial digits after the dial tone stops? __________Reset the initial timeout to the default.

Step 6 Since you will be working with two telephones all week, you may want one telephone to have a distinctive ring. Configure the ring cadence on your FXS port using the **define** option. You will be able to test this ring cadence in the next lab. What ring cadence did you define? ____________

Exercise Verification

You have completed this exercise when you attain these results:

- Configure and verify analog voice port parameters.

Task 2: Digital Voice Port Configuration

In this task, you will configure your T1 to connect to your PBX device. In the process of configuring the T1 for voice calls, a logical voice port will be created. You will be able to view this newly created digital voice port with the same commands as for analog ports.

Exercise Procedure

Complete these steps:

Step 1 Connect the PBX device to your router T1 interface with a crossover T1 cable. What port is your T1 cable plugged into? _________________

Step 2 Use the **show controller T1** command to view the default settings for framing, line code, and clock source. Note these settings below:

Framing _______________________

Line code ______________________

Clock source ___________________

Step 3 Configure your T1 controller to compliment the settings of the PBX as shown in the Job Aids section. Remember that since this is a back-to-back T1 connection, one side should have clock source line, and the other side should have clock source internal.

Verify that the settings match by checking that both controller LEDs are green. Use the **show controller T1** command to view status and new settings.

Step 4 When the T1 is functional, create digital voice ports using the **DS0-group** command. Once again, these settings must compliment those of the connected PBX. Check the required settings in the Job Aids section of this lab. Configure the DS0-group and use **show** commands to verify the newly created digital voice port. Fill in the information below:

How many voice ports were created? ______________

What is the voice port number? ___________________

How many channels were created? ________________

Which command would you use to view the voice port and the channels?
____

What is the current status of these channels? ______________________

Step 5 Save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- Verify the T1 connection to the PBX.
- Verify the existence of the newly created digital voice port.

Lab Exercise 4-1: POTS Dial Peers

Complete this lab exercise to practice what you learned in the related lesson.

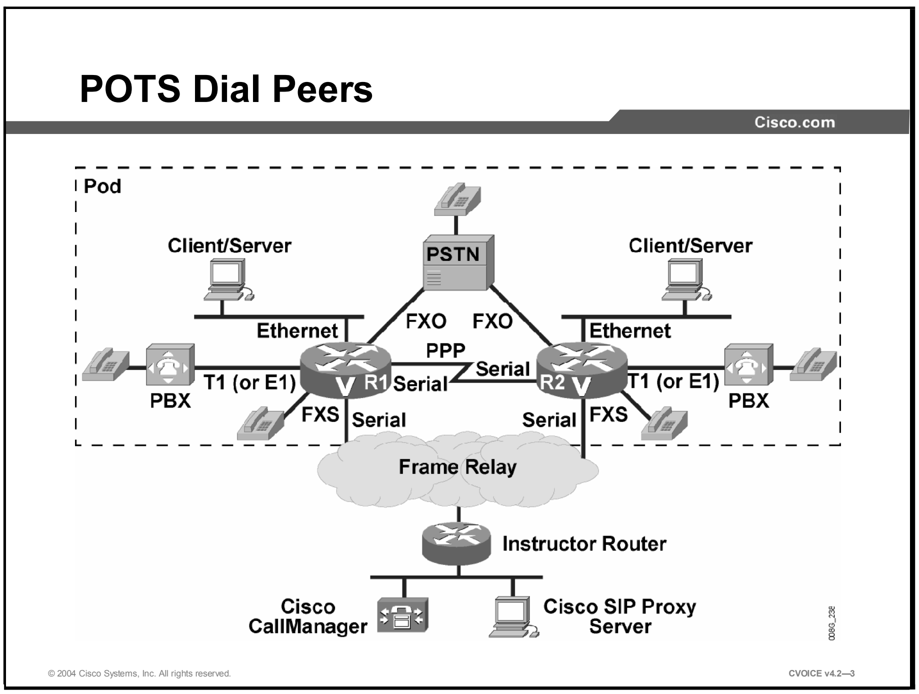
Exercise Objective

In this exercise, you will configure POTS dial peers to establish locally terminated calls, calls through a PBX, and calls to the PSTN. You will also experiment with two different configurations to control hunt capabilities. After completing this exercise, you will be able to meet these objectives:

- Configure dial peers for locally terminated calls, PBX calls, and PSTN calls
- Determine appropriate method of digit forwarding and manipulation
- Create hunt groups and determine hunting behavior

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment each pod requires to complete this exercise:

- One PSTN device
- One telephone
- One set RJ-11 cables

Command List

The commands used in this exercise are described in the table here.

Dial Peer Commands

Command	Description
debug vpm signal	Displays real-time voice port module signaling. This command is executed in privileged mode.
debug vtsp dsp	Displays digits as they are received by the voice port. This command is executed in privileged mode.
destination-pattern *string*	Configures a telephone number for this dial peer. This command is executed in dial-peer configuration mode.
dial-peer hunt *0-7*	Specifies a hunt selection order for dial peers. This command is executed in global configuration mode.
dial-peer voice *tag* pots	Enters dial-peer configuration mode. This command is executed in global configuration mode.
forward-digits	Specifies which digits to forward for voice calls. This command is executed in dial-peer configuration mode.
port *port-number*	Configures the port for this dial peer. This command is executed in dial-peer configuration mode.
preference *0-9*	Specifies the preferred order of a dial peer within a hunt group. This command is executed in dial-peer configuration mode.
show call active voice	Displays information on active calls. This command is executed in user mode.
show dial-peer voice (*tag*) | (summary)	Displays dial-peer configuration information. This command is executed in user mode. The **summary** option is available in privileged mode only.
show dialplan number *number*	Displays which dial peer is matched when a particular telephone number is dialed. This command is executed in privileged mode.
show voice call summary	Displays summary information on active calls. This command is executed in user mode.

Job Aids

These job aids are available to help you complete the lab exercise:

Dial Plan Conventions

As with IP addresses, the dial plan convention allows a student to anticipate the number of any telephone in the classroom. Again, for convenience, a table has been provided. The highlights of the strategy include:

- The classroom uses a four-digit dial plan.
- The first digit identifies the pod number (1 to 6).
- The second and third digits identify the device (PSTN=00, R1=10, PBX1=15, R2=20, PBX2=25).
- The fourth digit identifies the telephone.

Classroom Dial Plan

Pod Number		1	2	3	4	5	6
R1		1101 1102	2101 2102	3101 3102	4101 4102	5101 5102	6101 6102
PBX1		1151	2151	3151	4151	5151	6151
R2		1201 1202	2201 2202	3201 3202	4201 4202	5201 5202	6201 6202
PBX2		1251	2251	3251	4251	5251	6251
PSTN	Port 1	555-1001	555-2001	555-3001	555-4001	555-5001	555-6001
	Port 2	555-1002	555-2002	555-3002	555-4002	555-5002	555-6002
	Port 3	555-1003	555-2003	555-3003	555-4003	555-5003	555-6003

Task 1: Establish Voice Calls Between Locally Terminated Telephones

In this task, you will configure dial peers so that you can make calls between two telephones connected to your voice-enabled router.

Exercise Procedure

Complete these steps:

Step 1 Using the Classroom Dial Plan table in the Job Aids section, verify the telephone numbers you will use for your local telephones. Note these numbers below:

1st telephone ____________________ 2nd telephone ____________________

Step 2 Configure dial peers to enable calls between these two locally terminated telephones in both directions. Place calls in both directions to test configuration.

Step 3 Use appropriate commands to view your newly configured dial peers.

Step 4 Place a call between telephones and leave them both off hook. Use appropriate commands to view your active call. Find the following information:

How many and what type of call legs were created? ____________

Which codec is being used for this call? ____________

What is the original calling number? ____________ called number? ____________

Step 5 Use **debug** to see digits being collected by the voice port. Once debugging is turned on, place a call between telephones to view the digit collection.

Step 6 Verify and experiment with previously configured voice port parameters such as **cptone** and **ring cadence**. Listen to the distinctive rings you have configured.

Exercise Verification

You have completed this exercise when you attain these results:

- Make calls between your locally terminated telephones in both directions.

Task 2: Forward Calls to the PSTN

In this task, you will configure appropriate ports and dial peers to place calls to the PSTN. Initially this will be done with two-stage dialing, meaning you will hear two dial tones while placing the call. The first dial tone will be from your local router, and the second will be from the PSTN. Once this configuration is verified, you will change your configuration to place the call using one-stage dialing (only the local dial tone is heard), forwarding digits to the PSTN to automatically place the call.

Exercise Procedure

Complete these steps:

Step 1 Connect the PSTN device to each of the pod's routers using RJ-11 cables. Connect a single telephone to the PSTN device. Answer the following questions:

a) Which voice port type did you connect to the PSTN?______________________

b) What is the port number?____

c) What is the telephone number of your PSTN port?________________________

d) What is the telephone number of your partner's PSTN port? ________________

e) What is the telephone number of the PSTN-connected telephone?

____

Step 2 In this scenario, the requirement is to dial 9 to access the PSTN. Configure dial peers to allow access to the PSTN by dialing only 9, and then dialing the PSTN phone after the second dial tone is heard. Verify calls in both directions. Remember, at this point, you will hear two dial tones while placing the call.

Step 3 Edit the dial peer you just created to ring the PSTN phone and forward all appropriate digits to the PSTN so that only the local dial tone is heard. What is the default digit forwarding behavior on a POTS dial peer?

____

Step 4 What command did you use to forward digits to the PSTN?_____________________

Step 5 What other method could you have used? ____________________________________

Exercise Verification

You have completed this exercise when you obtain these results:

- Place calls to the PSTN telephone with both one-stage and two-stage dialing.

Task 3: Hunt Groups

In this task, you will configure a hunt group to send calls to both of your locally terminated telephones using the same number.

Exercise Procedure

Complete these steps:

Step 1 For the hunt group number, you will use the same first three digits as your telephones are now configured for. The fourth digit will be 9. For example, if your telephone numbers are 1101 and 1102, you will use 1109 as the hunt group number. Write your hunt group number.

Hunt group number ____

Step 2 Ensure that both telephones are still connected to your router voice ports.

Step 3 Configure additional dial peers for your local FXS ports to reach both voice ports using the same new hunt group number for each dial peer. Do not edit existing dial peers for this exercise, as they will be needed for future labs. This means that you will have two new dial peers, each for the same hunt group number (such as 1109), assigned to each of your FXS ports. You will also have the original dial peers configured from Task 1.

Step 4 To properly test the hunt behavior between your two phones, you will use the PSTN telephone. To reach the hunt group number from the PSTN phone, dial the seven-digit PSTN number of the FXO port that attaches to your router. At the secondary dial tone, dial your hunt group number. Repeat this many times, taking note of any patterns as to how calls are allocated to each telephone. The default behavior at this point will be random choosing of a port. However, to demonstrate this randomness with only two telephones, you may have to make a number of calls.

Step 5 One way to control the order of hunting is through the use of the **preference** command. What is the default setting for preference on a dial peer?

What command did you use to verify the setting? __________________________

Change the preference on one of your hunt dial peers to 1. Which setting is preferred, the 0 or 1 setting? ____________

Test the hunt group again by calling several times, making note of which telephone rings first. Is this what you expected? Now take the preferred telephone off hook and dial the hunt group number again. Did the other telephone ring? _______________

On the dial peer with preference set to 0, change the preference to 2. Which telephone should always ring first? Test the hunt group again. Is the outcome what you expected? ____

Step 6 You can configure hunt behavior for all dial peers globally with the **dial-peer hunt** command. Before changing this setting, find out what the default setting is and note it below.

Default dial-peer hunt setting ____

What command did you use to view this? ___________________________________

Configure the **dial-peer hunt** setting to 7. How do you expect your hunt group to choose which telephone to ring now? Test the hunt group again. Is the outcome what you expected? ____

Step 7 Delete the dial peers you created for the hunt groups.

Exercise Verification

You have completed this exercise when you attain these results:

- Control your hunt group behavior and explain how each command works.

Task 4: Establish Calls Between Telephones Interconnected by a Digital Facility

In this task, you will configure and test dial peers to place calls to your PBX-attached telephone.

Exercise Procedure

Complete these steps:

Step 1 Move one of your telephones and plug it into the lowest numbered port of your PBX device. Since the PBX device is preconfigured with the dial plan for the classroom lab, test the PBX outbound calling functionality by calling from the PBX-attached telephone to the router-attached telephone. This should work with the configuration that is in place.

Step 2 Configure dial-peer functionality on your voice-enabled router to place a call to the PBX-attached telephone.

What is the telephone number of your PBX-attached telephone? ________________

What is the voice port number you will use in this dial peer? ___________________

Step 3 Now that the number of dial peers is growing, check which dial peer will be matched when dialing your PBX-attached telephone number using a **show** command.

Step 4 Test the configuration by calling from your router-attached telephone to your PBX-attached telephone. Does the call go through? If not, check which digits are being forwarded to the PBX. Correct the problem and test again. If digit forwarding is correct, check the configuration of the DS0-group from Lab 4.1 or ask your instructor for assistance.

Step 5 Save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- Call in both directions between the PBX-attached telephone and the router-attached telephone.

Lab Exercise 4-2: Special-Purpose Connections

Complete this lab exercise to practice what you learned in the related lesson.

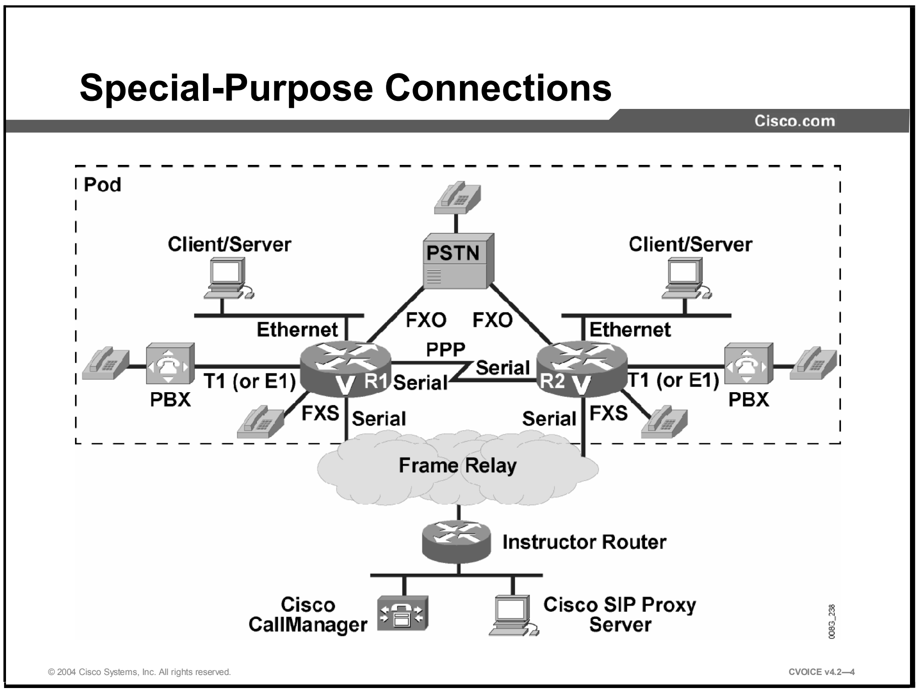
Exercise Objective

In this exercise, you will explore different connection types and their uses. After completing this exercise, you will be able to:

- Simulate autoattendant functions through use of PLAR and PLAR-OPX
- Create a tie-line connection for calls between two PBXs
- Use appropriate **show** and **debug** commands to monitor and troubleshoot the connections

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- No new resources are required.

Command List

The commands used in this exercise are described in the table here.

Connection Commands

Command	Description
connection plar opx *string*	Specifies a plar-opx connection. This command is executed in voice-port configuration mode.
connection plar *string*	Specifies a PLAR connection. This command is executed in voice-port configuration mode.
connection tie-line *string*	Specifies a tie-line connection. This command is executed in voice-port configuration mode.

Job Aids

There are no job aids for this lab exercise.

Task 1: Connecting to PLAR and PLAR OPX

By default, when a call comes into an FXO port, the router presents dial tone after going off hook to answer the call. At times this default behavior may not be desirable. In this task, you will configure the voice port to simulate an autoattendant function.

Exercise Procedure

Complete these steps:

Caution	Do not save your configuration during this lab. You will be asked to reload the router at the end of the lab to revert back to the last lab configuration.

Step 1 For this task, you will use the PSTN telephone to dial into the router FXO port. You want the FXO port to automatically direct the incoming call to your router's FXS telephone, thereby simulating autoattendant. Simulate the autoattendant function by configuring PLAR on the FXO port of your voice-enabled router. Direct all calls coming into the FXO port to the telephone connected to your FXS port on the same router.

Step 2 Test the functionality by placing a call from the PSTN-attached telephone to your router-attached telephone. Do you hear one or two dial tones? Place another call. Listen carefully to the ringback tone. You should hear an initial ringback from the PSTN, then a secondary ringback from the FXS port as it is ringing the telephone. From the PSTN's perspective, the call is complete when its ringing is answered and billing has started, even though the FXS port is still ringing.

Step 3 Change the PLAR configuration to plar-opx, pointing it to the same telephone number as in Step 2.

Step 4 Test the functionality by placing another call from the PSTN-attached telephone to the router-attached telephone. Is the ringback tone different than in Step 2? Did you hear a second ringback tone? __________________________

Exercise Verification

You have completed this exercise when you attain these results:

- Place a call to the FXO port of your router from the PSTN-attached telephone and, without further dialing, connect to the router-attached telephone.
- Explain the difference between PLAR and plar-opx.

Task 2: Connection Tie-Line

In the previous task, you configured a PLAR connection and saw that the configuration allowed for calls to a single specific number without access to a dial tone. Traditionally, tie-lines connect two PBXs together when a user at one site needs to connect to any number of users at the other site. Configuring tie-lines gives you the capability to call multiple numbers at the remote site by dialing the number directly upon receiving dial tone.

Exercise Procedure

Complete these steps:

Step 1 Configure tie-line functionality on your digital voice port. For the string entry, the odd-numbered routers will use the digit 7 to reach the even-numbered PBX, and the even-numbered routers will use the digit 8 to reach the odd-numbered PBX. This code ties all calls coming into the digital voice port from your PBX to a dial peer that will point the call across the network to the remote site and vice versa. Although you only dial a four-digit number to reach the remote site, the router will be processing a five-digit number because it will automatically prepend the code digit to your four-digit number.

Step 2 Now you need to configure a VoIP dial peer to point the call across the network toward your partner's PBX. Since there is no loopback address, you can send the call to any valid address on your partner's router.

Write your partner's IP address here. ________________________

Ping it to ensure connectivity.

Remember, the router is processing your code digit plus the four digits you will dial for a five-digit dialing string.

What four-digit number will you dial to reach your partner's PBX-attached telephone?____________________________

What five-digit number will the router process to connect to your partner's PBX-attached telephone? ___________________

Step 3 To complete the tie-line connection across the IP network, you will have to enter the following VoIP dial peer:

router(config)# dial-peer voice *tag* voip

router(config-dial-peer)# destination-pattern *string*
(Use the five digits noted in Step 2.)

router(config-dial-peer)# session target ipv4:*x.x.x.x*
(Use the IP address from Step 2.)

Step 4 As the last task, remember that your partner will be calling your PBX-attached telephone as well. Your partner's router will be passing a five digit string to your router to complete the call to your PBX.

What five-digit string will your partner's router be passing to your router for calls destined for your PBX telephone? _________________________

Configure a POTS dial peer to terminate the five-digit string as it comes in from your partner's router, and to forward only the four required digits to your PBX via the digital voice port.

Step 5 Coordinate with your partner to ensure that all tasks are complete before testing. Test the configuration by placing calls between the PBX-attached telephones in both directions. Use **show** and **debug** commands to view the processing of the call.

Step 6 Reload the router to revert back to your previous configuration. When you are asked if you wish to save your configuration, enter **no**.

Exercise Verification

You have completed this exercise when you attain these results:

- Configure and verify tie-line connections between PBXs.
- Place calls in both directions between PBX-attached telephones.

Lab Exercise 5-1: Basic VoIP

Complete this lab exercise to practice what you learned in the related lesson.

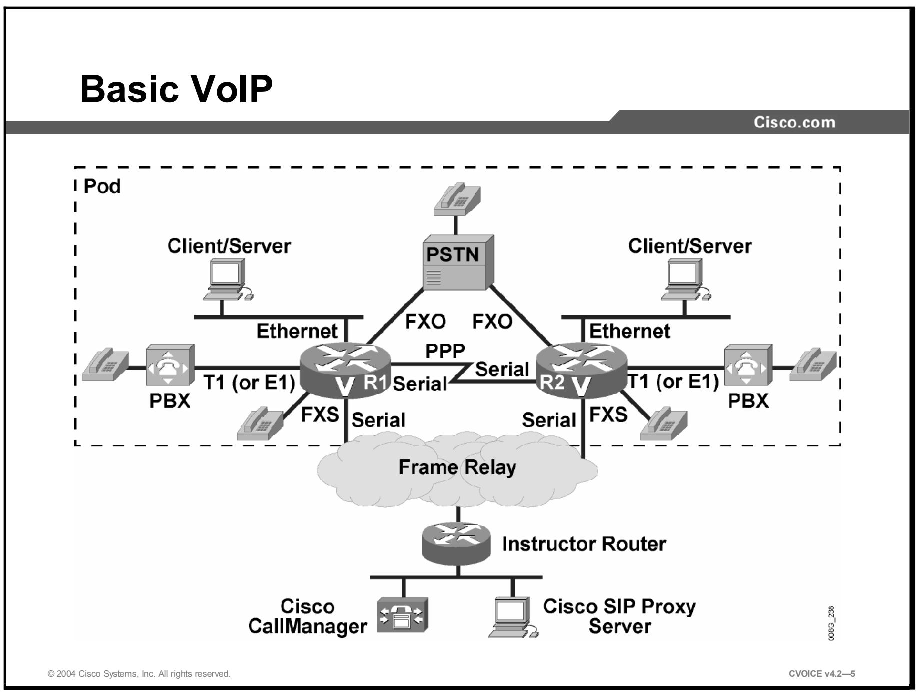
Exercise Objective

In this exercise, you will establish basic VoIP connectivity between telephones in your pod. You will also investigate the use of appropriate VoIP dial-peer parameters. After completing this exercise, you will be able to meet these objectives:

- Configure VoIP connections
- Describe how dial-peer matching occurs
- Describe and configure proper use of dial-peer codec parameters
- Verify basic call setup through **debug** commands
- Use appropriate **show** and **debug** commands to monitor and troubleshoot the connections

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- No new resources are required.

Command List

The commands used in this exercise are described in the table here.

VoIP Commands

Command	Description
codec *codec-name*	Specifies which codec is to be used for calls matching this dial peer. This command is executed in dial-peer configuration mode.
codec preference *1-14 codec-name*	Configures one entry in the codec list under the voice class codec command. Repeat this command as many times as you need to specify codecs in this list. This command is executed in class configuration mode.
debug voip ccapi inout	Displays real-time call control processing and call leg information. This command is executed in privileged mode.
debug voip rtcp	Displays real-time RTCP reports per call. This command is executed in privileged mode.
default *parameter*	Sets the specified parameter back to its default setting. For example, **default codec** will set the dial peer to use the default codec for that device. This command can be executed in various configuration modes.
dial-peer voice *tag* voip	Enters dial-peer configuration mode and specifies VoIP. This command is executed in global configuration mode.
frame relay ip rtp header-compression	Enables RTP header compression on a Frame Relay interface. This command is executed in interface configuration mode.
ip rtp header-compression	Enables RTP header compression on an interface. This command is executed in interface configuration mode.
session target ipv4:*x.x.x.x*	Specifies the destination IP address for the gateway terminating a VoIP call. This command is executed in dial-peer configuration mode.
show voice dsp	Displays digital signal processor (DSP) usage. This command is executed in user mode.
show ip rtp header-compression	Displays statistics relating to RTP header compression. This command is executed in user mode.
voice class codec *tag*	Enters voice class codec configuration mode. This command is executed in global configuration mode.
voice-class codec *tag*	Applies a predefined codec list to a dial peer. The tag must match the tag of the defined codec class. This command is executed in dial-peer configuration mode.

Job Aids

These job aids are available to help you complete the lab exercise:

- Understanding **debug voip ccapi inout** command output.

Task 1: VoIP Dial Peers

You will start this lab by configuring basic VoIP dial peers using the default parameters to process the call. You will verify configuration by placing calls across the IP network to your partner's telephones.

Exercise Procedure

Complete these steps:

Step 1 Configure VoIP dial peers to reach both of the router-attached telephones connected to your partner's equipment. In preparation, note your partner's two telephone numbers below. Also note a valid IP address on your partner's router.

Telephone numbers____________________ ________________________

IP address ___________________________

Step 2 Test your configuration by placing calls to both of your partner's telephones.

Step 3 Use **show** commands to verify:

- Which dial peer will be matched when a specific number is dialed
- Active call parameters
- What DSP resources are being used for the cal

Step 4 Use **debug** commands to verify:

- What the calling number is
- What the called number is
- Which dial peer was matched

Step 5 Use the **debug voip rtcp** command to view the control protocol information being reported for the call.

Exercise Verification

You have completed this exercise when you attain these results:

- Place calls to both of your partner's telephones.

Task 2: Codec Configuration

You will change the codec settings on your VoIP dial peers and investigate how it impacts the ability to make calls. Make sure that you and your pod partner are both working on the same step. Coordinate moving through the steps and performing tests together.

Exercise Procedure

Complete these steps:

Step 1 Use **show** commands to verify the default codec setting and note it below.

Default codec____________________

Step 2 Change the codec on R1's VoIP dial peer pointing to R2. Set the codec to g723r53.

Step 3 Both R1 and R2 should verify whether the call was successful, and if so, which codec was used.
Was the call successful? _______________ Codec used ________________

Step 4 Change the codec on R2's VoIP dial peer pointing to R1 to match R1's codec.

Step 5 Verify whether the codec was successful, and if so, which codec was used.
Was the call successful? _______________ Codec used ________________

Step 6 Change the codec setting back to default.

Exercise Verification

You have completed this exercise when you attain these results:

- Successfully configure the codec setting for VoIP calls.

Task 3: Effects of Bandwidth Requirements and Header Compression

You will investigate the effects of passing voice over low bandwidth links and the tools available to improve voice quality on those links.

Exercise Procedure

Complete these steps:

Step 1 Using the bandwidth requirement concepts, determine the required bandwidth for a G.711 call without header compression on: PPP link ____________________ Frame Relay link ____________________

Step 2 Change the codec on both R1 and R2 to g711ulaw. Place a call from one router-connected telephone to the other over the IP network. Is the quality acceptable? If not, why not? ____

Step 3 Enable RTP header compression on your PPP and Frame Relay interfaces on both routers in your pod.

Step 4 What command did you use on the PPP link? ___________________________

Step 5 What command did you use on the Frame Relay link? ___________________

Step 6 Test the voice quality. Has the quality improved? Why or why not? Is it acceptable?

Step 7 Change the codec on both R1 and R2 back to the default setting. What command did you use to reset to the default? _______________________________________

Exercise Verification

You have completed this exercise when you attain these results:

- Explain how and when to use RTP header compression.

Task 4: Configuring Codec Negotiation

As you saw in the previous sections, configuring a specific codec at the dial-peer level restricts that dial peer to responding with only a single codec choice during negotiation. At times it is desirable to respond with a list of codecs to match the incoming call. For example, when a call is coming from the LAN segment, it may negotiate G.711 codec for better voice quality because there is enough bandwidth to carry it. For a call coming into the router from a WAN segment, you will want to match a codec for compressed voice. In order for a single dial peer to match more than one codec, you must configure a list of codecs to negotiate.

Exercise Procedure

Complete these steps:

Step 1 Define a codec preference list. On R1 select g711ulaw as the first choice and g729r8 as the second choice. On R2 select g729r8 as the first choice and g711ulaw as the second choice.

Step 2 Apply the codec list to the VoIP dial peer pointing to your partner's IP address.

Step 3 Test calls in both directions. Use **show** commands to determine the order of preference for codec selection. Discuss the results with your partner.

Step 4 Remove the codec list and its application in the dial peer.

Step 5 Save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- Explain how and when to configure codec negotiation parameters.

Lab Exercise 6-1: VoIP with H.323

Complete this lab exercise to practice what you learned in the related lesson.

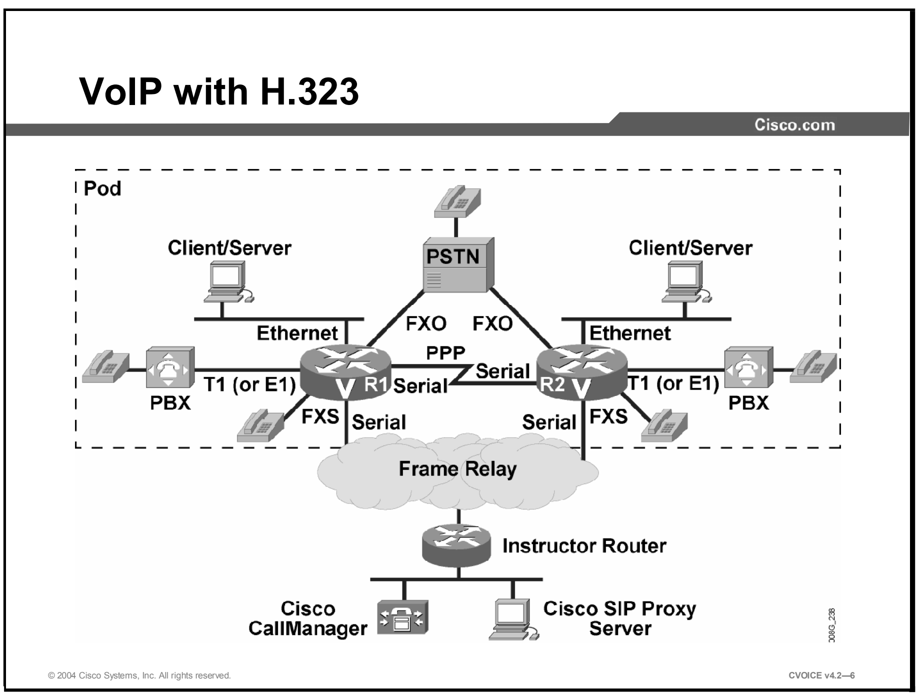
Exercise Objective

You need to be able to call all other router-attached telephones in the classroom. Since there are several sets of routers and numerous telephones in the classroom lab, you do not want to manually enter dial peers for all possible telephones. In this exercise, you will implement a scalable VoIP environment using H.323 gatekeepers. You will experiment with a single-zone, single-gatekeeper environment and a multizone environment including multiple gatekeepers. After completing this exercise, you will be able to meet these objectives:

- Configure single-zone and multizone H.323 gatekeeper environments for VoIP scalability
- Use **debug** and **show** commands to monitor the status and progress of call setup procedures in an H.323 environment

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- Instructor router with gatekeeper functionality enabled

Command List

The commands used in this exercise are described in the table here.

H.323 Commands

Command	Description
debug cch323 h225	Traces the state transition of the H.225 state machine based on the processed event. This command is executed in privileged mode.
debug gatekeeper server	Traces all the message exchanges between the Cisco IOS gatekeeper and the external applications. This command is executed in privileged mode.
gatekeeper	Enables gatekeeper functionality. This command is executed in global configuration mode.
gateway	Enables gateway configuration. This command is executed in global configuration mode.
h323-gateway voip interface	Specifies that this interface’s IP address will be used to register with the gatekeeper. This command is executed in interface configuration mode.
h323-gateway voip id *gatekeeper-id* ipaddr *gatekeeper-ip-addr*	Defines the name and location of the gatekeeper for this gateway. This command is executed in interface configuration mode.
h323-gateway voip h323-id	Defines the name that will be used to identify this gateway to the gatekeeper. This command is executed in interface configuration mode.
session target ras	Specifies that the RAS protocol is being used to determine the IP address of the session target. This command is executed in dial-peer configuration mode.
show gatekeeper *option*	Displays various parameters and status for a gatekeeper. This command is executed in user mode on the gatekeeper.
show gateway	Shows if the gateway is connected to the gatekeeper. This command is executed in user mode on the gateway.
zone local|remote	Statically specifies a local or remote zone. Parameters include gatekeeper name, domain name, and IP address. This command is executed in gk configuration mode.
zone prefix	Specifies which group of telephone numbers are reachable via a specific gatekeeper. The prefix typically contains wildcards for number summarization. This command is executed in gk configuration mode.

Job Aids

These job aids are available to help you complete the lab exercise:

- Instructor's router gateway is gk, IP address is 192.168.100.1

Task 1: Single Zone Environment

In this task, you will configure your router to be a gateway that registers with a gatekeeper. This is a single-zone configuration, since all the voice-enabled routers will register to a single gatekeeper.

Exercise Procedure

Complete these steps:

Step 1 Enable **debug** in order to see the interactions between H.323 components once gatekeeper support has been enabled. Ensure that you can reach the gatekeeper by pinging 192.168.100.1.

Step 2 Using the bulleted items, configure your router as an H.323 gateway. Initially, all routers will use the "instructor router" as their gatekeeper. The IP address of the instructor router is 192.168.100.1. Analyze the **debug** output to observe the interactions between your router and the gatekeeper. The following tasks are necessary to register with a gatekeeper:

- Configure your router to be a gateway.
- Specify which interface IP address will be used to register with the gatekeeper and also specify the identity of the gatekeeper. Use your Ethernet interface for gateway configuration.
- Specify an H.323 ID for your gateway by combining "gw" with your pod and router number. For example, if you were pod 5, router 2, your H.323 ID would be gwP5R2.
- Use the **show gateway** command on your router to verify that you have registered with the gatekeeper.

What is the gatekeeper name you have registered with? ______________________

Under the command-line interface (CLI) alias list, which numbers register with the gatekeeper? ____

Step 3 Connect to the instructor router and use variations of the **show gatekeeper** command to verify that your router is registered, and that it has registered the destination patterns from your POTS dial peers.

What IP address is registered to be used for calls to your device? ______________

What port number is being used for call signaling? __________________________

What is your router's H.323 ID? ____

What zone is your gateway part of? ____

Step 4 Create a dial peer to use RAS for all calls to destinations outside your pod. Be creative when creating this dial peer.

Step 5 Establish a voice call to a telephone in another pod. Use the **show call active voice** command to provide the following information:

How many and what type of call legs are established? ____________________

Calling number ____________________

Called number ____________________

Remote IP address ____________________

Remote UDP port ____________________

Step 6 Place another call to a telephone outside your own pod. Use the **debug cch323 h225** command to provide the following information:

What is the source address of the call? ____________________

What is the destination address of the call? ____________________

Note Do not proceed to Task 2 until all the pods have completed Task 1 and the instructor tells you to proceed.

Exercise Verification

You have completed this exercise when you attain these results:

- Configure and verify that your gateway has registered with a gatekeeper.
- Call all other telephones in the classroom.

Task 2: Multizone Environment

In this task, you will configure your voice-enabled router to be both a gateway and a gatekeeper. Your router's gateway function will be configured to register internally with your router's gatekeeper function. Since all the other routers are also configuring themselves to be gatekeepers, this will be a multizone configuration.

Exercise Procedure

Complete these steps:

Step 1 You already have VoIP dial peers to your partner's telephones in your own pod by way of specific dial peers. Leaving this functionality in place, you will now expand reachability to other pods by setting up gatekeeper capabilities on your router and configuring your router to know about other gatekeepers outside your pod. Using the bulleted items, enable gatekeeper functionality on your router. In your gatekeeper configuration, include the routers in the other pods as remote zones (gatekeepers), but do not include the other router in your pod as a remote. To enable gatekeeper functionality, perform the following tasks:

- Enable gatekeeper functionality.
- Define your local zone information. For your zone name, use "gk" + your pod and router numbers. For example, if you are in Pod 3 Router 1, your zone name will be gkP3R1. The domain name is cisco.com. Use your Ethernet IP address for the RAS address. Make sure your gatekeeper is not in shutdown state.
- Define all remote zones using the same naming and addressing convention as in the previous task. Do not include the other router in your pod as a remote. Configure **zone prefixes** for all remote zones only.
- Change the H.323 gateway configuration on your router to point to your own router as the gatekeeper. Ensure that you use the ID and IP address set up in the previous tasks. This configuration connects the gateway process in your router. The gateway process in your router then sets up telephone calls to the gatekeeper process that knows about all the other gatekeepers in the classroom.
- Establish calls to other pods and use **show** and **debug** commands to observe the interactions between H.323 components.

Step 2 Save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- Configure and verify that your gateway has registered with a gatekeeper.
- Configure and verify gatekeeper functionality on your router.
- Call all other telephones in the classroom.

Lab Exercise 6-2: VoIP with SIP

Complete this lab exercise to practice what you learned in the related lesson.

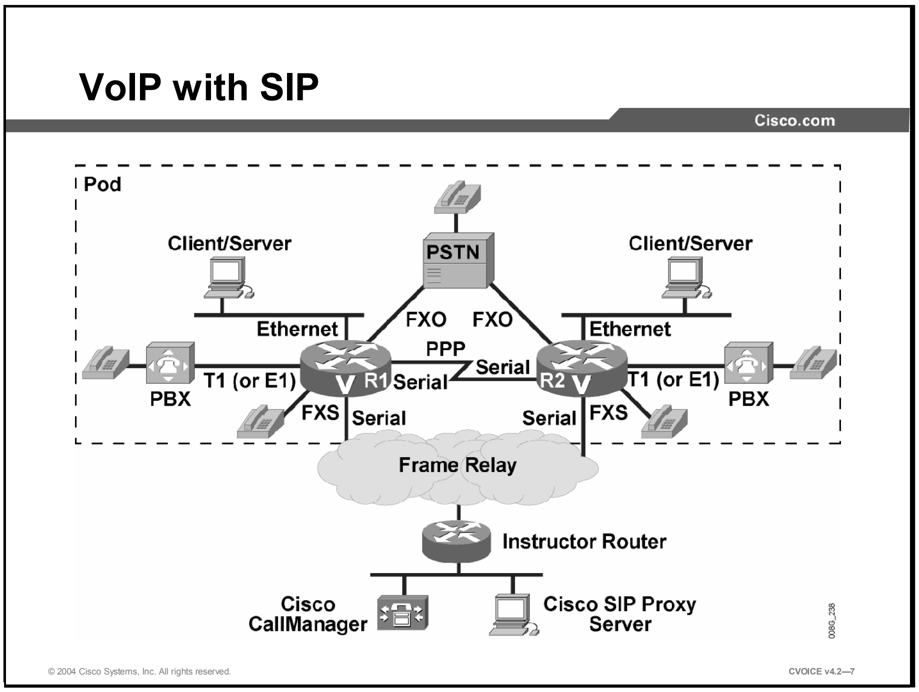
Exercise Objective

In this exercise, you will use SIP direct procedures (user agent to user agent) to establish VoIP calls. After completing this exercise, you will be able to meet these objectives:

- Configure dial peers to use SIP call control procedures to set up VoIP calls

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- Standard equipment listed for CVOICE labs

Command List

The commands used in this exercise are described in the table here.

SIP Commands

Command	Description
debug ccsip *options*	Displays various real-time SIP call information. This command is executed in privileged mode.
session protocol sipv2	Specifies that the VoIP dial peer should use SIP call control when processing the call. This command is executed in dial-peer configuration mode.
session target sip-server	Specifies the use of the proxy server. This command is executed in dial-peer configuration mode.
show sip-ua *options*	Displays various parameters for SIP. This command is executed in privileged mode.
sip-ua	Enters SIP user agent configuration mode. This command is executed in global configuration mode.

Job Aids

There are no job aids for this lab exercise.

Task 1: Configuring for SIP

In this exercise, you will be configuring your router to initiate calls with your partner's router using SIP. For this exercise, you will use SIP direct, user agent to user agent.

Exercise Procedure

Complete these steps:

Note	Do not save your configuration in this lab. You will be asked to reload the router at the end of the lab to revert to the previous lab configuration.

Step 1 Modify the existing VoIP dial peers that point to your partner's telephones to use direct unnumbered acknowledgment (UA to UA) SIP call control procedures when establishing voice calls. For direct calls, the IP address in the **session target** command will be a valid address of your partner's router.

Step 2 Use the **show call active voice** command to verify that you now have SIP call legs when placing a call to your partner's telephone.

Step 3 Enable SIP debugging and place a call between your telephone and your partner's telephone. Observe the call setup, capabilities negotiation, and assignment of ports for the call.

Step 4 Investigate the status of SIP with variations of the **show sip-ua** command.

Step 5 Do not save your configuration at this time. You will be asked to reload the router after the next lab.

Exercise Verification

You have completed this exercise when you attain these results:

- Establish voice calls between telephones connected to your routers by way of direct SIP call control procedures.

Lab Exercise 6-3: VoIP with MGCP

Complete this lab exercise to practice what you learned in the related lesson.

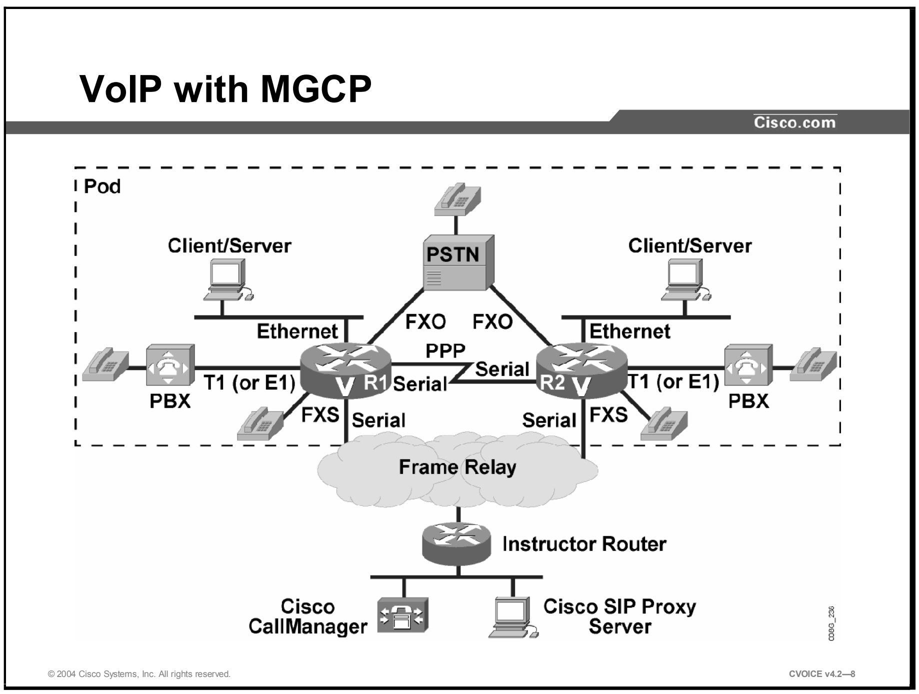
Exercise Objective

In this exercise, you will use a call agent to establish voice calls between telephones connected to MGCP residential gateways. After completing this exercise, you will be able to meet these objectives:

- Configure your routers as MGCP residential gateways and have the routers use an MGCP call agent to establish voice calls between them
- Use **debug** commands to analyze the interactions between MGCP gateways and a call agent
- Use **show** commands to view the status of MGCP endpoints, connections, and calls

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- MGCP call agent resident on the instructor LAN

Command List

The commands used in this exercise are described in the table here.

MGCP Commands

Command	Description
application mgcpapp	Enables MGCP for the voice port configured in this dial peer. This command is executed in dial-peer configuration mode.
ccm-manager mgcp	Enables support for Cisco CallManager MGCP. This command is executed in global configuration mode.
debug ccm-manager *options*	Displays real-time Cisco CallManager MGCP information. This command is executed in privileged mode.
debug mgcp *options*	Displays real-time MGCP call information. This command is executed in privileged mode.
mgcp	Enables MGCP functionality on the router. This command is executed in global configuration mode.
mgcp call-agent *ip-address*	Specifies the MGCP call agent IP address. This command is executed in global configuration mode.
show call application voice summary	Displays a list of available applications. This command is executed in user mode.
show ccm-manager	Displays Cisco CallManager MGCP information. This command is executed in user mode.
debug ccm-manager *options*	Displays real-time Cisco CallManager MGCP information. This command is executed in privileged mode.

Job Aids

These job aids are available to help you complete the lab exercise:

- MGCP call agent IP address is 192.168.100.100

Task 1: MGCP Calls

In this task, you will configure your router to use the MGCP protocol for calls throughout the classroom.

Exercise Procedure

Complete these steps:

Note	Do not save your configuration in this lab. You will be asked to reload the router at the end of the lab to revert to the previous lab's configuration.

Step 1 Ensure that you can reach the MGCP call agent by pinging IP address 192.168.100.100. Inform the instructor if you are unable to reach it.

Step 2 Configure both of your local POTS dial peers to use MGCP.

Step 3 Enable MGCP on the routers in your pod by specifying MGCP and configuring the IP address of the call agent.

Step 4 Modify the POTS dial peers for the other FXS on your router by removing the destination pattern and entering the **application mgcpapp** command.

Step 5 Use **show** commands to verify that the gateway is registered with the Cisco CallManager.

Step 6 Enable **debug** and establish a call between the telephones connected to your routers. Analyze the **debug** output, looking for the interactions between your router and the call agent and between your router and the destination router.

Step 7 Investigate the status of MGCP with variations of the **show mgcp** command.

Step 8 Reload the router. Make sure you say no when asked if you want to save your configuration.

Exercise Verification

You have completed this exercise when you attain these results:

- Establish voice calls between the telephones connected to your routers by way of MGCP.

Lab Exercise 7-1: QoS

Complete this lab exercise to practice what you learned in the related lesson.

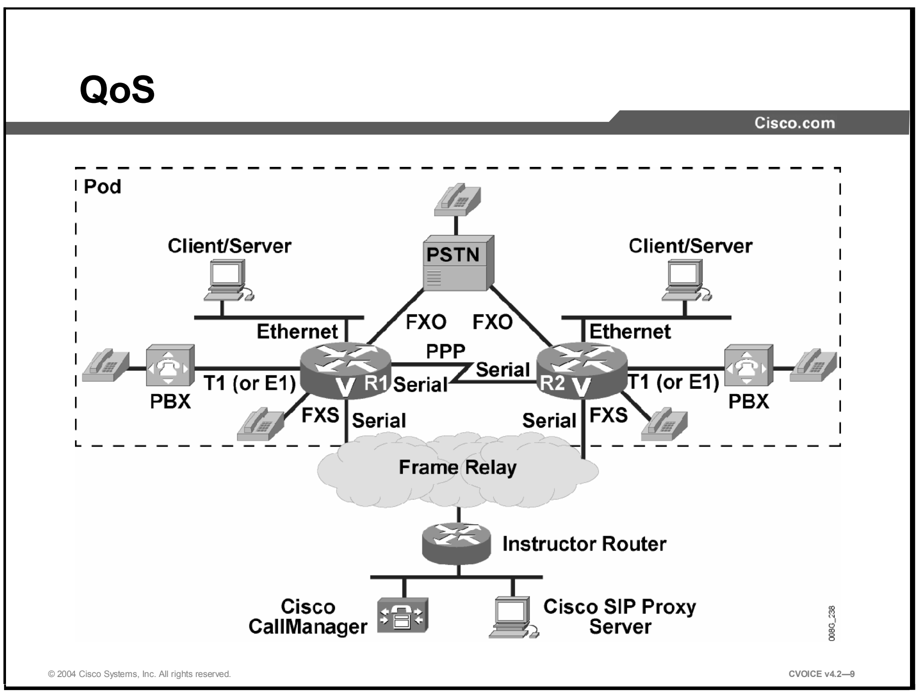
Exercise Objective

In this exercise, you will observe the incremental and combined effects of applying QoS concepts to improve voice quality end to end. After completing this exercise, you will be able to meet these objectives:

- Implement quality improvements on low-speed links with QoS features such as fragmentation, interleaving, and Frame Relay traffic shaping
- Implement features such as voice packet marking (tagging) and queuing to improve voice quality end to end
- Confirm, by testing, that the QoS features contribute to overall improvements in voice quality

Visual Objective

The figure illustrates what you will accomplish in this exercise.



Required Resources

These are the resources and equipment required to complete this exercise:

- No new resources are required.

Command List

The commands used in this exercise are described in the table here.

QoS Commands

Command	Description
auto qos voip	Enables auto QoS configuration for interfaces. This command is entered in interface configuration mode, except Frame Relay, which requires subinterface configuration mode.
bandwidth *bandwidth(Kbps)*	Sets the correct bandwidth on the physical interface. Must be set for correct fragment sizes to be calculated. This command is executed in interface configuration mode.
class *map-class-name*	Applies the Frame Relay map class that was created previously. This command is executed in data-link connection identifier (DLCI) configuration mode.
class-map *name*	Creates a class map and enters class map configuration mode. This command is executed in global configuration mode.
class *name*	Associates the previously created class map to this policy. Ensure that the same name is used as when you created the class map. This command is executed in policy map configuration mode.
clear frame-relay inverse-arp	Clears ghost frame-relay inverse-arp associations. This command is executed in privileged mode.
encapsulation ppp	Enables PPP encapsulation. Configure in both serial and multilink interfaces.
fair-queue	Enables WFQ on the interface. This command is executed in MP-interface configuration mode.
frame-relay bc *value*	Specifies the committed burst value. This value should be 1/100 of the CIR for a 10-ms maximum burst delay. This command is executed in map-class configuration mode.
frame-relay cir *value*	Specifies the CIR of the Frame Relay link. This command is executed in map-class configuration mode.
frame-relay fair-queue	Enables WFQ for the interface. This command is executed in map-class configuration mode.
frame-relay fragment *fragment-size-in-bytes*	Specifies the fragment size for frames. Fragment size should be set to accommodate a 10-ms delay in queue. Calculate this based on the link speed. This command is executed in map-class configuration mode.
frame-relay interface-dlci *dlci-number*	Enables DLCI configuration mode for a specific DLCI. This command is executed in interface configuration mode.
frame-relay traffic-shaping	Enables Frame Relay traffic shaping on the interface. This command is executed in interface configuration mode.

Command	Description
interface multilink ***interface-no***	Creates an MP interface and enters multilink interface configuration mode. This command is executed in global configuration mode.
interface ***interface-type interface-number subinterface-number*** **point-to-point**	Creates a subinterface on the main interface. This command is executed in interface configuration mode.
ip cef	Enables Cisco Express Forwarding switching on interfaces. This command is entered in global configuration mode.
ip qos dscp ***option***	Specifies the DSCP for the dial peer. Use class selector 5 (cs5) to specify IP precedence of 5. This command is executed in dial-peer configuration mode.
map-class frame-relay ***name***	Creates a Frame Relay map class and enters map class configuration mode. This command is executed in global configuration mode.
match ip dscp cs5	Specifies that the class map should match packets marked with a precedence of 5. This command is executed in class map configuration mode.
multilink-group ***group-no***	Used to link the multilink (logical) interface to the serial (physical) interface. Use the same command and group number in both interfaces to bind them together.
policy-map ***name***	Creates a policy map and enters policy map configuration mode. This command is executed in global configuration mode.
ppp multilink	Enables MLP on an interface. Configure in both serial and multilink interfaces.
ppp multilink fragment delay ***delay-max***	Specifies a maximum size in units of time for packet fragments on an MLP bundle. This command is executed in MP-interface configuration mode.
ppp multilink interleave	Enables interleaving of packets among the fragments of larger packets on an MLP bundle. This command is executed in MP-interface configuration mode.
priority ***bandwidth***	Assigns the voice class traffic to the priority queue and specifies how much bandwidth to allow for that queue. This command is executed in policy map configuration mode.
service-policy ***direction name***	Associates a policy map to an interface. Ensure that you use the same name as that of the policy map you created. This command is executed in interface configuration mode.
show policy-map interface ***int-number***	Displays the policy applied to the interface. This command is executed in use mode.

Job Aids

There are no job aids for this lab exercise.

Task 1: Enabling AutoQoS

In this task, you will configure AutoQos on the Frame Relay interface. You will be tasked with working within your pod to accomplish voice calls with QoS enabled.

Exercise Procedure

Complete these steps:

Step 1 Enable Cisco Express Forwarding (CEF) switching on the router.

Step 2 Shut down the PPP interface that connects to the other router in your pod.

Step 3 Using the **show frame-relay map** command, make a note of the correct DLCI number that connects to your pod partner's router. The correct map will show the IP address of your partner's Frame Relay interface. The DLCI number is: __________

Step 4 To prepare for AutoQos, your Frame Relay interface must be changed to a subinterface. To do this, first delete the IP address from the main Frame Relay interface. Next, go to privileged mode, and clear the frame-relay inverse-arp table with the command **clear frame-relay inverse-arp**. Next, create a point-to-point subinterface on the main Frame Relay interface. Assign the IP address that you deleted from the main interface. Then, add a bandwidth statement reflecting 72 Kbps (remember that AutoQoS requires a bandwidth statement). Finally, add the **frame-relay interface-dlci** statement with the DLCI number you noted in Step 3.

Step 5 Be sure all of the steps outlined in Step 4 are completed.

Step 6 Reconfigure your dial peer that points to your partner's phones so that the session target points to the IP address of their Frame Relay interface.

Step 7 Ensure that you can ping your partner's Frame Relay IP address. Each of you must have completed all of the steps up to this point in order for the ping to work correctly. If necessary, wait for the other router to get to this point.

Step 8 Ensure that calls can be made in both directions. If not, check your dial peers. Notify your instructor if calls cannot be made.

Step 9 Using extended ping from one router only, send 100,000 pings at 1500 bytes each to the other router. The pings should be successful. Allow the pings to run.

Step 10 From the router where the pings were initiated, place a call to your partner's router. Test the voice quality by counting to 20 quickly. What is the quality of the call? Are there dropouts or missing audio periods? Discuss the results with your partner. Hang up the phones, abort the ping, and repeat the test in the other direction.

Step 11 Go to configuration mode for the DLCI on the subinterface you created. In DLCI configuration mode, add the command **auto qos voip**. The router will pause for a few seconds, then return a prompt.

Step 12 Return to the privileged mode and look at the running configuration.

What access list(s) was or were created?

____

How many classes were created in the policy map? ___________________________

In the map class, what fragment size was configured? _________________________

Why? ____

What map class was assigned to the Frame Relay subinterface DLCI?

____

Step 13 From one router, send a continuous count of 100,000 pings at 1500 bytes each. Place a call from the router initiating the ping and check voice quality by counting to 20 quickly. Did the voice quality improve? Discuss the results with your partner, then hang up, abort the ping, and repeat the test in the other direction.

Exercise Verification

You have completed this exercise when you attain these results:

- Maintain voice quality while transferring data over a shared path.

Lab Exercise Answer Key

Lab Exercise 2-1: Lab Familiarity

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod3R1
!
enable password san-fran
!
ip subnet-zero
!
!
ip host Pod3R2 192.168.1.32
ip host Pod3R1 192.168.1.31
ip host Pod2R2 192.168.1.22
ip host Pod2R1 192.168.1.21
ip host Pod1R2 192.168.1.12
ip host Pod1R1 192.168.1.11
!
voice call carrier capacity active
!
mta receive maximum-recipients 0
!
interface Ethernet0/0
 ip address 10.3.10.1 255.255.255.0
 half-duplex
!
interface Serial0/0
 ip address 192.168.1.31 255.255.255.0
 encapsulation frame-relay
!
interface Ethernet0/1
 no ip address
 shutdown
```

```
 half-duplex
!
interface Serial0/1
 description PPP to Pod3R1
 ip address 10.3.3.1 255.255.255.0
 encapsulation ppp
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
call rsvp-sync
!
voice-port 2/1/0
!
voice-port 2/1/1
!
mgcp profile default
!
dial-peer cor custom
!
gatekeeper
 shutdown
!
!
line con 0
line aux 0
line vty 0 4
 password router
 login
!
!
end
```


Lab Exercise 3-1: Voice Port Configuration

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password san-fran
!
voice-card 1
!
ip subnet-zero
!
!
!
!
!
voice call carrier capacity active
!
!
!
!
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
```

```
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
```



```
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
 cptone AU
 ring cadence define 4 1 8 10
!
voice-port 2/1/1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
!
gatekeeper
 shutdown
!
!
line con 0
 password router
 logging synchronous
 login
line aux 0
line vty 0 4
 password router
 login
line vty 5 15
```

```
 password router
 login
!
!
end
```

Lab Exercise 4-1: POTS Dial Peers

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
ip subnet-zero
!
!
!
!
!
voice call carrier capacity active
!
!
!
!
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
```



```
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
```

```
!
ip classless
ip http server
!
!
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
 cptone AU
 ring cadence define 4 1 8 10
!
voice-port 2/1/1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 port 2/1/0
!
dial-peer voice 1102 pots
 destination-pattern 1102
 port 2/1/1
!
dial-peer voice 3 pots
```



```
 destination-pattern 555....
 port 2/0/0
 forward-digits all
!
dial-peer voice 4 pots
 preference 1
 destination-pattern 1109
 port 2/1/0
!
dial-peer voice 5 pots
 preference 2
 destination-pattern 1109
 port 2/1/1
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
!
dial-peer hunt 7
!
gatekeeper
 shutdown
!
!
line con 0
 password 7 131718071F0916
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
line vty 5 15
 password 7 02140B4E1F031D
 login
!
!
```

end

Lab Exercise 4-2: Special-Purpose Connections

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
ip subnet-zero
!
!
!
!
!
voice call carrier capacity active
!
!
!
!
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
```



```
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
```

```
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
 connection Tie-line 7
!
voice-port 2/0/0
 connection plar opx 1102
!
voice-port 2/0/1
!
voice-port 2/1/0
 cptone AU
 ring cadence define 4 1 8 10
!
voice-port 2/1/1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 port 2/1/0
!
dial-peer voice 1102 pots
 destination-pattern 1102
 port 2/1/1
!
dial-peer voice 3 pots
 destination-pattern 555....
 port 2/0/0
 forward-digits all
```



```
!
dial-peer voice 4 pots
 preference 1
 destination-pattern 1109
 port 2/1/0
!
dial-peer voice 5 pots
 preference 2
 destination-pattern 1109
 port 2/1/1
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
!
dial-peer voice 7 voip
 destination-pattern 7....
 session target ipv4:10.1.1.2
!
dial-peer voice 8 pots
 destination-pattern 8....
 port 1/0:1
!
dial-peer hunt 7
!
gatekeeper
 shutdown
!
!
line con 0
 exec-timeout 0 0
 password 7 131718071F0916
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
line vty 5 15
```

```
 password 7 02140B4E1F031D
 login
!
!
end
```

Lab Exercise 5-1: Basic VoIP

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod3R1
!
enable password san-fran
!
memory-size iomem 10
voice-card 1
!
ip subnet-zero
!
!
ip host Pod1R1 192.168.1.11
ip host Pod1R2 192.168.1.12
ip host Pod2R1 192.168.1.21
ip host Pod2R2 192.168.1.22
ip host Pod3R1 192.168.1.31
ip host Pod3R2 192.168.1.32
!
!
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
```

```
!
!
!
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Ethernet0/0
 ip address 10.3.10.1 255.255.255.0
 half-duplex
!
interface Serial0/0
 ip address 192.168.1.31 255.255.255.0
 encapsulation frame-relay
 frame-relay ip rtp header-compression
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
 description PPP to Pod3R1
 ip address 10.3.3.1 255.255.255.0
```

```
 encapsulation ppp
 ip tcp header-compression iphc-format
 ip rtp header-compression iphc-format
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
 ring cadence pattern05
!
voice-port 2/1/1
 ring cadence pattern05
!
!
mgcp profile default
!
```



```
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
 destination-pattern 3101
 port 2/1/0
!
dial-peer voice 2 pots
 destination-pattern 3102
 port 2/1/1
!
dial-peer voice 3 pots
 destination-pattern 9.......
 port 2/0/0
!
dial-peer voice 91 pots
 preference 2
 destination-pattern 3109
 port 2/1/0
!
dial-peer voice 92 pots
 preference 1
 destination-pattern 3109
 port 2/1/1
!
dial-peer voice 20 pots
 destination-pattern 3151
 port 1/0:1
 forward-digits all
!
dial-peer voice 11 voip
 destination-pattern 3201
 voice-class codec 1
 session target ipv4:10.3.3.2
!
dial-peer voice 12 voip
 destination-pattern 3202
 voice-class codec 1
 session target ipv4:192.168.1.32
```

```
!
dial-peer voice 319 voip
 destination-pattern 21..
 session target ipv4:192.168.1.21
!
dial-peer hunt 7
!
gatekeeper
 shutdown
!
!
line con 0
line aux 0
line vty 0 4
 password router
 login
!
!
end
```

Lab Exercise 6-1: VoIP with H.323

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
ip subnet-zero
!
!
!
!
```



```
!
voice call carrier capacity active
!
!
!
!
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Loopback1
 no ip address
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
 h323-gateway voip interface
 h323-gateway voip id gkp1r1 ipaddr 10.1.10.1 1719
 h323-gateway voip h323-id gwP1R1
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
 frame-relay ip rtp header-compression
```

```
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
 ip tcp header-compression iphc-format
 ip rtp header-compression iphc-format
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
 connection Tie-line 7
!
voice-port 2/0/0
!
```



```
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 port 2/1/0
!
dial-peer voice 1102 pots
 destination-pattern 1102
 port 2/1/1
!
dial-peer voice 3 pots
 destination-pattern 555....
 port 2/0/0
 forward-digits all
!
dial-peer voice 4 pots
 preference 1
 destination-pattern 1109
 port 2/1/0
!
dial-peer voice 5 pots
 preference 2
 destination-pattern 1109
 port 2/1/1
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
```

```
!
dial-peer voice 7 voip
 destination-pattern 7....
 session target ipv4:10.1.1.2
!
dial-peer voice 8 pots
 destination-pattern 8....
 port 1/0:1
!
dial-peer voice 9 voip
 destination-pattern 1201
 session target ipv4:10.1.1.2
!
dial-peer voice 10 voip
 destination-pattern 1202
 session target ipv4:10.1.1.2
!
dial-peer voice 11 voip
 destination-pattern ....
 session target ras
!
dial-peer hunt 7
gateway
!
!
gatekeeper
 zone local gkp1r1 cisco.com 10.1.10.1
 zone remote gkp2r1 cisco.com 10.2.10.1 1719
 zone remote gkp2r2 cisco.com 10.2.20.1 1719
 zone remote gkp3r1 cisco.com 10.3.10.1 1719
 zone remote gkp3r2 cisco.com 10.3.20.1 1719
 zone prefix gkp2r1 21..
 zone prefix gkp2r2 22..
 zone prefix gkp3r1 31..
 zone prefix gkp3r2 32..
 no shutdown
!
!
line con 0
 exec-timeout 0 0
```



```
 password 7 131718071F0916
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
line vty 5 15
 password 7 02140B4E1F031D
 login
!
!
end
```

Lab Exercise 6-2: VoIP with SIP

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
ip subnet-zero
!
!
!
!
!
voice call carrier capacity active
!
!
!
!
```

```
!
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Loopback1
 no ip address
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
 h323-gateway voip interface
 h323-gateway voip id gkp1r1 ipaddr 10.1.10.1 1719
 h323-gateway voip h323-id gwP1R1
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
 frame-relay ip rtp header-compression
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
```

```
interface Serial0/1
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
 ip tcp header-compression iphc-format
 ip rtp header-compression iphc-format
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
 connection Tie-line 7
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
```

```
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 port 2/1/0
!
dial-peer voice 1102 pots
 destination-pattern 1102
 port 2/1/1
!
dial-peer voice 3 pots
 destination-pattern 555....
 port 2/0/0
 forward-digits all
!
dial-peer voice 4 pots
 preference 1
 destination-pattern 1109
 port 2/1/0
!
dial-peer voice 5 pots
 preference 2
 destination-pattern 1109
 port 2/1/1
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
!
dial-peer voice 7 voip
 destination-pattern 7....
 session target ipv4:10.1.1.2
!
dial-peer voice 8 pots
```

```
 destination-pattern 8....
 port 1/0:1
!
dial-peer voice 9 voip
 destination-pattern 1201
 session protocol sipv2
 session target ipv4:10.1.1.2
!
dial-peer voice 10 voip
 destination-pattern 1202
 session protocol sipv2
 session target ipv4:10.1.1.2
!
dial-peer voice 11 voip
 destination-pattern ....
 session target ras
!
dial-peer hunt 7
gateway
!
sip-ua
 no oli
!
!
gatekeeper
 zone local gkp1r1 cisco.com 10.1.10.1
 zone remote gkp2r1 cisco.com 10.2.10.1 1719
 zone remote gkp2r2 cisco.com 10.2.20.1 1719
 zone remote gkp3r1 cisco.com 10.3.10.1 1719
 zone remote gkp3r2 cisco.com 10.3.20.1 1719
 zone prefix gkp2r1 21..
 zone prefix gkp2r2 22..
 zone prefix gkp3r1 31..
 zone prefix gkp3r2 32..
 no shutdown
!
!
line con 0
 exec-timeout 0 0
 password 7 131718071F0916
```

```
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
line vty 5 15
 password 7 02140B4E1F031D
 login
!
!
end
```

Lab Exercise 6-3: VoIP with MGCP

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
!
version 12.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
ip subnet-zero
!
!
!
!
!
voice call carrier capacity active
!
!
!
!
!
```



```
!
!
!
!
mta receive maximum-recipients 0
!
controller T1 1/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-24 type fxs-loop-start
!
controller T1 1/1
 framing sf
 linecode ami
!
!
!
!
interface Loopback1
 no ip address
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
 h323-gateway voip interface
 h323-gateway voip id gkp1r1 ipaddr 10.1.10.1 1719
 h323-gateway voip h323-id gwP1R1
!
interface Serial0/0
 ip address 192.168.1.11 255.255.255.0
 encapsulation frame-relay
 frame-relay ip rtp header-compression
!
interface Ethernet0/1
 no ip address
 shutdown
 half-duplex
!
interface Serial0/1
```

```
 ip address 10.1.1.1 255.255.255.0
 encapsulation ppp
 ip tcp header-compression iphc-format
 ip rtp header-compression iphc-format
!
interface FastEthernet3/0
 no ip address
 shutdown
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip http server
!
!
!
route-map eirgp permit 100
!
!
call rsvp-sync
!
voice-port 1/0:1
 output attenuation 0
 connection Tie-line 7
!
voice-port 2/0/0
!
voice-port 2/0/1
!
voice-port 2/1/0
!
voice-port 2/1/1
!
!
```



```
mgcp
mgcp call-agent 192.168.100.100 service-type mgcp version 0.1
!
!
mgcp profile default
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 application mgcpapp
!
dial-peer voice 1102 pots
 destination-pattern 1102
 application mgcpapp
!
dial-peer voice 3 pots
 destination-pattern 555....
 port 2/0/0
 forward-digits all
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
!
dial-peer voice 7 voip
 destination-pattern 7....
 session target ipv4:10.1.1.2
!
dial-peer voice 8 pots
 destination-pattern 8....
 port 1/0:1
!
dial-peer voice 9 voip
 destination-pattern 1201
 session protocol sipv2
 session target ipv4:10.1.1.2
```

```
!
dial-peer voice 10 voip
 destination-pattern 1202
 session protocol sipv2
 session target ipv4:10.1.1.2
!
dial-peer voice 11 voip
 destination-pattern ....
 session target ras
!
dial-peer hunt 7
gateway
!
sip-ua
 no oli
!
!
gatekeeper
 zone local gkp1r1 cisco.com 10.1.10.1
 zone remote gkp2r1 cisco.com 10.2.10.1 1719
 zone remote gkp2r2 cisco.com 10.2.20.1 1719
 zone remote gkp3r1 cisco.com 10.3.10.1 1719
 zone remote gkp3r2 cisco.com 10.3.20.1 1719
 zone prefix gkp2r1 21..
 zone prefix gkp2r2 22..
 zone prefix gkp3r1 31..
 zone prefix gkp3r2 32..
 no shutdown
!
!
line con 0
 exec-timeout 0 0
 password 7 131718071F0916
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
line vty 5 15
```



```
 password 7 02140B4E1F031D
 login
!
!
end
```

Lab Exercise 7-1: QoS

When you complete this lab exercise, your router configuration will be similar to the following, with differences that are specific to your pod.

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Pod1R1
!
boot-start-marker
boot-end-marker
!
enable password 7 08324D40441F17161C
!
voice-card 1
!
no aaa new-model
ip subnet-zero
!
!
!
!
ip cef
!
!
voice call carrier capacity active
!
!
!
!
!
!
!
```

```
!
!
!
!
!
controller T1 1/0
 framing esf
 crc-threshold 320
 clock source internal
 linecode b8zs
 ds0-group 1 timeslots 1-12 type fxs-loop-start
!
class-map match-any AutoQoS-VoIP-Remark
 match ip dscp ef
 match ip dscp cs3
 match ip dscp af31
class-map match-any AutoQoS-VoIP-Control-UnTrust
 match access-group name AutoQoS-VoIP-Control
class-map match-any AutoQoS-VoIP-RTP-UnTrust
 match protocol rtp audio
 match access-group name AutoQoS-VoIP-RTCP
!
!
policy-map AutoQoS-Policy-UnTrust
 class AutoQoS-VoIP-RTP-UnTrust
  priority percent 70
  set dscp ef
 class AutoQoS-VoIP-Control-UnTrust
  bandwidth percent 5
  set dscp af31
 class AutoQoS-VoIP-Remark
  set dscp default
 class class-default
  fair-queue
!
!
!
!
interface Loopback1
```



```
 no ip address
!
interface Ethernet0/0
 ip address 10.1.10.1 255.255.255.0
 half-duplex
 h323-gateway voip interface
 h323-gateway voip id gkp1r1 ipaddr 10.1.10.1 1719
 h323-gateway voip h323-id gwP1R1
!
interface Serial0/0
 no ip address
 encapsulation frame-relay
 frame-relay traffic-shaping
 frame-relay ip rtp header-compression
!
interface Serial0/0.1 point-to-point
 bandwidth 72
 ip address 192.168.1.11 255.255.255.0
 frame-relay interface-dlci 102
  class AutoQoS-VoIP-FR-Serial0/0-102
  auto qos voip
 frame-relay ip rtp header-compression
!
interface Serial0/1
 no ip address
 encapsulation ppp
 ip tcp header-compression iphc-format
 shutdown
 clockrate 72000
 ip rtp header-compression iphc-format
!
router eigrp 100
 network 10.0.0.0
 network 192.168.1.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip http server
ip classless
!
```

```
!
!
ip access-list extended AutoQoS-VoIP-Control
 permit tcp any any eq 1720
 permit tcp any any range 11000 11999
 permit udp any any eq 2427
 permit tcp any any eq 2428
 permit tcp any any range 2000 2002
 permit udp any any eq 1719
 permit udp any any eq 5060
ip access-list extended AutoQoS-VoIP-RTCP
 permit udp any any range 16384 32767
!
map-class frame-relay AutoQoS-VoIP-FR-Serial0/0-102
 frame-relay cir 72000
 frame-relay bc 720
 frame-relay be 0
 frame-relay mincir 72000
 service-policy output AutoQoS-Policy-UnTrust
 frame-relay fragment 90
!
route-map eirgp permit 100
!
!
!
control-plane
!
rmon event 33333 log trap AutoQoS description "AutoQoS SNMP
traps for Voice Drop
s" owner AutoQoS
rmon alarm 33333 cbQosCMDropBitRate.1081.1083 30 absolute
rising-threshold 1 333
33 falling-threshold 0 owner AutoQoS
!
!
voice-port 1/0:1
 output attenuation 0
!
voice-port 2/0/0
!
voice-port 2/0/1
```



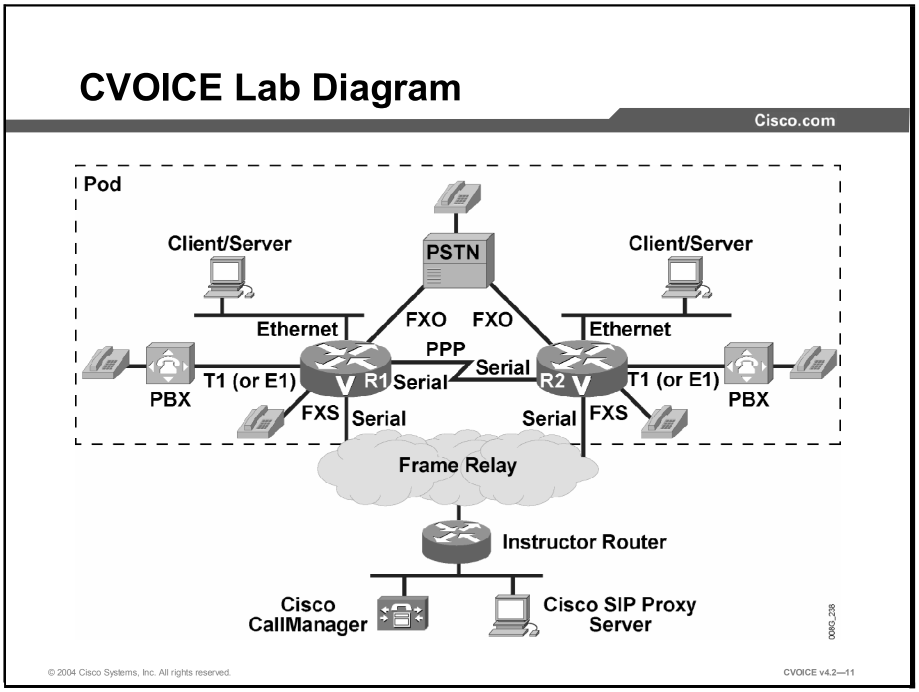
```
!
voice-port 2/1/0
!
voice-port 2/1/1
!
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 1101 pots
 destination-pattern 1101
 port 2/1/0
!
dial-peer voice 1102 pots
 destination-pattern 1102
 port 2/1/1
!
dial-peer voice 3 pots
 destination-pattern 555....
 port 2/0/0
 forward-digits all
!
dial-peer voice 6 pots
 destination-pattern 1151
 port 1/0:1
 forward-digits all
!
dial-peer voice 7 voip
 destination-pattern 7....
 session target ipv4:10.1.1.2
!
dial-peer voice 8 pots
 destination-pattern 8....
 port 1/0:1
!
dial-peer voice 9 voip
 destination-pattern 1201
```

```
 session target ipv4:192.168.1.12
!
dial-peer voice 10 voip
 destination-pattern 1202
 session target ipv4:192.168.1.12
!
dial-peer voice 11 voip
 destination-pattern ....
 session target ras
!
dial-peer hunt 7
gateway
!
!
gatekeeper
 zone local gkp1r1 cisco.com 10.1.10.1
 zone remote gkp2r1 cisco.com 10.2.10.1 1719
 zone remote gkp2r2 cisco.com 10.2.20.1 1719
 zone remote gkp3r1 cisco.com 10.3.10.1 1719
 zone remote gkp3r2 cisco.com 10.3.20.1 1719
 zone prefix gkp2r1 21..
 zone prefix gkp2r2 22..
 zone prefix gkp3r1 31..
 zone prefix gkp3r2 32..
 no shutdown
!
!
line con 0
 exec-timeout 0 0
 password cisco
 logging synchronous
 login
line aux 0
line vty 0 4
 password 7 111B1610031719
 login
 length 0
line vty 5 15
 password 7 02140B4E1F031D
 login
```



```
!
!
!
end
```

Lab Pull-Out Resource



IP Address Assignment

Pod Number	Device	Client	Ethernet	PPP	Frame Relay
1	R1	10.1.10.100	10.1.10. 1	10.1.1.1	192.168.1.11
	R2	10.1.20.100	10.1.20.1	10.1.1.2	192.168.1.12
2	R1	10.2.10.100	10.2.10.1	10.2.2.1	192.168.1.21
	R2	10.2.20.100	10.2.20.1	10.2.2.2	192.168.1.22
3	R1	10.3.10.100	10.3.10.1	10.3.3.1	192.168.1.31
	R2	10.3.20.100	10.3.20.1	10.3.3.2	192.168.1.32
4	R1	10.4.10.100	10.4.10.1	10.4.4.1	192.168.1.41
	R2	10.4.20.100	10.4.20.1	10.4.4.2	192.168.1.42

Host Name

Pod Number	Router	Host Name
1	R1	Pod1R1
	R2	Pod1R2
2	R1	Pod2R1
	R2	Pod2R2
3	R1	Pod3R1
	R2	Pod3R2
4	R1	Pod4R1
	R2	Pod4R2
5	R1	Pod5R1
	R2	Pod5R2
6	R1	Pod6R1
	R2	Pod6R2

Dial Plan Conventions

As with IP addresses, the dial plan convention allows a student to anticipate the number of any telephone in the classroom. The highlights of the strategy include the following points:

- The classroom uses a four-digit dial plan.
- The first digit identifies the pod number (1 to 6).
- The second and third digits identify the device (PSTN=00, R1=10, PBX1=15, R2=20, PBX2=25).
- The fourth digit identifies the telephone.

Classroom Dial Plan

Pod Number		1	2	3	4	5	6
R1		1101 1102	2101 2102	3101 3102	4101 4102	5101 5102	6101 6102
PBX1		1151	2151	3151	4151	5151	6151
R2		1201 1202	2201 2202	3201 3202	4201 4202	5201 5202	6201 6202
PBX2		1251	2251	3251	4251	5251	6251
PSTN	Port 1	555-1001	555-2001	555-3001	555-4001	555-5001	555-6001
	Port 2	555-1002	555-2002	555-3002	555-4002	555-5002	555-6002
	Port 3	555-1003	555-2003	555-3003	555-4003	555-5003	555-6003