Mark Hoeffel

CVOICE

Cisco Voice over Frame Relay, ATM, and IP

Student Guide

Version 1.0

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Text Part Number: 78-5288-01

Table of Contents

Introduction

Introduction

Cisco Voice over Frame Relay, ATM, and IP (Version 1.0) is an instructor-led customer course presented by training partners (TPs) to end-user customers. This four-day course discusses Cisco's 2600, 3600, MC3810 and AS5300 product lines, focusing on reviewing/introducing voice concepts, deepening students' existing technical ability, and introducing new baseline technology.

Upon completion of this training course, students will be able to design, integrate, and configure Voice over Frame Relay, ATM, and IP using the Cisco 2600, 3600 MC3810 and AS5300.

Course Objectives

Upon completion of the course, the student will be able to:

- Given voice and telecommunications fundamentals, apply the principles and concepts to develop a process for integrating voice and data networks
- Design, configure, integrate, and optimize an enterprise network in remote branch and regional offices by using integrated access technology that combines voice and data transmission over Frame Relay, ATM, and IP connections, using Cisco's 2600, 3600, MC3810, and AS5300 multiservice access devices
- Appraise existing branch and regional office voice network and services, and choose the optimum transmission method for voice traffic: Frame Relay, ATM, or IP
- Analyze existing voice hardware/software, and select which Cisco multiservice access device (2600, 3600, MC3810, or AS5300) would best serve the needs
- Configure Voice over Frame Relay, ATM, or IP using Cisco's IOS™software

Chapter Objectives

Objectives

- Introduce Cisco Voice over Frame Relay, ATM, and IP course to students, and explain the general administration procedures for the class
- Given a brief overview of Cisco Voice over Frame Relay, ATM, and IP hardware, describe the voice-specific features for this hardware

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Objectives (cont.)

- Explain the different functions of Cisco voice-capable routers and demonstrate where these routers fit in a data network
- Illustrate Cisco voice-capable routers with some examples of voice over IP for the 26/3600 and Frame Relay/ATM for the MC3810

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Prerequisites

- At a minimum, attended the Introduction to Cisco Router Configuration (ICRC) course, the Advanced Cisco Router Configuration course (ACRC)
- Hands-on experience installing Cisco routers using Cisco IOS™ software

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Prerequisites

The CVOICE course focuses on introducing new baseline technology and on deepening students' existing technical ability. To fully benefit from CVOICE, students must possess prerequisite skills and knowledge gained from successfully completing other Cisco educational offerings.

- Students must have experience configuring Cisco IOS software. At a minimum, they should have attended the ICRC (Introduction to Cisco Router Configuration) and the ACRC (Advanced Cisco Router Configuration) courses.
- Having hands-on experience installing and configuring Cisco routers is a requirement.

Prerequisites (cont.)

- Hands-on experience installing Frame Relay, ATM, or ISDN networks
- Hands-on experience installing voice technology components
- Understanding of voice technology terms

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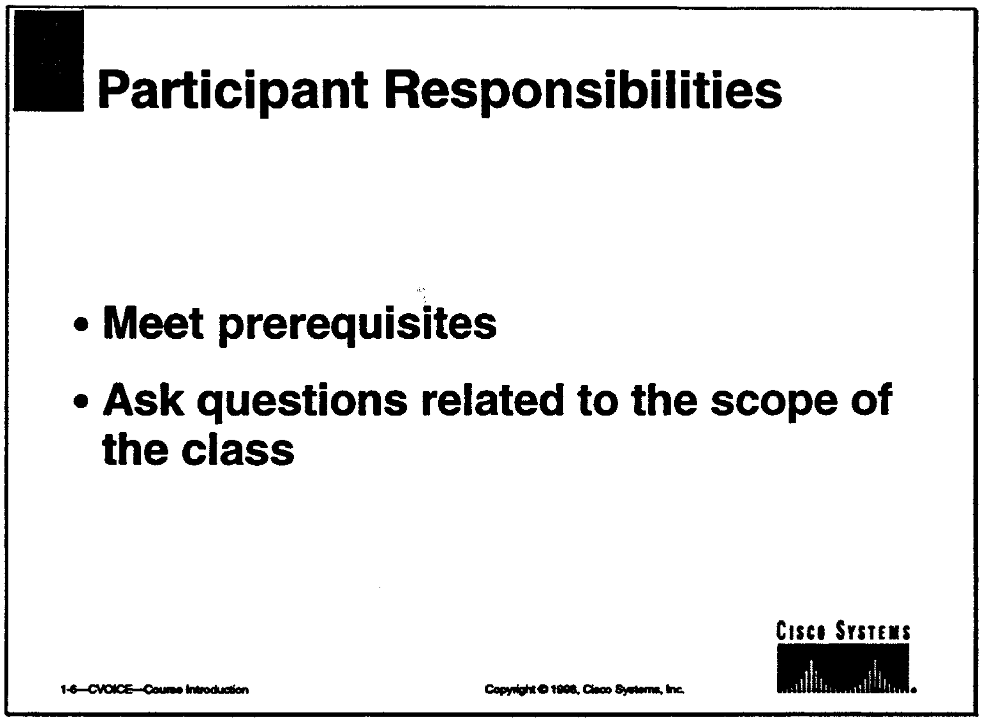
- Having hands-on experience installing Frame Relay, ISDN, or ATM networks, and familiar with these technologies' terms is also important.
- In addition, students should have taken a voice or telecommunications class from third-party trainers or studied voice technology using self-paced training lessons.

Note It is imperative that the prerequisites for this course be met by the student before attending the class.



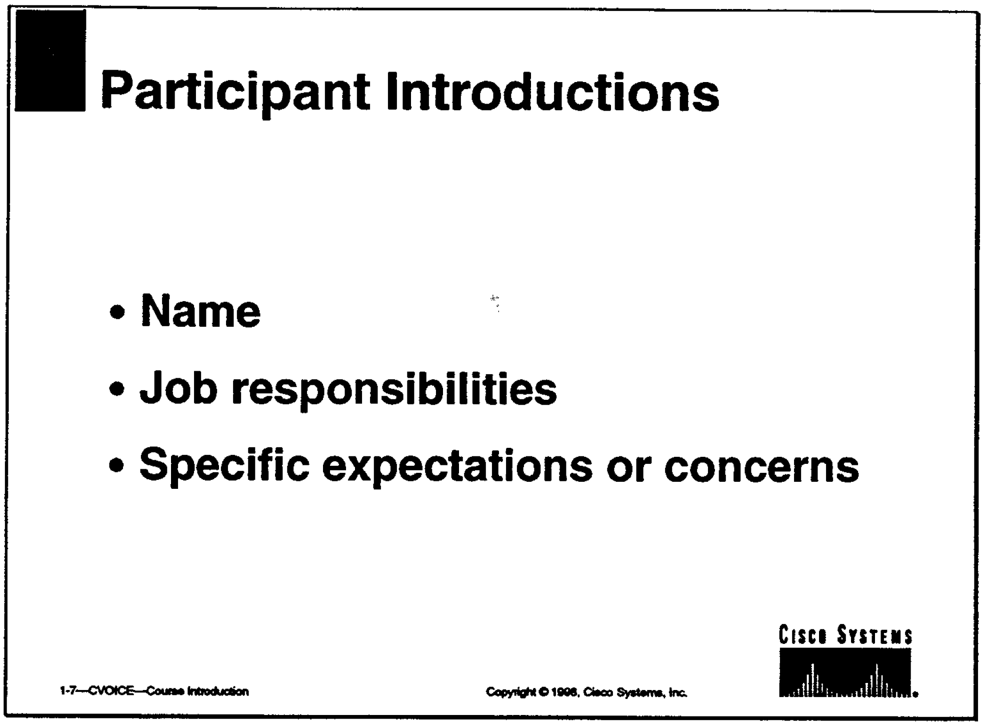
Participant Responsibilities

Participants must have fulfilled the prerequisites.

Some experience with voice network technologies will be beneficial, however, we will cover some basic voice functionality as well as voice commands as they pertain to the configuration and operation of the Cisco MC3810 and 3620 in particular.

Participants are encouraged to ask any questions relevant to the course materials. This course, however, covers only the current 2600, 3600, MC3810, and AS53000 feature set and does not address future enhancements in any detail.

The instructor or participants may have some pertinent information or questions covering future MC3810 features. Please bring these topics up during breaks or after class and the instructor will try to answer the questions or direct participants to an appropriate information source and pass the information along to all participants in a timely manner.



Participant Introductions

Emphasize to the instructor what you want to learn from this course.

General Administration

Class Related

- Sign-in sheet
- Length and times
- Attire
- Participant materials

Facilities Related

- Site emergency procedures
- Break and lunch room locations
- Rest rooms
- Telephones/faxes

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General Administration

The instructor will discuss each of these administrative issues in detail so that students will know exactly what to expect from both the class and facilities.

Sources of Information: MC3810

- Manuals
- Web sites
- Release notes
- CD-ROMs

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Other Sources of Information: MC3810

Other sources of MC3810 information are as follows:

- *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide*
- *Cisco MC3810 Multiservice Concentrator Software Configuration Guide*
- *Cisco MC3810 Multiservice Concentrator Command Reference Guide*
- *Cisco 3800 Series Release Notes*

All of these documents can all be found on the following URL:

http://www.cisco.com/

The Cisco Connection Documentation CD-ROM is available as a single CD and through an annual subscription.

Note When accessing CCO, use the search function and search on terms such as "Voice," "Voice over IP," "Voice over ATM," "Voice over Frame Relay," or the specific hardware you want, i.e., "MC3810." These new technologies keep changing and Cisco keeps publishing new documentation.

Sources of Information: 3620

- Manuals
- Web sites
- Release Notes
- CD-ROMs

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Other Sources of Information: 3620

More information about the 3600 series can all be found on the following URL:

http://www.cisco.com/

Chapter Descriptions

- Chapter 1: Introduction
- Chapter 2: Analog voice technology
- Chapter 3: Digital voice technology
- Chapter 4: Cisco voice products
- Chapter 5: General voice configuration parameters

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Chapter Descriptions

Chapter 1 is an introductory chapter. General administration information is covered. In addition, the chapter gives a brief description of the various Cisco hardware that is covered in the course plus a generic overview of where this voice over Frame Relay, ATM, and IP hardware fits into the overall network picture. The chapter ends with examples of Cisco voice capable routers in actual network situations.

Chapter 2 is a review of analog voice fundamentals and basics. You should be generally familiar with this material. However, task analysis information indicates that substantial numbers of students are not familiar with basic voice concepts. This chapter will ensure that the student has at least a basic understanding of analog voice and telecommunications fundamentals.

Chapter 3 takes a closer look at digital voice technology. This module builds upon the knowledge and information acquired in Chapter 2. The module starts off with an overview of digital signaling, then covers digital framing formats for T1 and E1, how analog signals are digitized, different channel signaling types, voice signal compression techniques, and measuring voice quality. The chapter ends with an overview of ISDN.

Chapter 4 gives an in-depth overview of Cisco hardware that will be covered in this class. Continuing from where the brief overview of Cisco equipment left off in Chapter 1, this chapter will cover how this equipment handles voice over IP, Frame Relay, and IP in more depth. The chapter will also cover how to properly install this equipment, including cabling. The specific equipment covered is 2600, 3600, MC3810, and AS5300.

Chapter 5 presents the generic Cisco IOS voice configuration commands common across all voice-over technologies and how to configure a Cisco multiservice access device for voice over Frame Relay, ATM, and IP. Subsequent chapters will cover specific information and Cisco IOS voice configuration commands for Frame Relay, ATM, and IP.

Chapter Descriptions (cont.)

- Chapter 6: Voice over Frame Relay
- Chapter 7: Voice over ATM
- Chapter 8: Voice over IP
- Chapter 9: Network design guide
- Appendix

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Chapter 6 presents an overview of Frame Relay voice and the Cisco IOS configuration commands for configuring a Cisco MC3810 for voice over Frame Relay. The chapter ends with a lab for the students to actually configure the MC3810 for voice over Frame Relay.

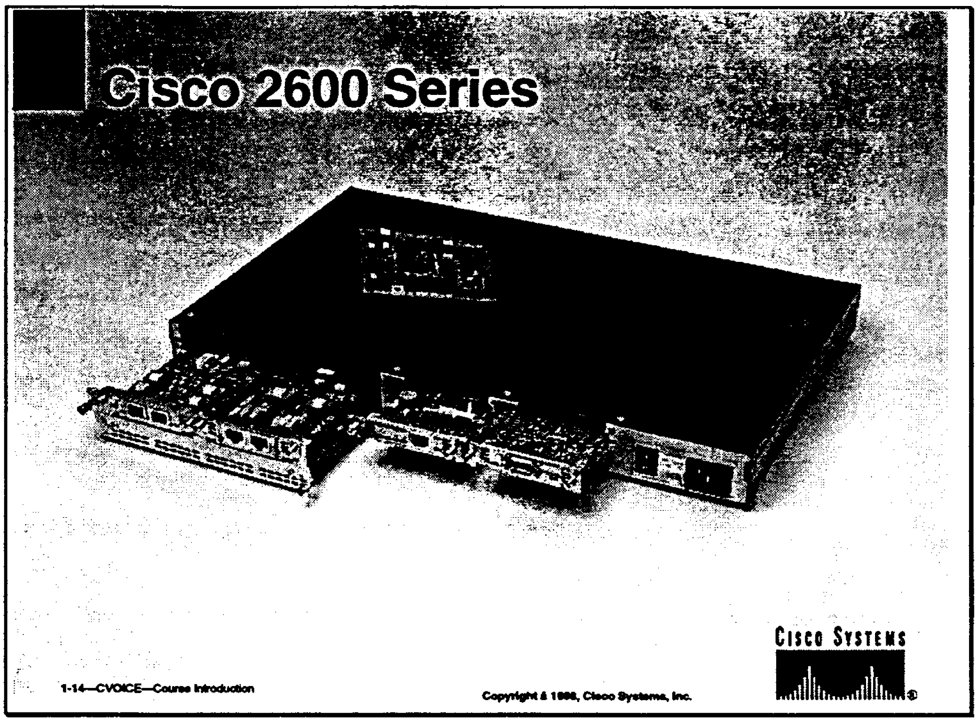
Chapter 7 presents an overview of ATM voice and the Cisco IOS configuration commands for configuring a Cisco MC3810 for voice over ATM. The chapter ends with a lab for the students to actually configure the MC3810 for voice over ATM.

Chapter 8 presents an overview of IP voice and the Cisco IOS configuration commands for configuring a Cisco 3600 for voice over IP. The chapter ends with a lab for the students to actually configure a 3620 for voice over IP.

Chapter 9 presents a six-step network design process to help students perform an audit of their existing voice and data equipment, set objectives and goals for their network, determine which available technology (Frame Relay, ATM, or IP) is best for their particular network, and formulate a technical network design using Cisco's multiservice access devices of the 2600, 3600, MC3810, or AS5300.

Appendix A contains lab information and diagrams as well as various technical information for the course on the different technologies covered in the course.

CVOICE Hardware



Cisco 2600 Series

The following are the features, functions, and benefits of the 2600 series routers:

- Field-upgradable interfaces shared with Cisco 1600 and 3600 series
- Full Cisco IOS software support (Release 11.3(2) and later)
- Data/voice/fax integration, dial and VPN access
- Flexible voice interfaces (VICs):
  - FXS
  - 2, 4 wire E&M types I, II, III, V
  - FXO
  - BRI VIC committed
- Diverse signaling:
  - Delay, immediate, and wink start
  - Loop and ground start
- Any combination of WAN interface card in two available slots. Over 100 unique LAN/WAN interface combinations using WAN interface cards alone.

Note The following URL provides information on the 2600 series: http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/cis2600/software/index.htm

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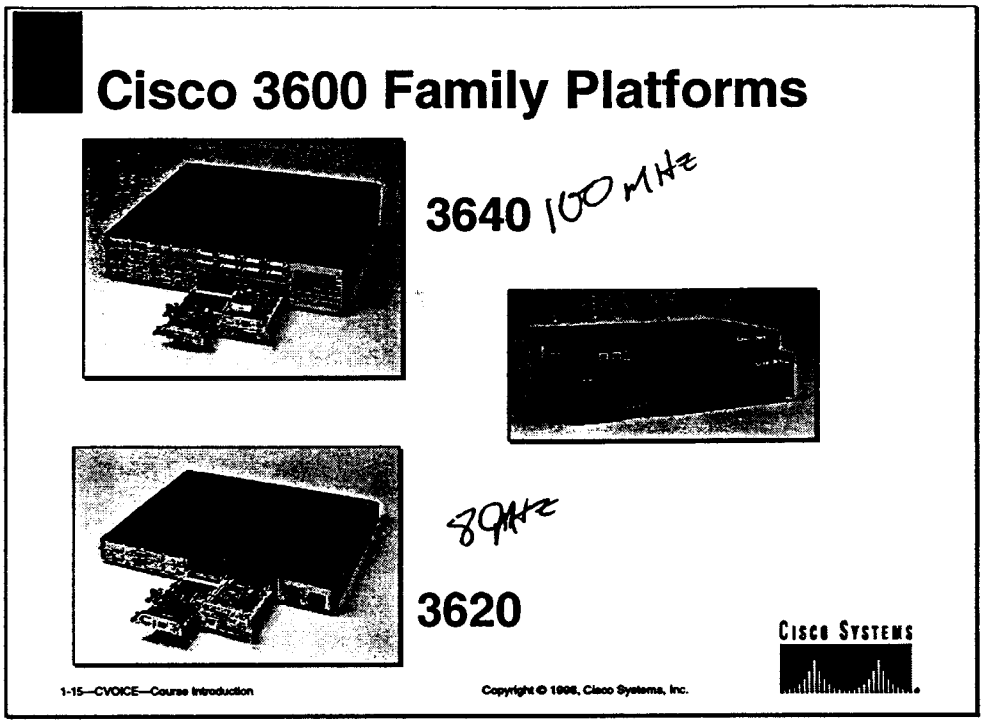

Cisco 3600 Series

The Cisco 3600 addresses three distinct market segments:

- LAN to LAN routing
- Dial access
- Multiservice (voice, video, and data)

The features of the Cisco 3620 router include the following:

- High-performance 80-MHz Reduced Instruction Set Computer (RISC) processor
- Slots for:
  - Two network modules
  - Two Personal Computer Memory Card International Association (PCMCIA) cards
  - Flash memory
- Four slots for DRAM, configurable as shared memory or main (processor) memory
- Supports connection to an optional external redundant power supply
- High-speed console and auxiliary ports (up to 115.2 kbps)

Note The following URL provides information on VoIP for the 3600:
http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip3600/index.htm

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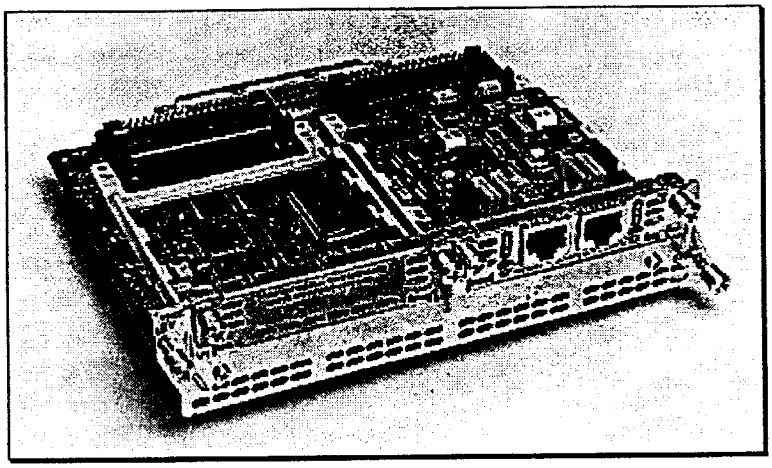
Digital Signal Processor
DSP is on the NM

Voice/Fax Network Module for the Cisco 2600 and 3600



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The 2600/3600 voice/fax network module has the following features and benefits:

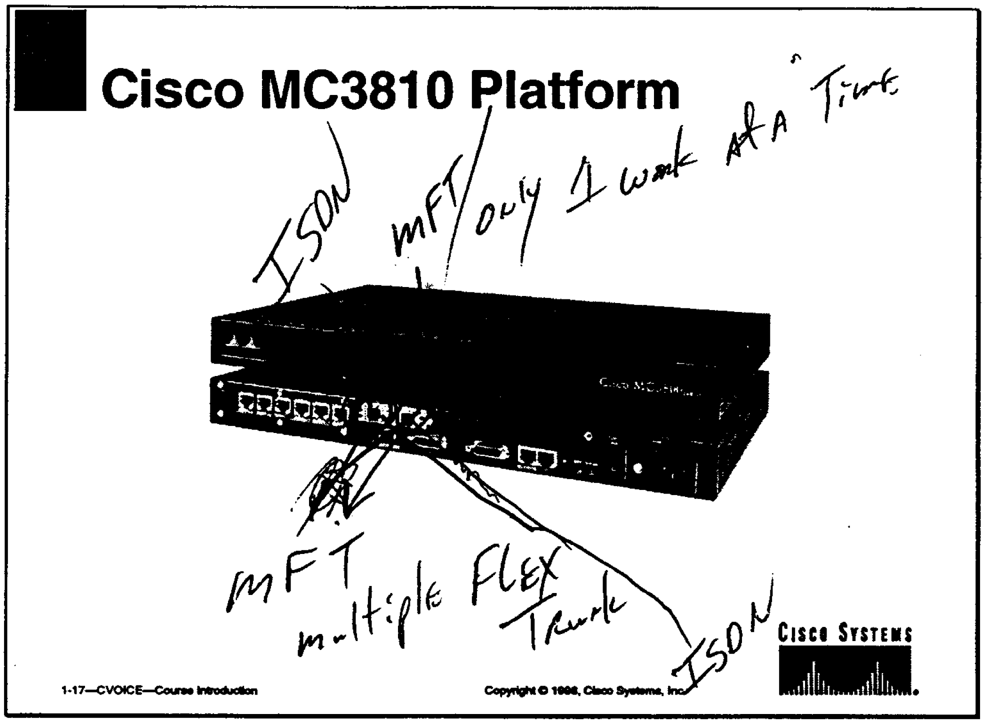
- Voice/fax over IP
- Flexible interfaces: FXS, E&M, FXO
- High-performance digital signal processor (DSP) architecture:
  - CSA-CELP
  - PCM
  - Group III fax relay

Speech-coding features are as follows:

- Silence suppression
- Echo cancellation
- Adaptive jitter buffering

Cisco IOS QoS features are as follows:

- Resource Reservation Protocol (RSVP), weighted fair queuing (WFQ), fragmentation, and interleave
- Compatibility with H.323 standard for audio- and video conferencing
- Works with existing phone and fax equipment
- Requires minimum Cisco IOS Release 11.3(1)T. Voice features require a "Plus" feature set.



Cisco MC3810 Platform

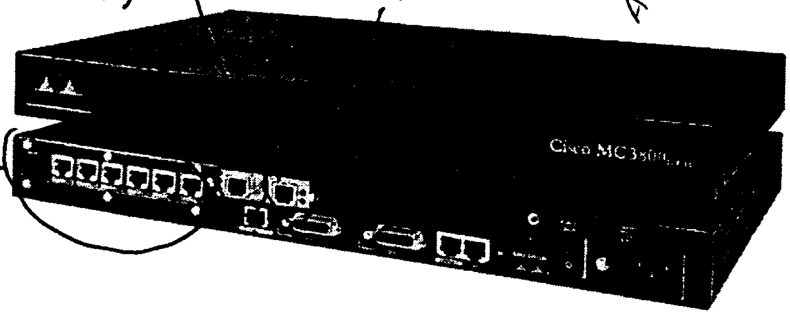
The Cisco MC3810 integrates LAN, synchronous data, legacy, voice, video, and fax traffic for transport over a public or private Frame Relay, ATM, or TDM network. The MC3810 optimizes network bandwidth by multiplexing voice and data on the same circuit or physical interface.

The MC3810 has a single Ethernet port and two serial ports that support speeds up to 2 Mbps. The analog model of the MC3810 has six voice ports, and the digital version houses a single digital voice access port (T1/E1). The analog configuration provides up to six compressed voice channels and the digital configuration offers up to 24 compressed voice channels. If voice is passed via TDM channels, up to 30 channels are available. Combinations of compressed and PCM voice are also possible.

MC3810 Voice Features

ISDN
MFT
24-DS0's
T-1
ATM

Swappable



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Built on Cisco IOS software, the Cisco MC3810 features complete voice switching and call handling.

The Multiflex Trunk supports Frame Relay, ATM, or TDM. It even allows customers to divide a trunk such that part of it carries TDM traffic for high-bandwidth video conferencing, for example, while other DS0s carry Frame Relay traffic and still others carry PSTN traffic. Frame Relay to ATM interworking is available.

Features include the following:

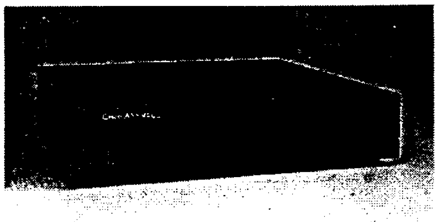
- G.729 and G.729A voice compression over Frame Relay, ATM, or HDLC. G.729A allows two calls per DSP.
- Multiflex trunking for Frame Relay, ATM, or TDM trunk
- Frame Relay to ATM network interworking
- Flexible voice interfaces: FXS, FXO, E&M

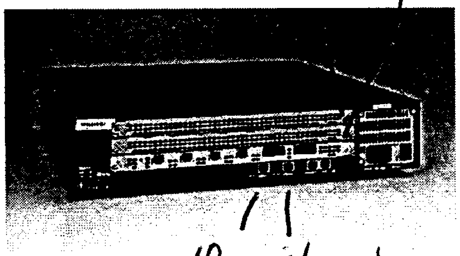
The MC3810 can connect to the following types of telephone systems:

- Analog telephone set via 2-wire connections
- Analog PBX via 2- or 4-wire interfaces
- Key system via 2- or 4-wire connections
- Digital PBX via T1/E1

Cisco AS5300 Platform

Modems or Voice Cards





10 10/100 Auto Sense

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Cisco AS5300 Platform

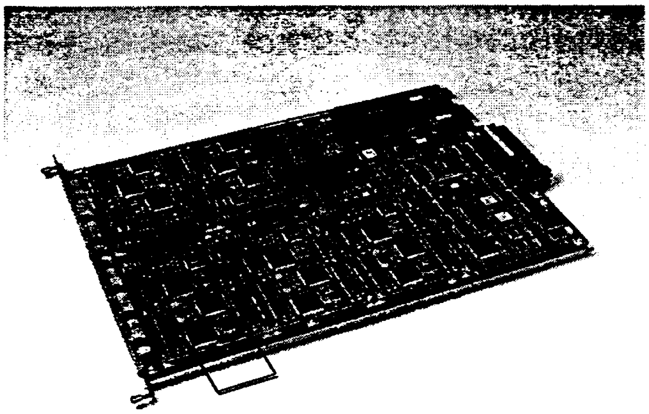
1- 4- 8- Ports

- Up to four CT1/CE1/PRI
- One 10-MB and one 10/100-MB Ethernet interface
- Cisco Inter-Switch Link Protocol (ISL) on 10/100 interface
- Fast 150-MHz R4700 RISC CPU
- Available AC or DC power options
- Two feature card slots for voice/fax or data/fax carrier cards
- Up to ~~60~~ voice calls per AS5300

Now 120

Note The following URL provides information on the AS5300 series: http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip5300/index.htm

Cisco AS5300 Voice Feature Card



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30 DSP per Card

FAX Require 1 DSP per FAX

G.729A up to 2 calls per DSP

The following are the features of the AS5300 voice card:

- Voice/fax carrier card:
  - One or two per AS5300
  - Accepts one to five DSP modules
  - No external interfaces
  - Onboard R4700 CPU
- DSP modules:
  - Snap-on DSP module PCBs
  - Six DSPs per module
  - TI542
  - G.729 CSA-CELP–8K
  - G.711 PCM
  - Group III fax relay

Note VoIP 5300 configuration examples can be found at the following URL:
http://www.cisco.com/univercd/cc/td/doc/product/access/nubuvoip/voip5300/examples.htm

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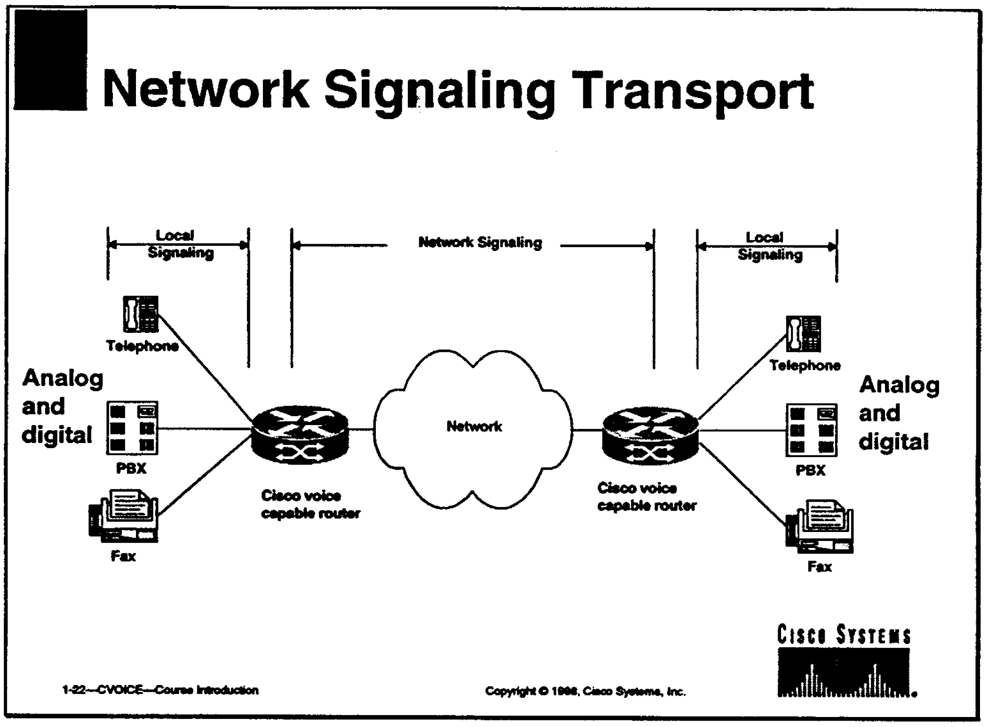
Where Cisco Voice-Capable Routers Fit in the Network



Voice signaling is accomplished by interpreting and responding to the different types of signaling supported by Cisco voice capable routers. Supported signaling includes ABCD bits, ground start, loop start, etc.

Instead of passing the signaling information end to end, in its native form, the signaling is locally acknowledged by the Cisco voice capable router. The Cisco router then uses a proprietary signaling protocol (based on Q.SIG) to contact the remote Cisco router. The remote Cisco voice capable router then invokes the proper local signaling to complete the call.

Cisco Voice Connection Types

1. Local	**Calls within the same system**
2. On-Net	**Interoffice calls**
3. Off-Net	**Connection to "outside line" (to PSTN)**
4. PLAR	**Automatically dials an extension**
5. PBX-to-PBX	**Tie-line type connection between PBX**
6. On-Net to Off-Net	**Reroutes calls off net**

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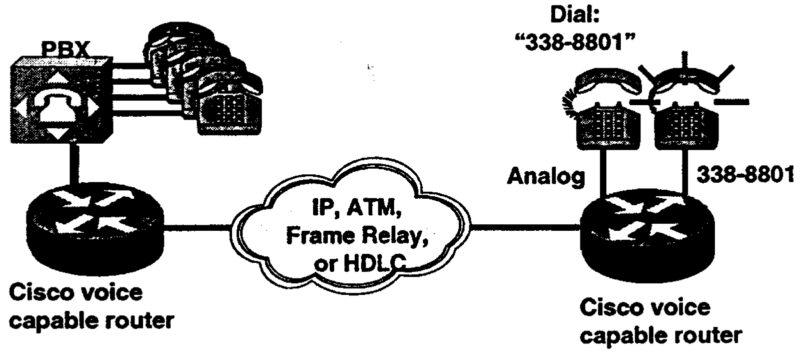
Cisco Voice Connection Types

The connection types listed in the figure above show the voice functions of Cisco routers, and how Cisco voice capable routers fit in the data network.

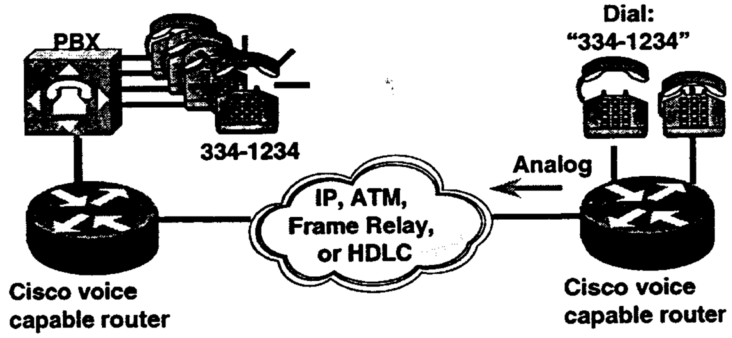
Each of these voice connection types will be discussed on the following pages.

Local Calls (Intraoffice Calls)



Cisco voice capable routers permit calls from one phone to another, both of which are attached to the same voice capable router.

On-Net Call Setup (Interoffice Calls)



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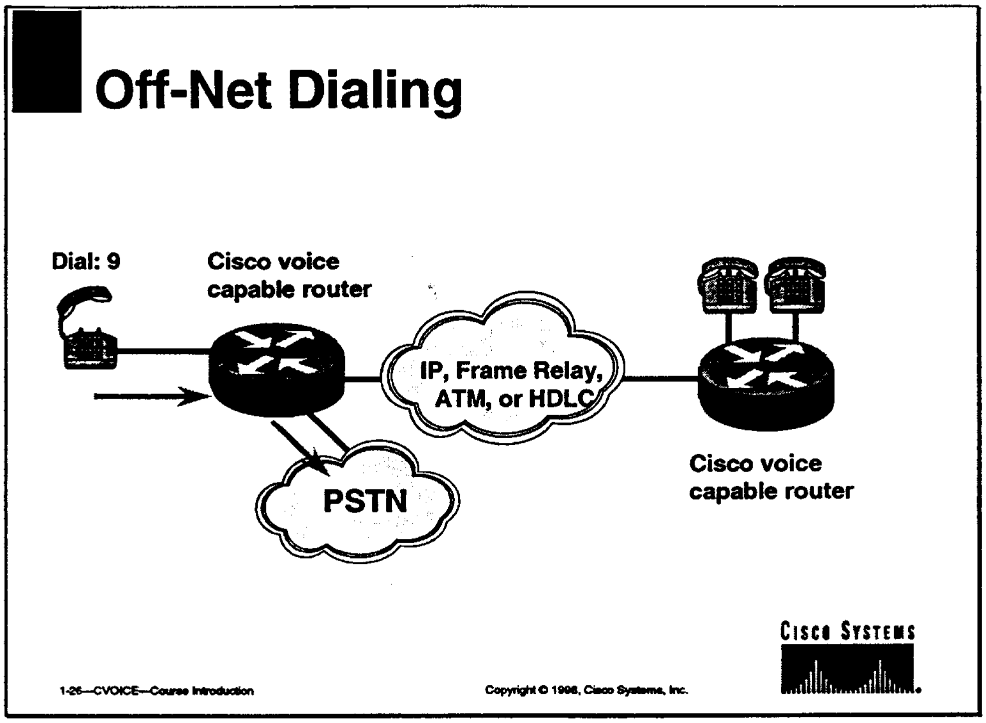
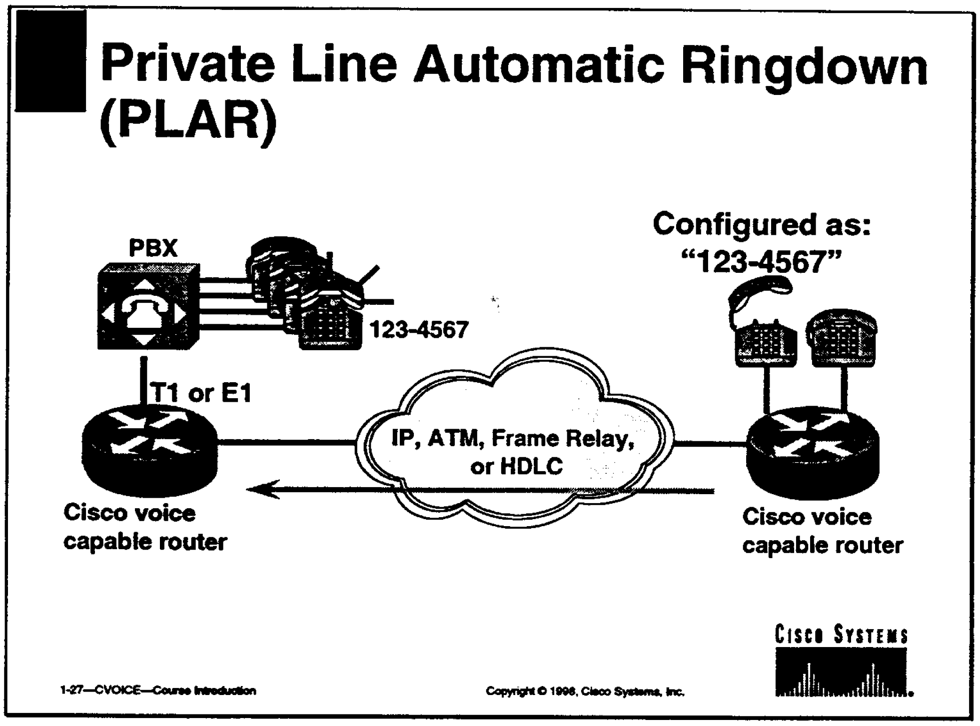
Calls can be made from a phone attached to a Cisco voice capable router, across the date network, to another phone also attached to a voice capable router.

The call is routed based upon its "destination pattern." Extension digits can also be passed to the PBX.



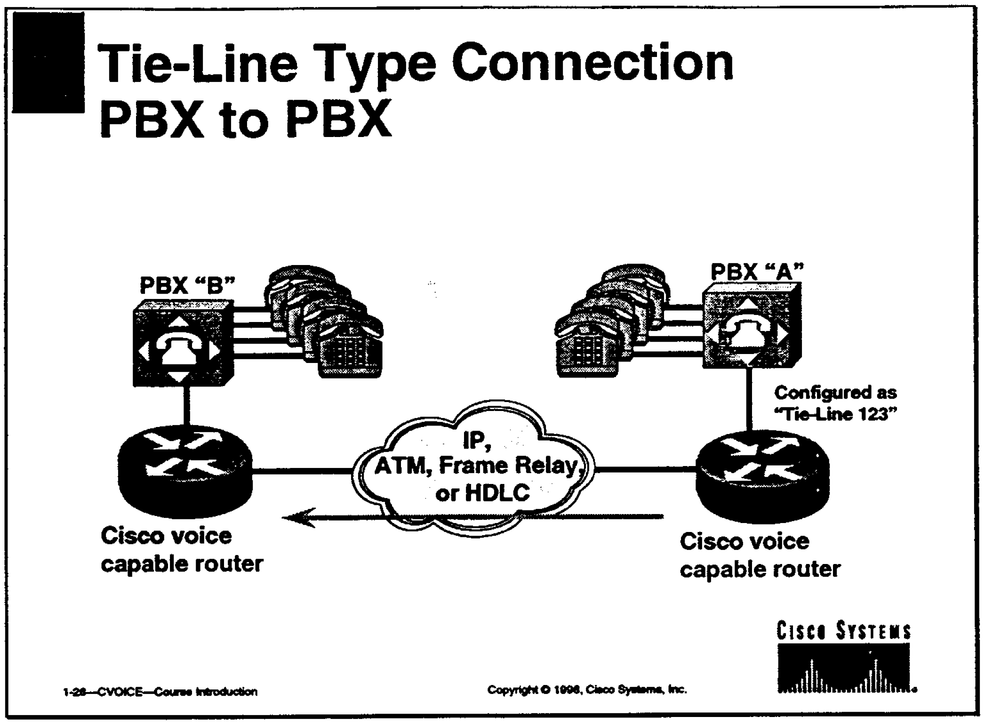
In off-net dialing, the user dials 9 from an analog phone directly connected to a Cisco voice capable router to gain access to the public switched telephone network (PSTN), or other service provider voice network. When the call reaches the PSTN, a second dial tone can be heard. At this point, the user dials the destination pattern (i.e., phone number) of the party it wishes to reach.



Private Line, Automatic Ringdown (PLAR) automatically connects a telephone to a second phone, as soon as the first phone is lifted from the cradle (that is, goes off-hook). There is no need to dial any digits, because the first phone is automatically connected to the second phone.

A data analogy is: simplex transmission—in one direction only.

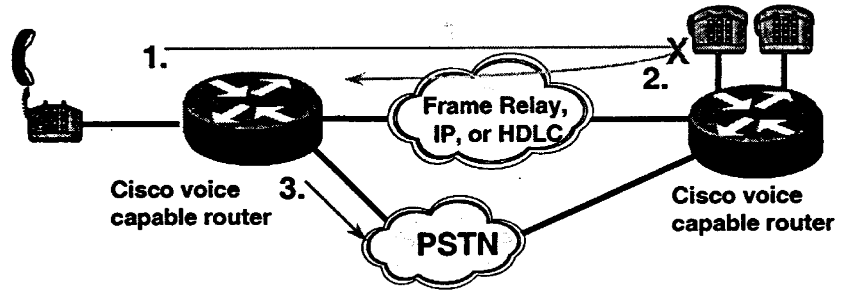
- Used for one phone directly connected to another phone. No chance to dial any other extension.
- Supports PBX autodial functions



When PBX "A" seizes a channel, the Cisco voice capable router inserts the correct destination pattern into the signal before dialing.

PLAR is virtually the same thing. In this case, unlike PLAR, the call is PBX to PBX, instead of handset (phone) to handset (phone).

On-Net to Off-Net Call Rerouting



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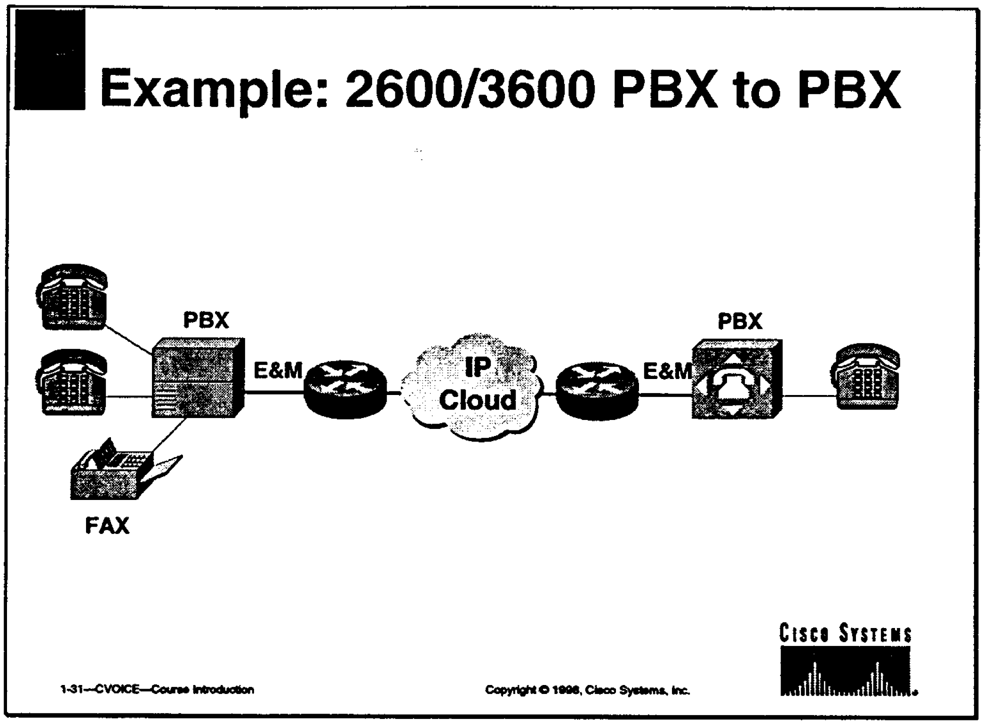
Because on-net resources are limited, the Cisco voice capable router must handle various types of call situations.

1. The phone on the left attempts to make a call, on-net. The Cisco voice capable router can respond in different ways.
2. If resources are not available to complete the call, the Cisco router on the far end can send "busy-back" (a fast busy signal) to a user to discourage on-net usage.
3. The Cisco router on the near end can seize an off-net trunk and place the call from its routing table over the PSTN. The router cannot reroute once a channel is "seized."

Note Hook flash is NOT supported.

Examples of Cisco Voice-Capable Routers



Use an E&M Voice Interface Card (VIC) when connecting PBX to PBX, or switch to switch.

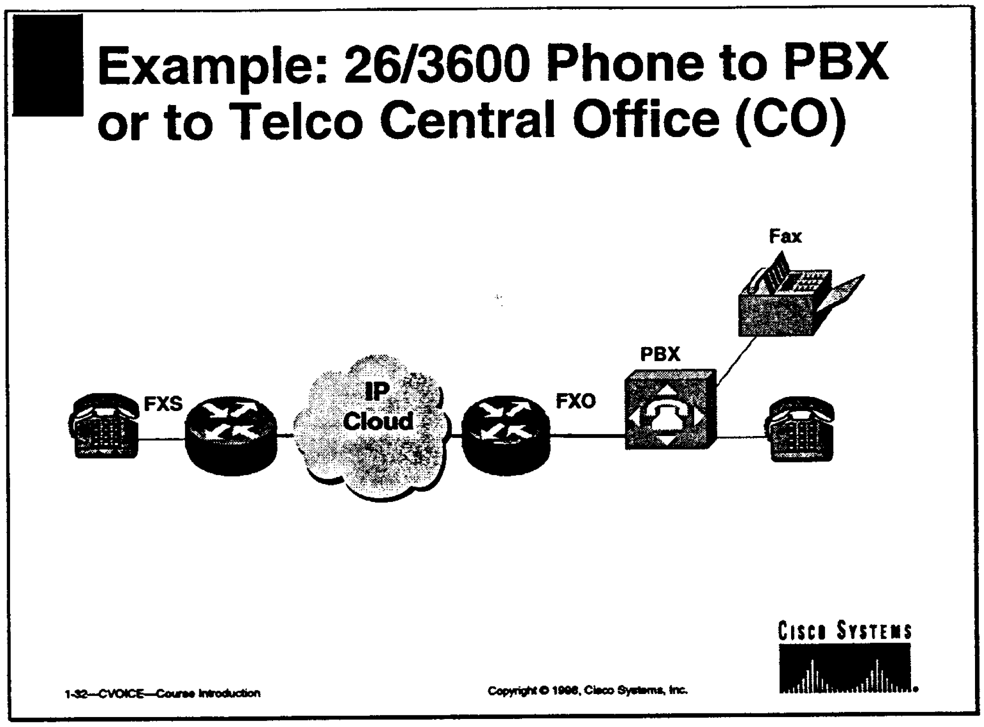
The following are URLs that will help you to configure voice over IP:

- *VoIP Software Configuration Guide*:

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/113t/113t_1/voip/index.htm

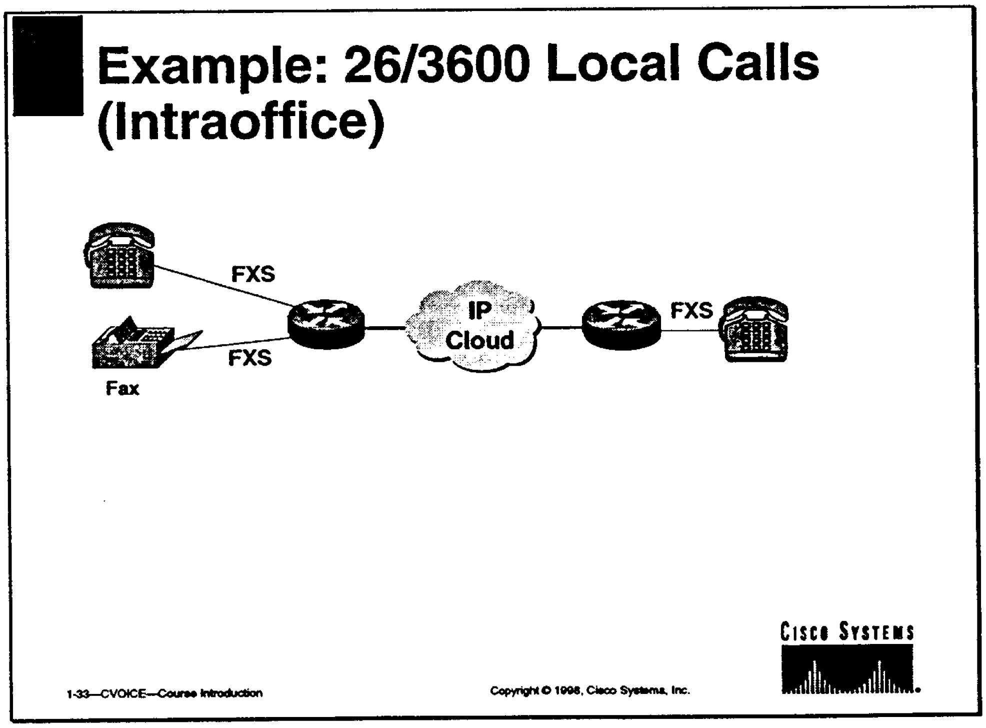
- VoIP configuration examples:

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios113ed/113t/113t_1/voip/examples.htm



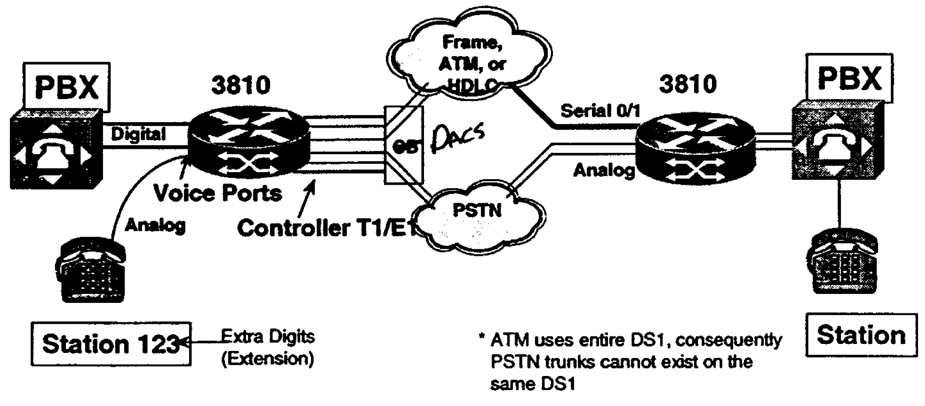
Use a foreign exchange office (FXO) VIC when connecting to the station side of a PBX or directly to the telco (telephone company) CO (central office).

FXOs allow for off-premise connections.



Use a foreign exchange station (FXS) VIC when connecting directly to a standard phone or fax machine.

Example: MC3810 Connection Capabilities



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The Cisco MC3810 connects to any standard PBX switch, key system, or telephone, and provides up to 30 channels of voice, with compression down to 8 kbps using the standard G.729 CSA-CELP algorithm. The MC3810 provides echo cancellation for all voice channels and uses voice activity detection (VAD), which halts voice traffic during the silence between words and sentences in speech.

The MC3810 supports an extensive array of call-handling capabilities for voice connections. The MC3810 can support tie-line and ring-down modes. It can also support Dual Tone Multifrequency (DTMF) digit-based per-call switching, using dialed digits to select destination sites and network calls without needing to go through tandem voice switching.

At small sites, telephones and trunks can be connected to the MC3810 and it acts as a voice switch locally, obviating the need for Centrex, key, or PBX switching. Voice and data can be sent out to the network over multiflex trunking, TDM connections, or serial port trunking.

Summary

Summary

- Introduced Cisco Voice over Frame Relay, ATM, and IP course to students, and explained the general administration procedures for the class
- Gave a brief overview of Cisco voice over Frame Relay, ATM, and IP hardware, describing the voice-specific features for this hardware

1-35—CVOICE—Course Introduction

Chapter 1 covered general administration information. The second section of the chapter gave a brief description of the various Cisco hardware that is covered in the course. The last section was a generic overview of where this Cisco voice over Frame Relay, ATM, and IP hardware fits into the overall network picture.

Summary (cont.)

- Explained the different functions of Cisco voice-capable routers and demonstrated where these routers fit in a data network
- Illustrated Cisco voice-capable routers with some examples of voice over IP for the 26/3600, and Frame Relay/ATM for the MC3810

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2

Introduction to Analog Telephony

Objectives

Objectives

Upon completion of this chapter, you will be able to perform the following tasks:

- **Given an analog telephony network, identify the role of each component within the network. The components are telephones, lines, loops, switches, and trunks.**
- **Given an analog telephony network, describe the telephony signaling on lines and trunks**
- **Given a telephony network, trace the path of a telephone call**

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Chapter Content

Voice can be transmitted across a traditional telephony network in two forms:

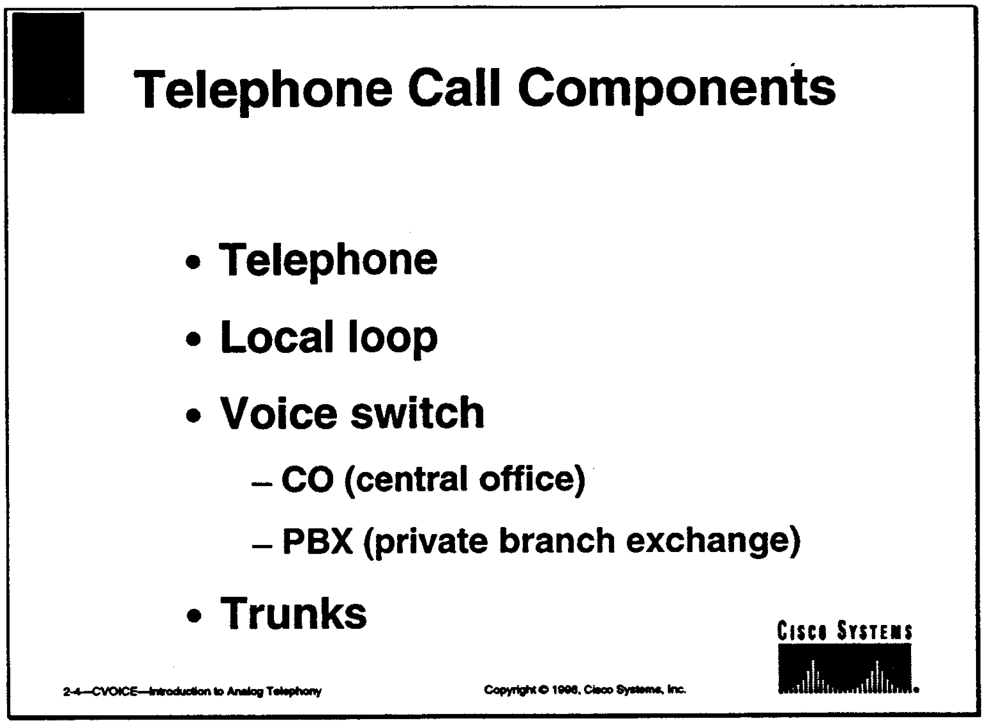
- Analog
- Digital

Chapter 2 describes the telephone network with a focus on analog technology. You will then learn about the evolution of digital voice technology in Chapter 3.

This chapter includes an introductory overview of the basic telephone handset and its internal components. Analog phone line types, switch functions, trunk types, and trunk signaling are then described.

The chapter outline follows:

- Telephone Call Components
- Telephone Components
- Telephone Signaling
- Local Loops
- Voice Switches
- Trunks
- Trunk Signaling
- Telephone Call Procedure
- Written Exercise
- Summary



Telephony Hardware

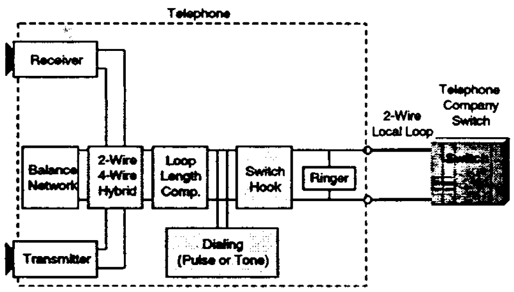
In this chapter, you will learn about where the following components fit into the telephone network:

- Telephone
- Local loop
- Voice switch
- Trunks

You will also learn about the signaling protocols the devices use to communicate.

Telephone Components

- **Handset (receiver and transmitter)**
- **Switch hook**
- **Hybrid (2-to-4 wire converter)**
- **Sidetone**
- **Dialer**
- **Ringer**



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2-6—CVOICE—Introduction to Analog Telephony

Telephone Set Components

Your telephone includes the following components:

- Handset —This is the part of the telephone you hold in your hand to speak and listen to a conversation. Inside the handset are a transmitter and a receiver. You speak into the transmitter and listen from the receiver.
- Switch hook —This is the lever that is pushed down when the handset rests on its cradle (on-hook). When you lift the handset to place a call you release the switch hook and it pops up. The circuit is now off-hook and current flows through the telephone. When the telephone is placed back on-hook, current flow ceases.
- Hybrid 2-to-4 wire converter —Four wires, organized in two pairs, run from the handset, one pair from the transmitter and another pair from the receiver. The hybrid provides the conversion between the 4-wire handset and the 2-wire local loop. The converter is the communications bridge between the handset equipment and the 2 wires to the telephone company.
- Sidetone —This is a planned, audible result emanating from the hybrid in the phone, through which a portion of speech is allowed to bleed over into the ear piece during a conversation so that users can judge how loudly they are speaking.
- Dialer —This is the touch pad or rotary dial that signals the telephone company that you are placing a call. When you push the buttons on a touch pad or spin the dial on a rotary telephone, you send a signal to the telephone company, specifying the location you are calling.
- Ringer —When someone is trying to call you, the telephone company notifies you by sending voltage through the wires to your telephone set. The voltage triggers a device, the ringer, that makes a ringing sound.

Telephone Signaling

- **Supervisory signaling**
  - Release connection (on-hook)
  - Request for service (off-hook)
  - Ringing
- **Addressing (dialed digits)**
  - Pulse
  - DTMF

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Telephone Signaling

When you use your telephone to place or receive telephone calls, you and your telephone company must communicate each other's intentions through basic signaling. Basic signaling is the analog communication that takes place between the phone and switch.

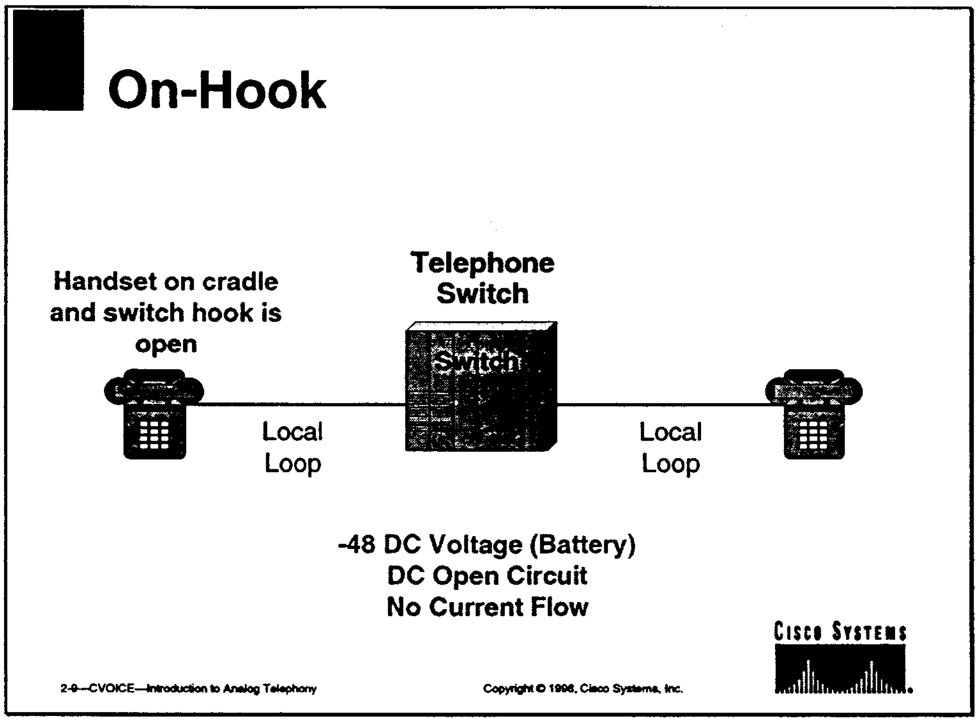
This section addresses two types of signaling:

- Supervisory
- Address

Supervisory Signaling

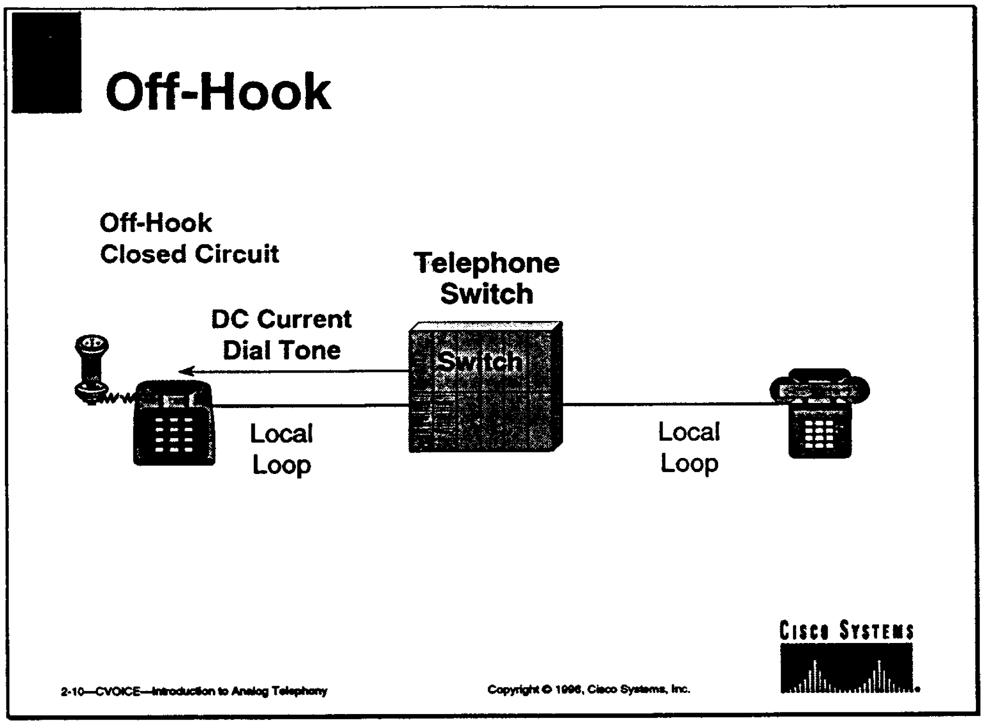
You and your telephone company notify each other of call status with audible tones and an exchange of electrical current. This exchange of information is termed supervisory signaling. Different types of supervisory signaling are:

- On-hook
- Off-hook
- Ringing



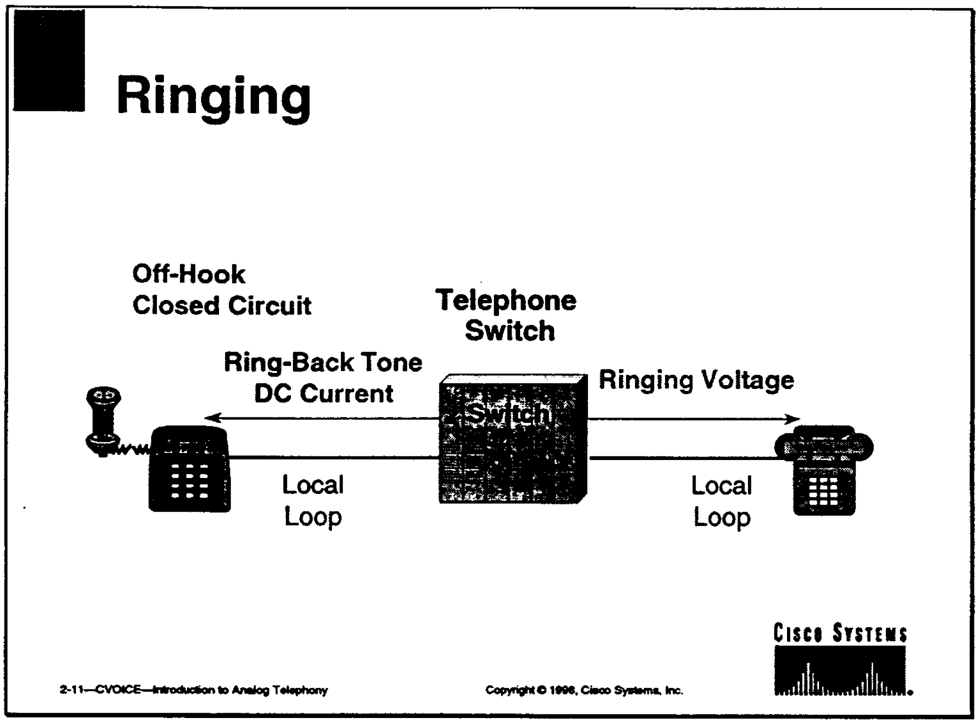
On-Hook

Allowing the handset to rest on the cradle opens the switch hook and current is thereby prevented from flowing freely through the phone. Regardless of the actual type of signaling used, the circuit is said to have gone on-hook when the handset is placed on the cradle and the switch hook is toggled to an open state, which prevents the current from flowing through the phone. Only the ringer is active when the telephone is in this position.



Off-Hook

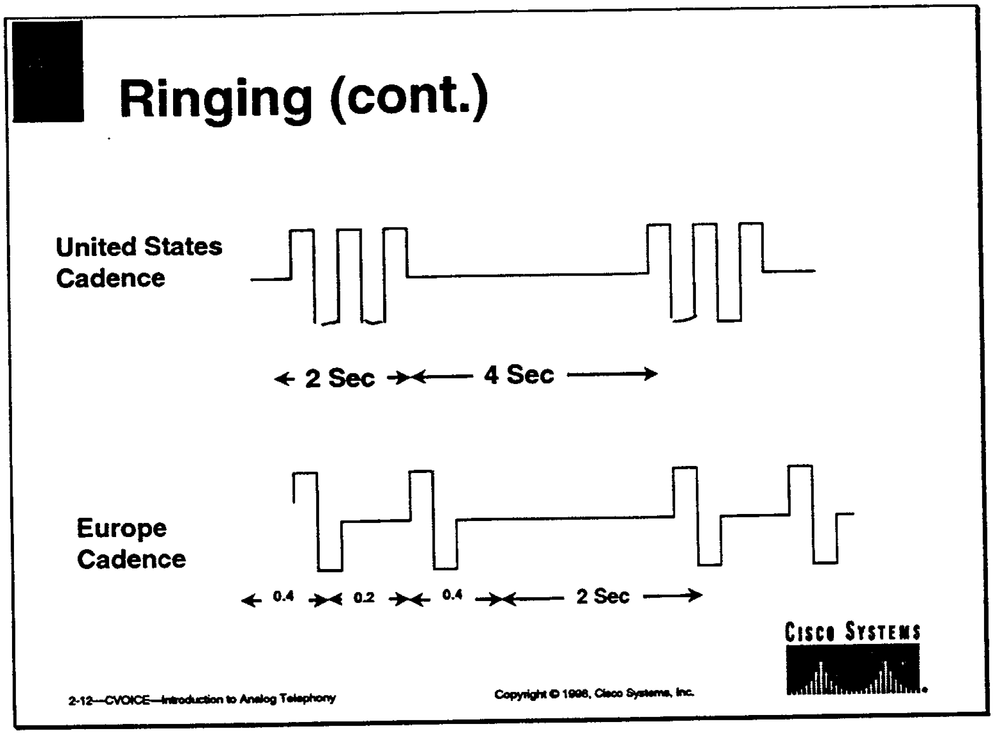
When you wish to place a call you first lift the handset from the telephone cradle. Lifting the handset off the cradle closes the switch hook and current is thereby allowed to flow freely through the phone. When the handset is removed from the cradle, the circuit is said to have gone off-hook and the switch hook is toggled to a closed state, causing circuit current to flow through the electrical loop. The current notifies the phone company that you are requesting to place a telephone call. When a call is initiated, the telephone network senses the off-hook connection by the flow of current and provides a signal known as dial tone to indicate it is ready.



Ringing

When someone else is calling you, the telephone sends voltage to the ringer to notify you of an inbound call. The voltage causes an armature within the phone to pivot, which in turn drives a hammer to provide the ringing signal to your telephone.

The telephone company also sends a ring-back tone to the caller alerting the caller that it is sending ringing voltage to the recipient's telephone. Although ring-back tone is not the same as ringing voltage, it sounds similar.

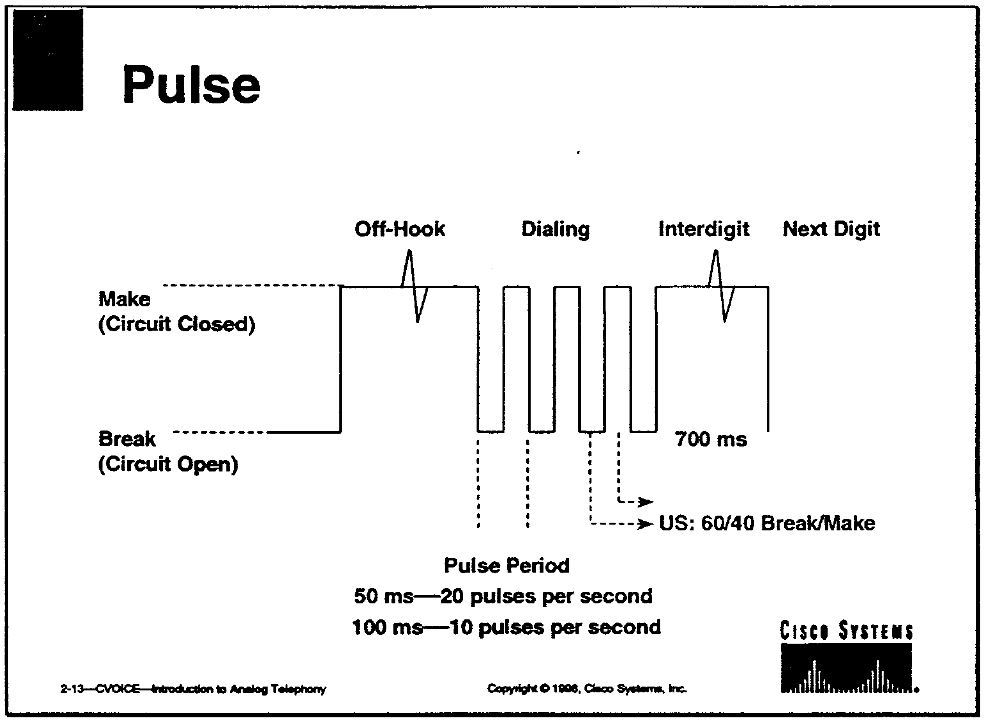


The ringing supervision tones in the United States, for example, the ring cadence, is one to two seconds of tone followed by three to four seconds of silence. Europe uses a double ring followed by two seconds of silence.

Address Signaling

Your telephone can be either a rotary dial telephone or a push-button (touch tone) telephone. When you place a call, each type of telephone uses different types of address signaling to notify the telephone company where you are calling. There are two types of address signaling:

- Pulse
- Dual Tone Multifrequency (DTMF)



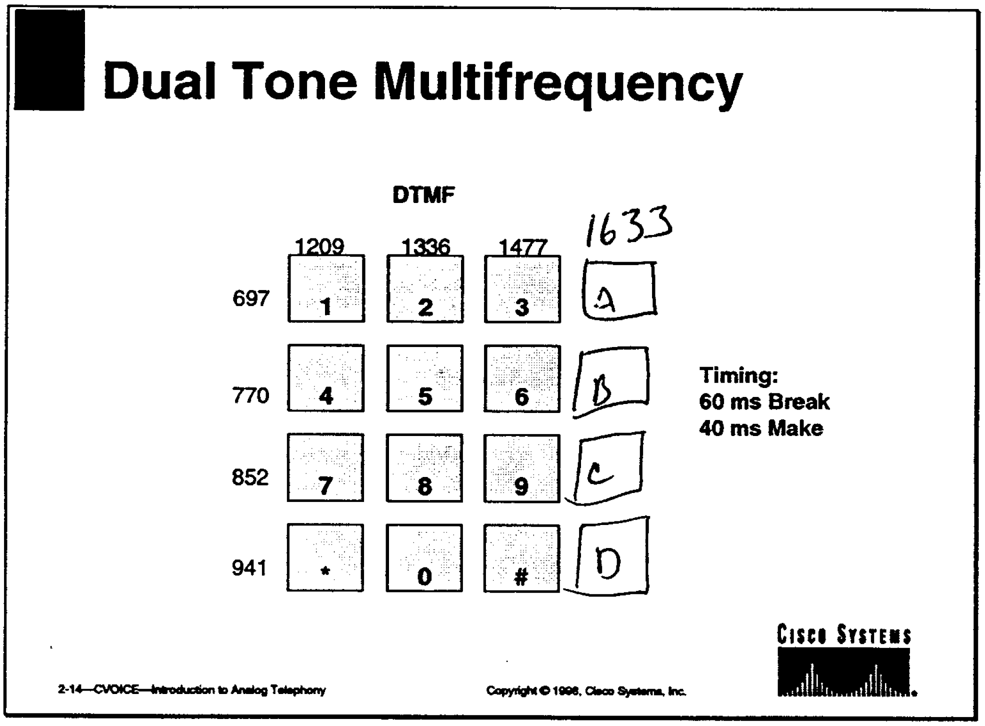
Pulse

Although somewhat outdated, rotary dial telephones phones are still in use and are easily recognized by their big numeric dial-wheel, which is spun to send digits when placing a call. When you dial digits to place a call, the digits must be produced at a specific rate and be within a certain level of tolerance. Each pulse consists of a "make" and a "break." The break segment is the time that the circuit is open. The make is the period during which the circuit is closed. The cycle should correspond to the following ratio: 60 percent break, 40 percent make.

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

A summary of the dial pulse signaling process is presented as follows:

1. When you call 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 represents the dialed digit.



Dual Tone Multifrequency

If you have a touch pad or push-button phone, you will push buttons on a keypad when placing a call. Each button on the keypad is associated with a set of high and low frequencies. When you look at the keypad, each row of keys is identified by a low frequency tone and each column is associated with a high frequency. The combination of both tones notifies the telephone company of the number you are calling, hence the term DTMF.

The graphic illustrates the combination of tones you can generate for each button on the keypad.

Note In the United States, the tone generators should be within a tolerance of 1.5 percent. Any tones exceeding a tolerance of 3.5 percent are ignored by the telephone company equipment.

Informational Signaling—Call Progress Indicators (United States)

Tone	Frequency (Hz)	On Time (sec)	Off Time (sec)
Dial	350 + 440	Continuous	
Busy	480 + 620	0.5	0.5
Ring-back, line	440 + 480	2	4
Ring-back, 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, Freq. Mod 1 Hz	
Confirmation tone			

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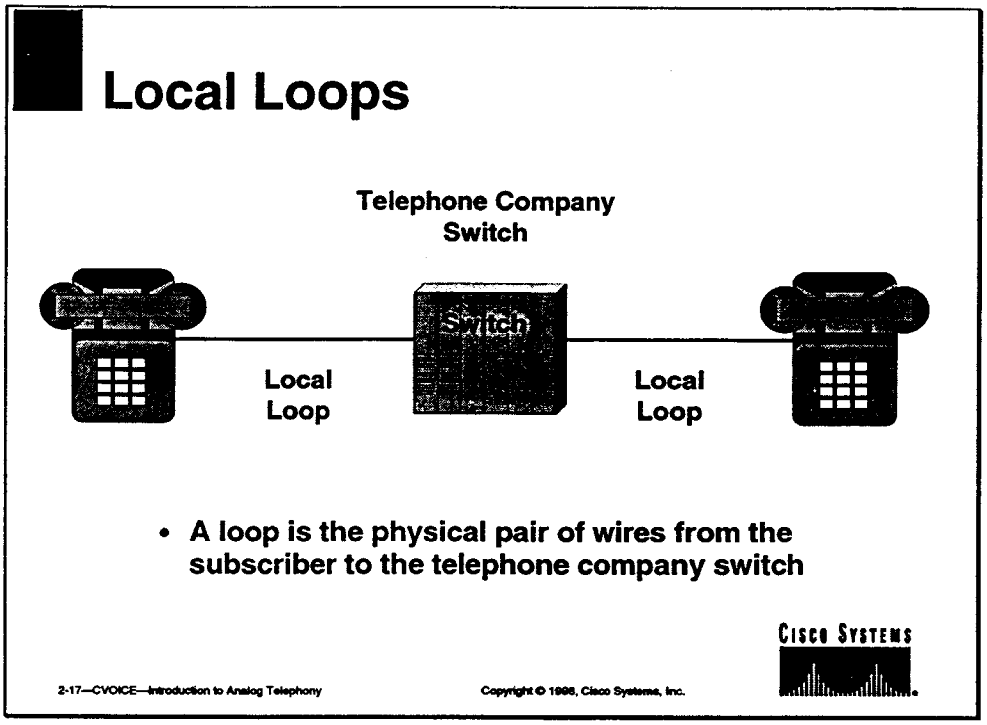
Informational Signaling—Call Progress Indicators

Call Progress Indicators are used to notify you of your call's status. Each combination of tones notifies you of different events in the call's progress. The graphic indicates the tones used to notify you of your call's progress. The tones tell you the following:

- Dial—A dial tone is given when the telephone company is ready to receive digits from the user's telephone.
- Busy—A busy signal is given when a call could not be completed because the phone at the remote end was busy or already in use.
- Ring-back (normal or PBX)—Ring-back is the sound you hear when you are calling someone else. It is a confirmation that the telephone company is attempting to complete your call.
- Congestion—A fast busy signal is given when there is congestion in the long distance telephone network that prevents your telephone call from being processed.
- Reorder—A reorder or fast busy signal is given when all the local telephone circuits are busy, which prevents your telephone call from being processed.
- Receiver off-hook—When you leave your receive off-hook for an extended period without placing a call, the telephone company will notify you to place it back on-hook.
- No such number—When you place a call to a nonexistent number, the telephone company will notify you with a no such number signal.
- Confirmation tone—Some telephones provide a confirmation tone. It lets you know that it is working on completing the call. Users usually like the tone better than "dead air" while the call is connecting.

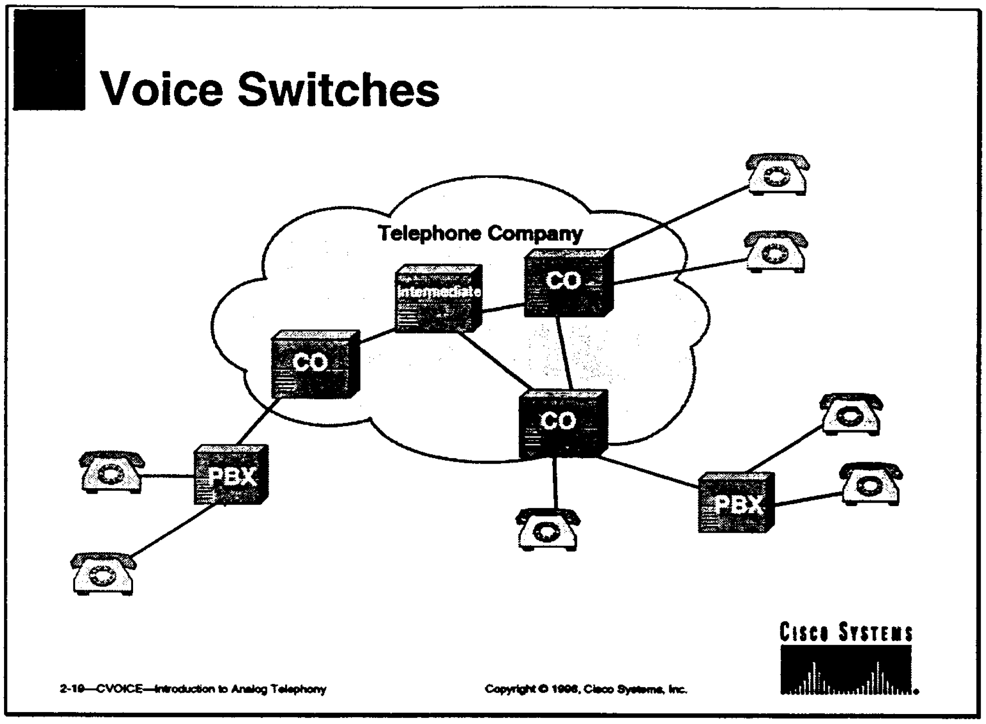


Local Loops

Your telephone is connected to the telephone company by a telephone line or loop. The loop is the physical pair of wires that connects your telephone to the telephone company. You may have heard the term "local loop." Local indicates the connection between your house and the telephone company.

The loop contains an electrical communication path of two wires, one for transmitting and one for receiving voice signals. The two-wire circuit is referred to at the tip and ring. The ring is tied to the negative side of the battery at the telephone company and the tip is tied to the ground.

When you take your telephone off-hook, current flows down the wires and your telephone company provides service to you.



Voice Switches

A voice switch is a mechanical or electrical device that will direct your voice call to the proper destination. A voice switch can be at your site or at the telephone company. If you are placing calls from home, you are most likely relying on the telephone company's switch to complete your call. This switch is called the local central office (CO) switch. Likewise, most businesses rely on PBXs to do some of their own switching.

Without a switch in your telephone network, you would need a telephone line to each destination you wish to call. Switches make it possible to construct a dedicated line for the extent of your telephone conversation and break it down again when the conversation ends.

Note A switched telephone network is analogous to switched virtual circuits. The connection only exists for the duration of the call and is then broken down.

Various vendors manufacture telephone company switches. Typical manufacturers are Lucent Technologies and Nortel in North America; Ericsson, Siemens, and Alcatel in Europe; and NEC and Fujitsu in Japan.

Note Some countries refer to the telephone company's CO as the "public exchange."

The switch selectively establishes and releases connections between transmission facilities to provide dedicated paths for the exchange of messages between two users. Paths are established before the information exchanges begin, and, until the users terminate the sessions, these paths are maintained for the switch's exclusive use.

This section will discuss the following types of switches:

- Central office
- Intermediate switch
- PBX

Central Office Switch

- **All private phones are directly connected to a CO switch**
- **Two-wire connection to individual analog phones**
- **Calls take one of the following paths**
  - **Switched within the CO**
  - **Sent to another CO**
  - **Sent to an intermediate switch (scalability)**

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Central Office Switch

When you place or receive a call the line is directly connected to the telephone company's CO switch, which will route the call to the proper destination. Private phones are directly connected to the CO.

When you place a telephone call to your CO, it will forward the call to one of the following:

- Another CO switch
- Another end user's telephone if its connected to the same CO
- Intermediate switch

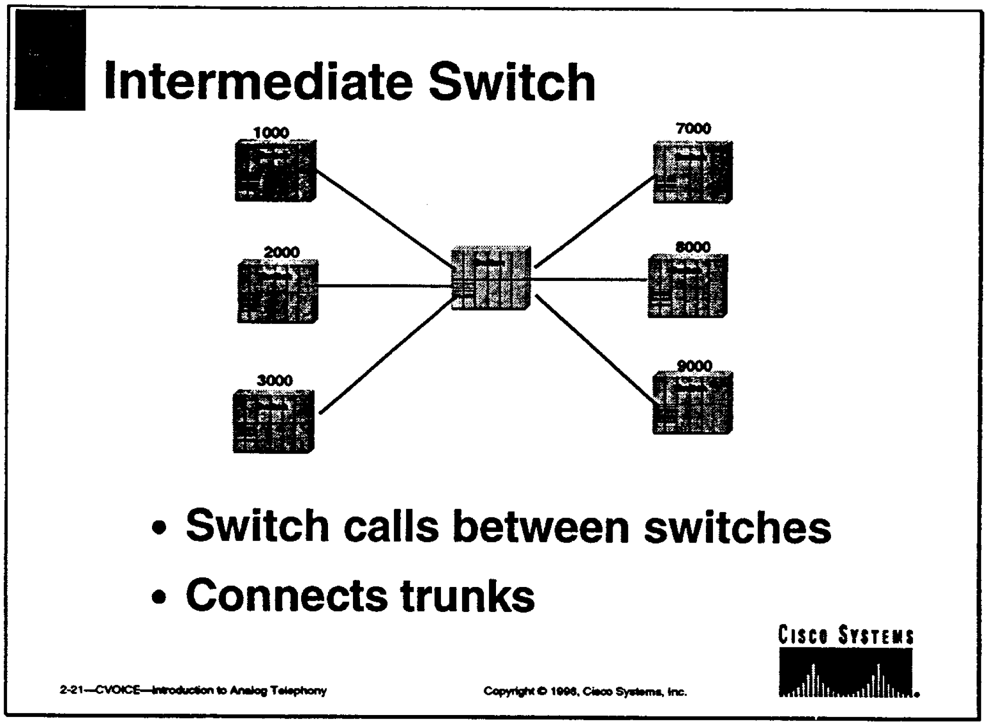
The CO provides the following components to make your telephone work:

- Battery—This is the source of power to both the circuit and the phone and is also used to determine the status of the circuit. When you lift the handset to let current flow, your telephone company is providing the source that powers the circuit and your telephone. Because your telephone company is powering your telephone from the CO, electrical power outages should not affect your basic telephone.

Note Some telephones on the market offer additional features that may require an additional power source that you supply. Cordless telephones are one such example. You may not be able to use a cordless telephone in the event of a power outage.

- Current detector—The current detector monitors the status of the circuit by detecting whether the circuit is open or closed. When the handset rests in the cradle, the circuit is on-hook and there is no current flow. When the handset is raised from the cradle, the circuit is off-hook and current flows in the circuit.

- Dial tone generator—Once the digit register is ready, the dial tone generator generates a dial tone to acknowledge the request for service.
- Dial register—When the PBX detects current flow on the interface, the dial register receives the dialed digit.
- Ring generator—Once the PBX detects a call for a specific subscriber, the ring generator alerts the called party by sending a ring signal to the subscriber.



Intermediate Switch

An intermediate switch is a go-between switch that forwards a call to other switches in the network. It connects trunks.

Private Branch Exchange

- **Used in private sector**
- **Remote branch subsystem**
- **Switches (eXchanges) calls locally**
  - **Keeps "local" calls off public network**
  - **Fewer trunks to CO**
  - **Private line between PBXs**

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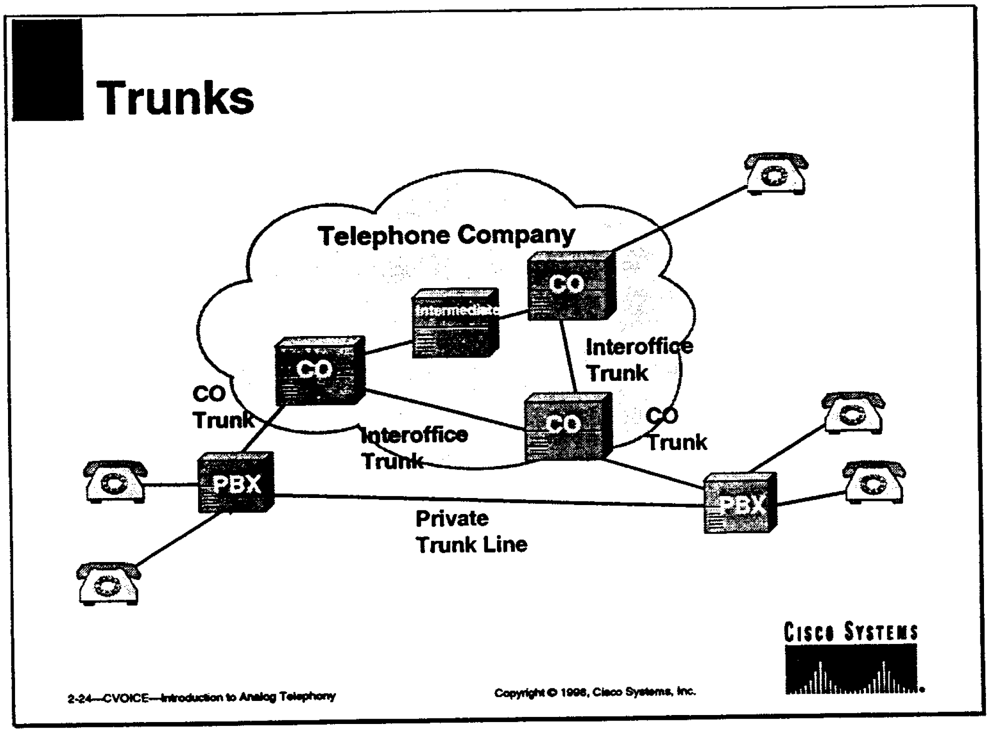
Private Branch Exchange

When you are at your office, for example, there may be many telephones, one for each of your fellow coworkers. It is inefficient to run a line for each telephone to the CO. Likewise, you and your coworkers are generally not on the phone at the same time. So, your employer may place a PBX on site to connect the company telephones. Your employer will also connect to the telephone network with fewer lines than telephones in the office because you and your peers will almost never all be on the phone at the same time. When you need an outside line, the PBX will switch your call to a line to the telephone company.

Note A PBX is often called a Private Automatic Branch Exchange (PABX) in Europe.

PBXs are easily identified by the following characteristics:

- Location on the customer site
- Purchase and maintenance is the PBX owner
- Separate battery backup to system generally required
- Connection medium to other customer switches and to the outside world
- Voice and data switching capabilities enabled through digital technology
- Interface options with other equipment (i.e., voice mail)



Trunks

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

Note Switches provide logic and trunks provide the path between switches.

The trunk's primary function is to provide the path between switches. It is the switch that must route the call to the correct trunk or telephone line. A trunk is shared by many different subscribers, although only one uses it at any given time. As telephone calls complete, trunks are released and made available to the switch for subsequent calls. Between two switches, there may be many trunks.

A few of the more common trunk types are described in this section. They are:

- Private trunk lines
- Central office trunks
- Foreign exchange (FX) trunks
- Direct inward/direct outward dialing (DID/DOD)

Private Trunk Lines

Your employer or others who have multiple PBXs can connect them with private trunk lines. Generally, private trunk lines serve as dedicated circuits that connect PBXs to each other. Subscribers who have PBXs that they want linked will lease trunks from the telephone companies on a monthly basis and reduce their cost by avoiding paying for the use of telephone lines, the other option, on a per-call basis.

CO Trunks

A CO trunk is a direct connection between a local CO and the PBX that enables calls to be routed from the private network at your work site, for example, to the public telephone network. CO trunks also connect COs.

Interoffice Trunks

Interoffice trunks connect COs.

Foreign Exchange Trunks

- **Foreign exchange office**
  - **Plugs directly into the line side of a switch**
  - **Acts like a phone/switch, thinks it is a phone**
- **Foreign exchange station**
  - **Sits at the remote site**
  - **Acts like a switch/phone, thinks it is a switch**

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Foreign Exchange Trunks

FXs are trunk interfaces that allow you to fool a switch into “thinking” that a remote telephone is directly attached to the switch. For example, if you were an employee at a remote office and wanted to connect directly to the PBX, you could use foreign exchange trunks to give the appearance that your telephone is directly connected.

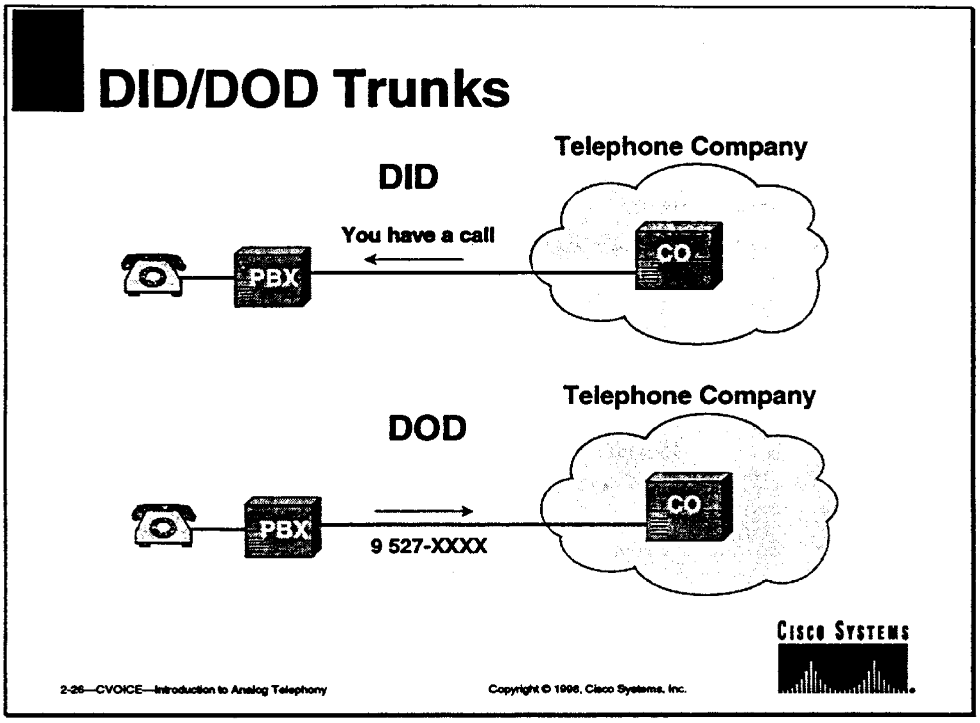
A foreign exchange office (FXO) and foreign exchange station (FXS) are needed to fool the switch into thinking your telephone is directly connected to the switch.

FXO

The FXO sits on the switch end of the connection. It plugs directly into the line side of the switch so the switch thinks the FXO interface is a telephone. The switch notifies the FXO of an incoming call by sending ringing voltage to the FXO. Likewise, the FXO answers a call by closing the loop to let current flow. Once current is flowing, the FXO interface uses any current technology to transport the signal to the FXS.

FXS

The FXS sits at the remote site and looks to the telephone like a switch. It provides the dial tone and other signaling to the telephone. The telephone thinks it is the switch.



Direct Inward Dial (DID) Trunks

DID trunks are one-way trunks that allow you to dial into a PBX without operator intervention. The outside subscriber will dial the extension digits of the desired destination that the connecting CO passes to the PBX. The CO will know which calls to pass through the DID trunk because it associates a block of numbers with each DID trunk.

Direct Outward Dial (DOD) Trunks

DOD trunks are also one-way trunks that allow you to connect directly to the CO. They are outbound trunks. If you are at the office, for example, and want to place a call outside your company's network, you simple dial an access code like "9" and the PBX will forward your call out to the CO. At that time the CO will provide a second dial tone and use the remaining digits you dialed to forward the call to its destination.

Trunk/Line Seizure Signaling Types

FSX

- CO to phone/FXO/~~EXS~~
  - Loop start
  - Ground start
- PBX to PBX E&M
  - Type I to V
  - Start signaling
    - Wink start
    - Immediate start
    - Delay start

no type 4 with Cisco routers

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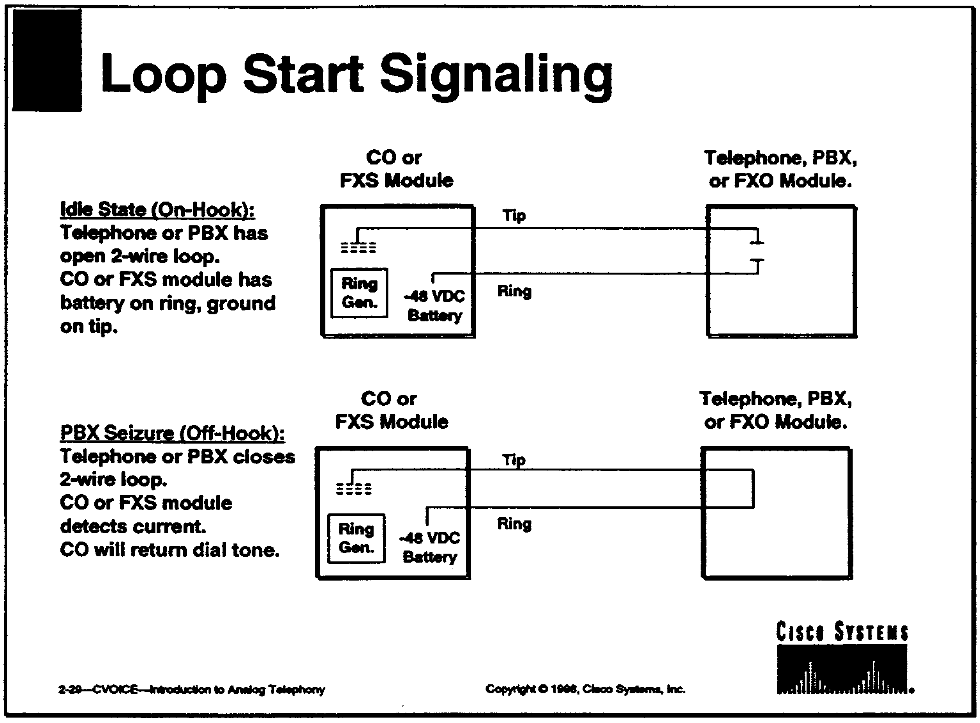
Trunk/Line Seizure Signaling Types

As with the signaling between your telephone and the telephone company, there must be signaling standards between the lines and trunks in the telephone network. This section describes the trunk/line seizure signaling types. The section topics are:

- Loop start signaling
- Ground start signaling
- E&M signaling



Loop Start Signaling

Loop start signaling allows you or the telephone company to seize a line or trunk when a call is being initiated. It is primarily used on local loops rather than on trunks.

Your telephone connection can be in one of the following states:

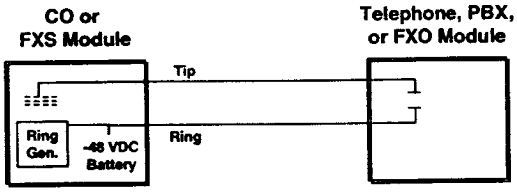
- Idle (on-hook)
- PBX or telephone seizure (off-hook)
- CO seizure (ringing)

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

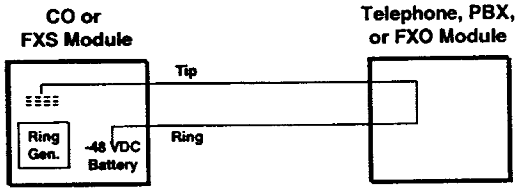
1. When the line is in idle state it is said to be on-hook and the telephone or PBX opens the 2-wire loop. The CO or FXS has battery on ring and ground on tip.
2. If you lift the handset off the cradle to place a call, you cause the switch hook to go off-hook and close the loop. Current can now flow through the telephone circuit. The CO will detect the current and return dial tone.

Loop Start Signaling (cont.)

CO Seizure:
CO applies AC ring voltage, superimposed over the -48 VDC.



When phone goes off-hook, CO removes ring voltage and completes circuit.



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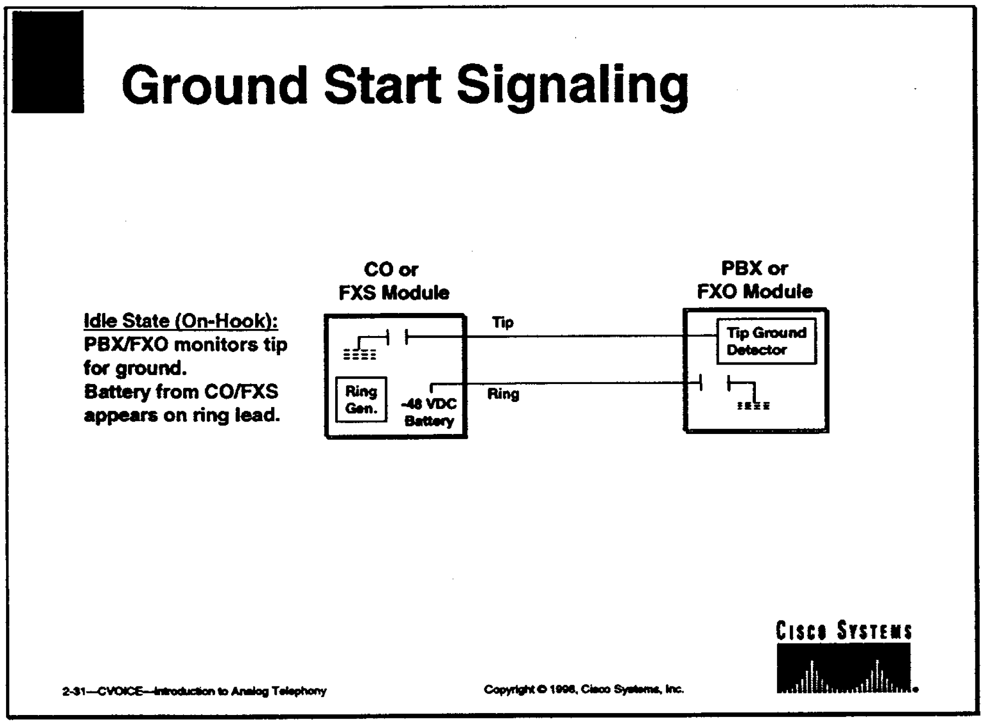
3. If your telephone is ringing to notify you of an incoming call, the CO applies AC ring voltage superimposed over the –48 VDC battery, causing the ring generator to notify you of a telephone call. When the telephone or PBX answers the call, closing the loop, the CO will remove the ring voltage.

Loop start signaling is a poor signaling solution for high volume trunks because it is possible to seize the trunk simultaneously from both ends. This problem is known as glare. Glare periodically occurs on your home telephone. If you are at home, and you pick up the telephone to call out, and the person you were calling is already at the other end of the connection, you just experienced glare. You both seized the loop simultaneously.

Glare is not a significant problem at home, but imagine being at work with ten times the phone usage. Signaling methods that detect loop or trunk seizure at both ends will solve the problem.



Ground Start Signaling

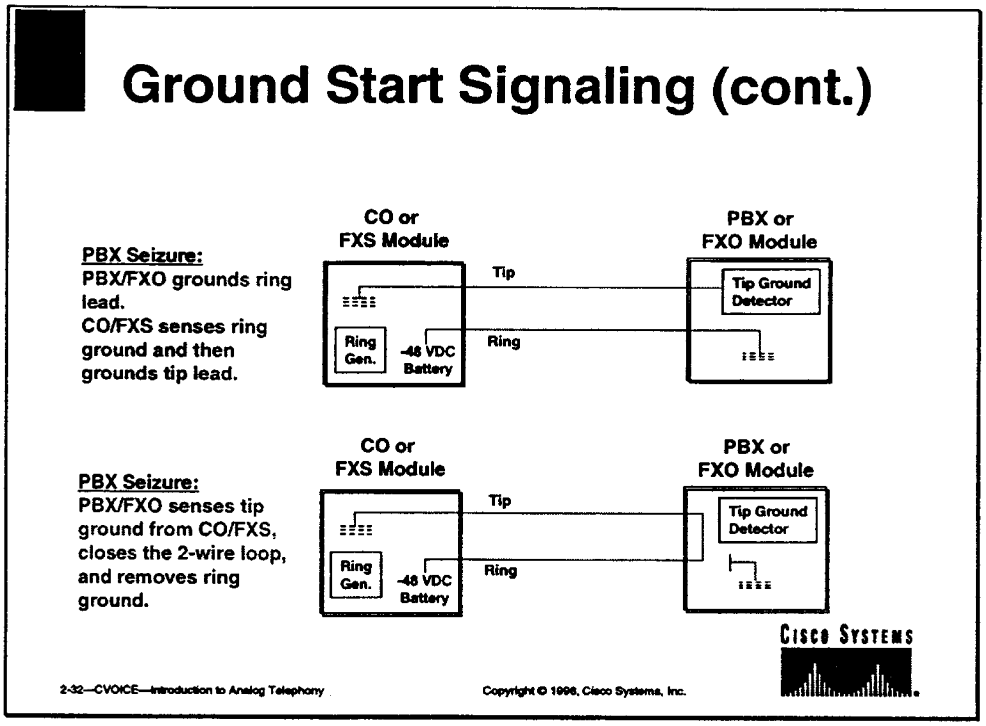
Ground start signaling is a modification of loop start signaling that corrects for the probability of glare. It solves the problem by providing current detection at both ends. Loop start signaling works when you use your telephone at home, but ground start is preferable when high volume trunks are involved.

Your telephone connection can be in one of the following states:

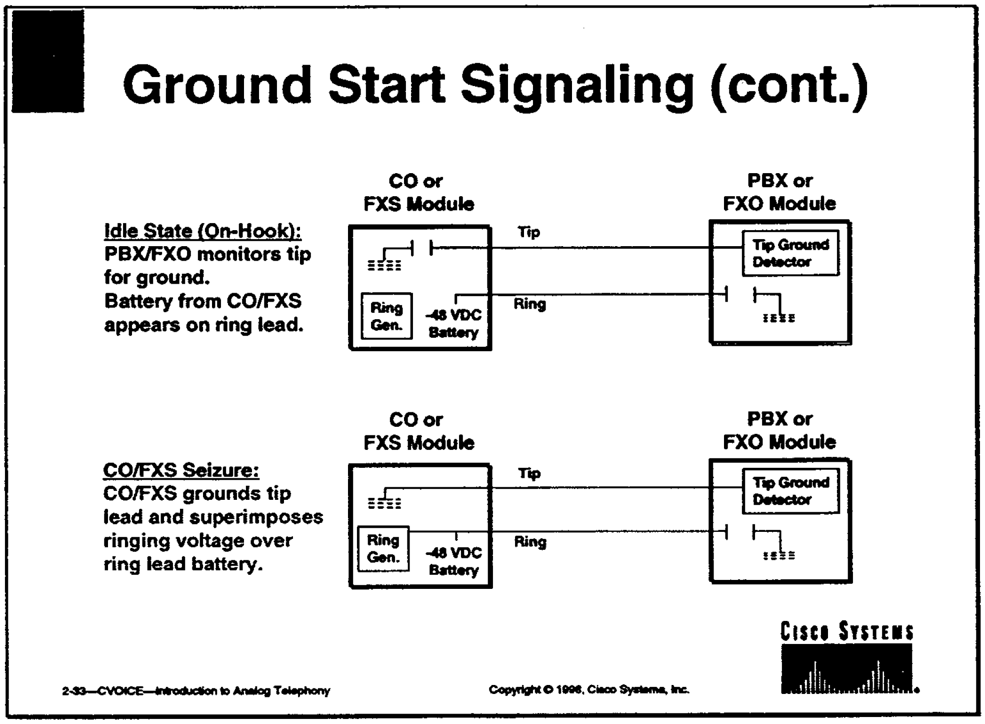
- Idle (on-hook)
- PBX or telephone seizure (off-hook)
- CO seizure (ringing)

A summary of the ground start signaling process is as follows:

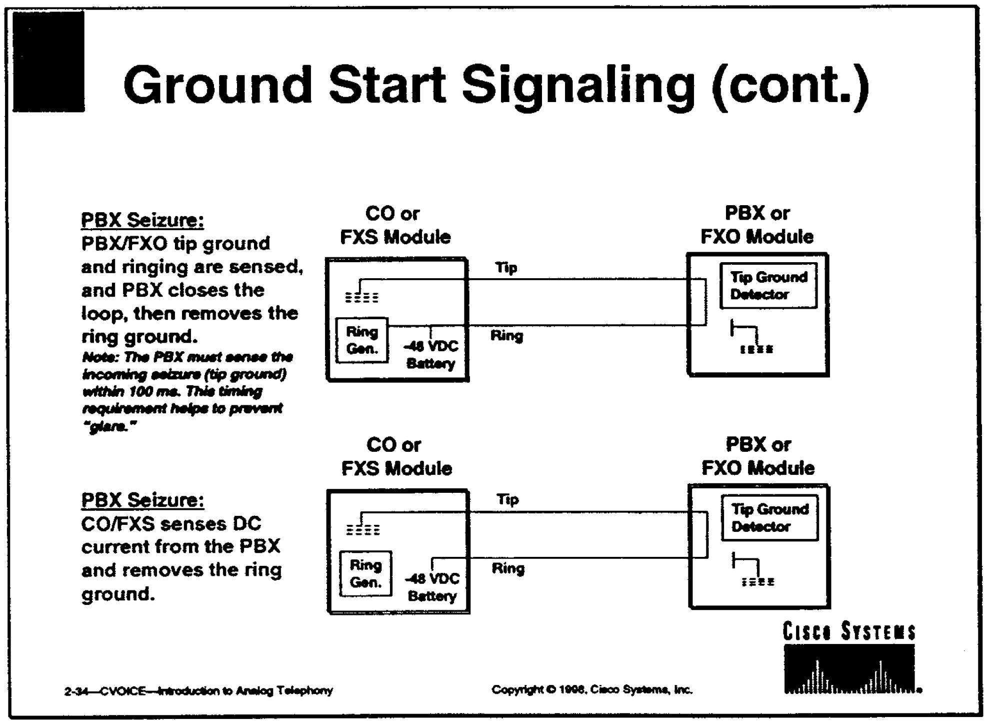
1. When the line is in idle state, the PBX monitors for ground on the tip lead. Battery from the CO appears on the ring lead.



2. If you lift the handset off the cradle to place a call, your PBX grounds ring lead. The CO senses the ring ground and then grounds tip lead. The PBX senses the tip ground from the CO, closes the 2-wire loop, and removes ring ground.



3. If someone is calling you, the CO grounds tip lead and superimposes ringing voltage over ring lead battery. The PBX must recognize the incoming seizure within 100 ms. The tip ground and ringing conditions are sensed and the PBX closes the loop and removes the ring ground.



E&M Signaling

- **Separate signaling leads for each direction**
- **E-lead (inbound direction)**
- **M-lead (outbound direction)**
- **Allows independent signaling**

PBX to Intermediate Device

Type	Lead	On-hook	Off-hook
I	M	Ground	Battery (-48 VDC)
II	M	Open	Battery (-48 VDC)
III	M	Ground	Battery (-48 VDC)
IV	M	Open	Ground
V	M	Open	Ground

Intermediate Device to PBX

Type	Lead	On-hook	Off-hook
I	E	Open	Ground
II	E	Open	Ground
III	E	Open	Ground
IV	E	Open	Ground
V	E	Open	Ground

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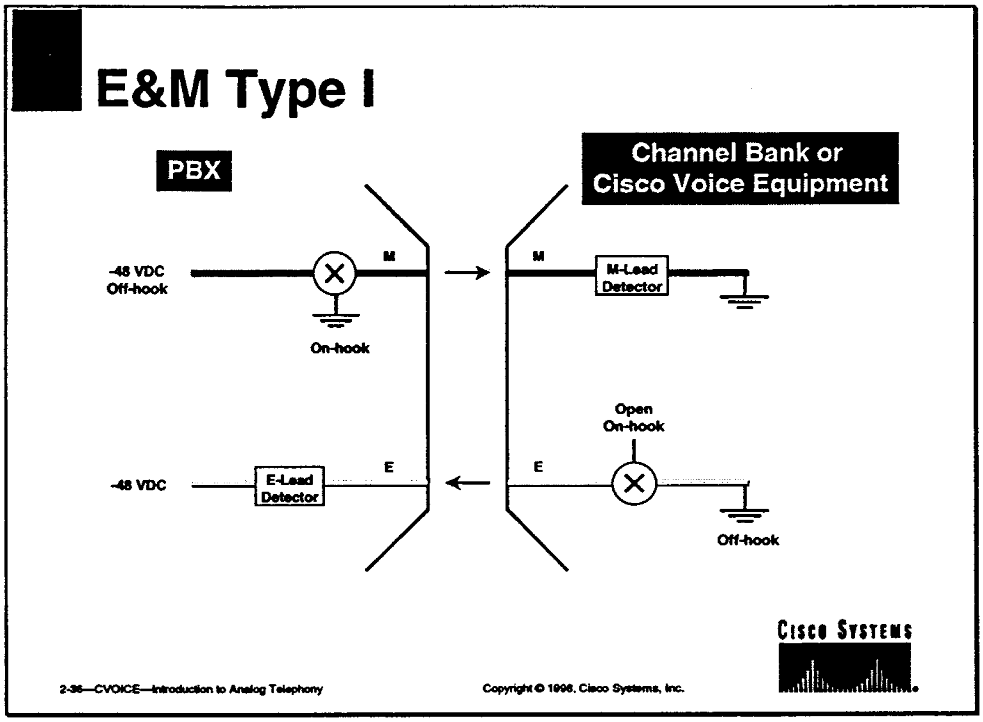
E&M Signaling

The term E&M means Earth and Magneto or Ear and Mouth. E&M signaling is used to support tie-line type facilities or signal between voice switches. Five types of E&M signaling exist. Instead of superimposing both voice and signaling on the same wire, E&M uses separate paths or leads for each. The M (mouth) lead sends the signal and the E (Ear) lead receives the signal.

For example, if you wish to call your friend at a remote office, your PBX must route a request over its signal leads for use of the trunk between the two sites. Your PBX makes the request by raising its M-lead. The other PBX will detect the request when it detects current flowing on its E-lead. It then attaches a dial register to the trunk and your PBX. Your PBX sends the called digits. The remote PBX raises its M-lead to notify you that the call is complete. This is the basic flow of E&M signaling.

There are five different types of wiring schemes:

- Type I
- Type V
- Type II
- Type III
- Type IV



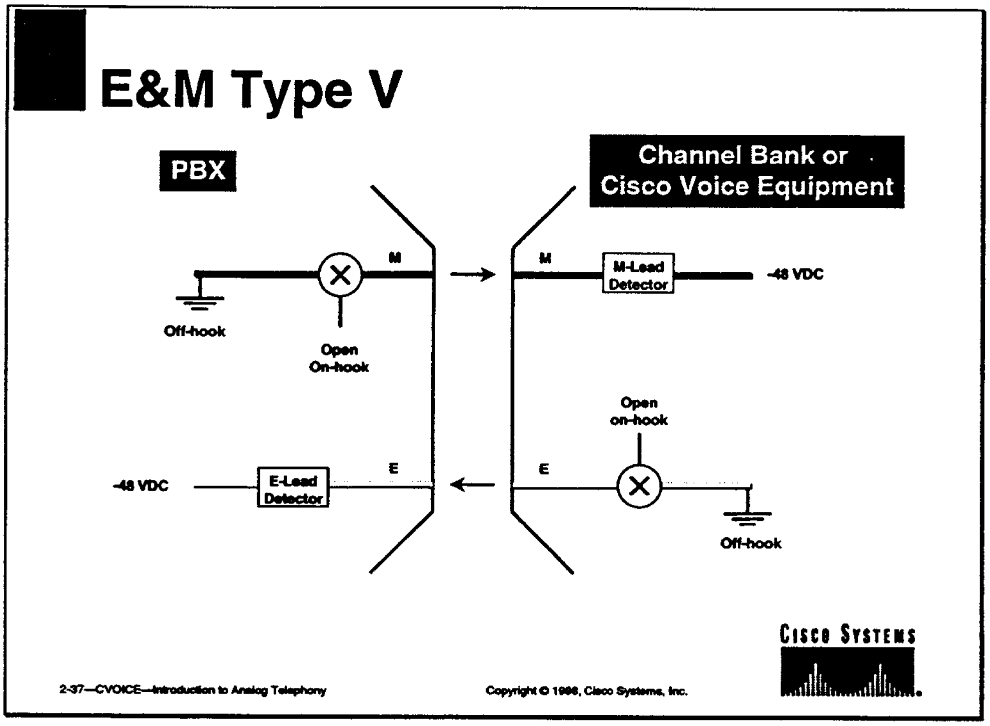
E&M Type I

Type I is a 2-wire E&M signaling type common in North America. One wire is the E-lead and the second wire is the M-lead.

With the Type I interface the 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 tie-line equipment, which detects it via a resistive load.

The Type I interface requires that the PBX and tie-line equipment share a common signaling ground reference.

Note Approximately 75 percent of North American PBXs are Type I.

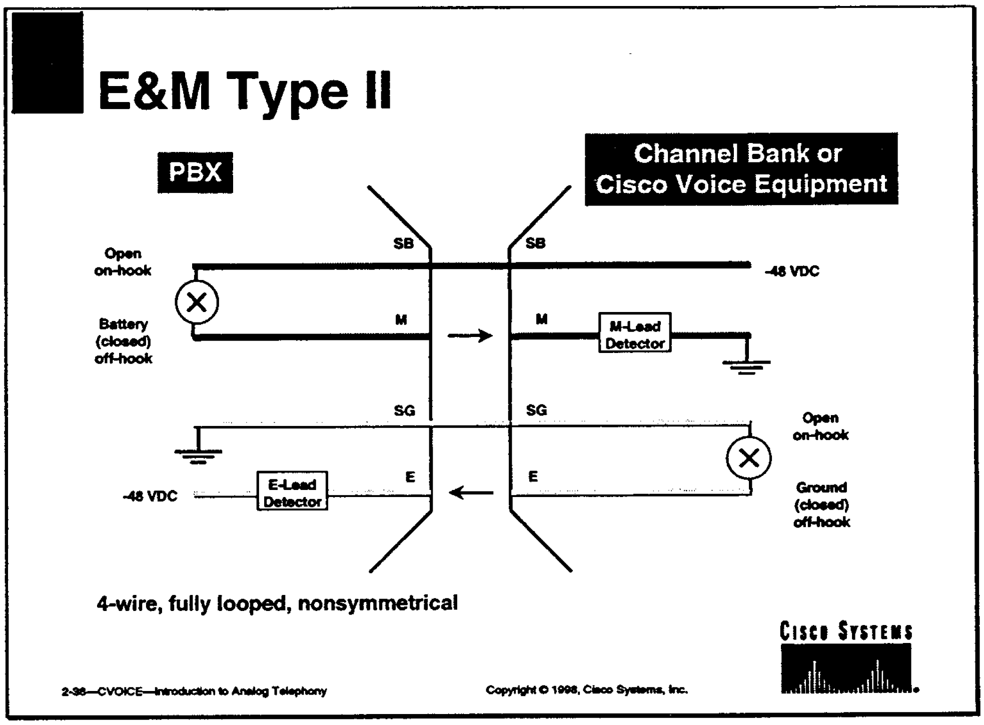


E&M Type V

Type V is also a 2-wire E&M signaling type common outside North America. One wire is the E-lead and the second wire is the M-lead.

Type V is a simplified version of the Type IV interface you will see. This is a symmetric interface, using only two wires. Type V requires a common ground between the PBX and the tie-line equipment; this is provided via the signal ground (SG) leads.

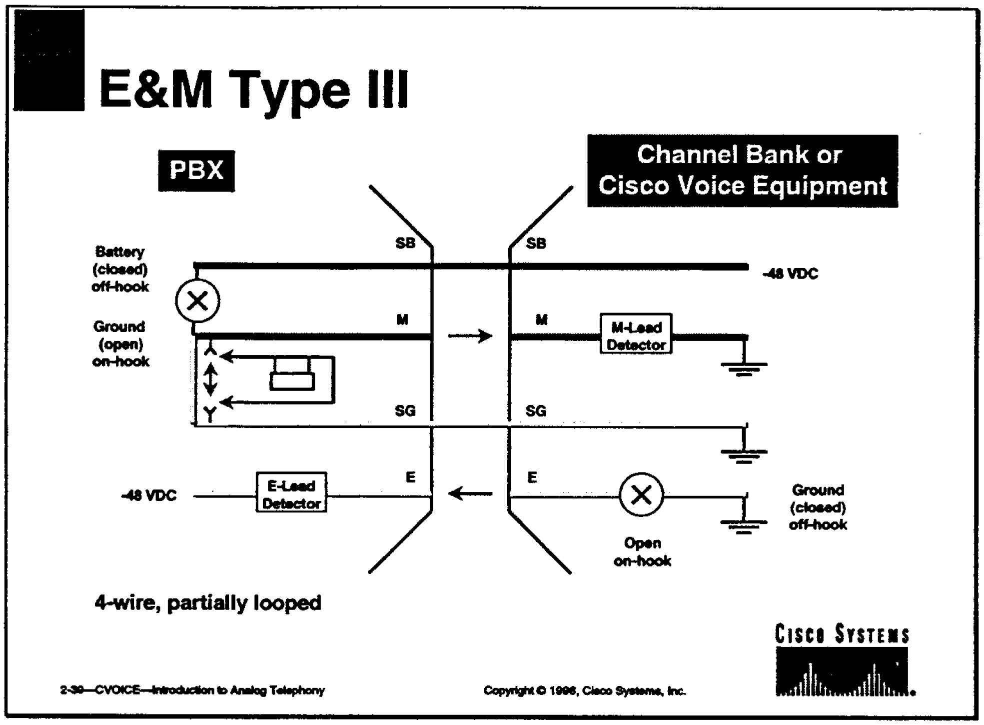
Note This is the most common E&M signaling form outside North America.



E&M Type II

Types II, III, and IV are 4-wire interfaces. One wire is the E-lead and the second wire is the M-lead. The remaining two wires are SG and signal battery (SB). In type II, SB and SG are the return paths for the M-Lead and E-lead, respectively.

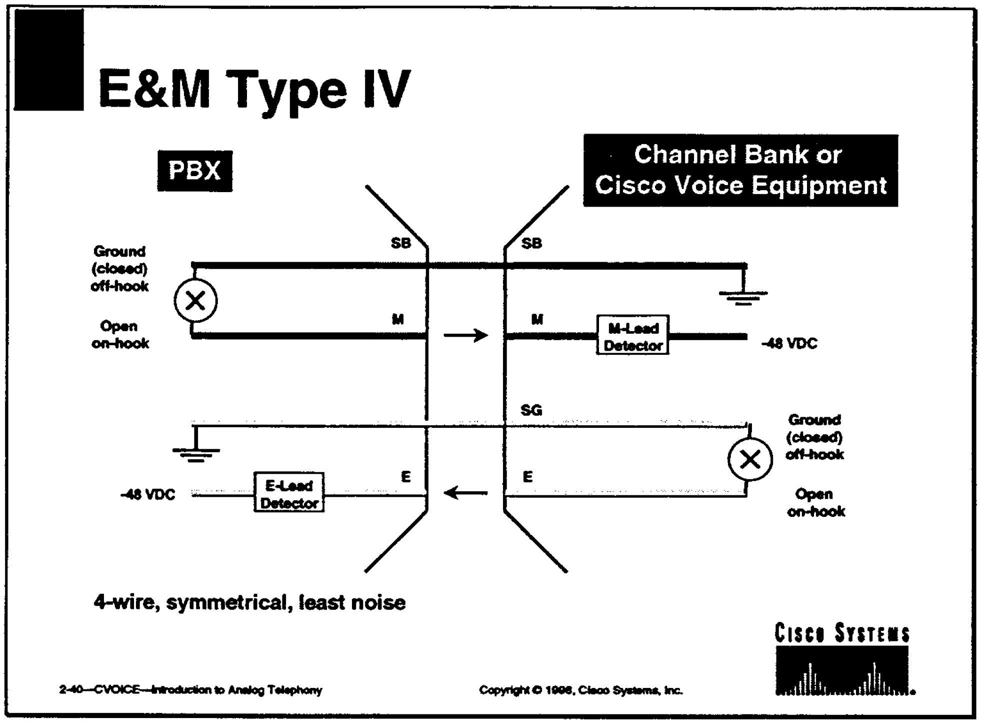
The Type II interface requires no common ground; instead, each of the two signals has its own return. For the E signal, the tie-line equipment permits current to flow from the PBX; the current returns to the PBX's SG lead or reference. Similarly, the PBX closes a path for current to generate the M signal to the tie-line equipment or the signal battery (SB) lead.



E&M Type III

A variation of Type II, Type III uses the SG lead to provide common ground. The E-lead operates similar to Type I. With this configuration, the PBX drops the M signal by grounding it, rather than by opening a current loop.

Note This is not a common signaling type.



E&M Type IV

Type IV is symmetric and requires no common ground. Each side closes a current loop to signal; the flow of current is detected via a resistive load to indicate the presence of the signal.

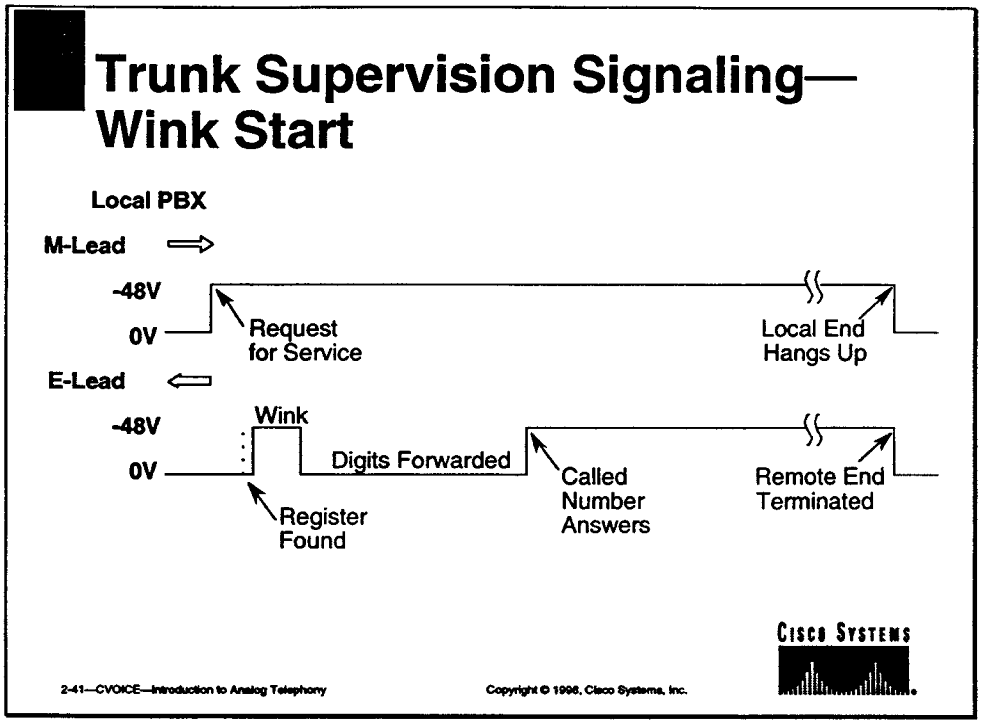
Start Protocols for Trunk Supervision

Tie trunks have bidirectional supervisory signaling, which allows either end to initiate a seizure of the trunk. In this way, one PBX seizes the trunk, after which it expects an acknowledgment reply from the remote end. The local end must differentiate between a return acknowledgment and a remote end request for service.

Like the ground start protocol, E&M start protocols lessen the potential for glare by ensuring that the calling PBX receives a double acknowledgment from the remote PBX before a trunk is seized.

Start protocols are as follows:

- Wink start
- Delay start
- Immediate start



Trunk Supervision Signaling—Wink Start

Wink start signaling is the most common E&M trunk seizure signal type.

The following scenario provides a summary of the wink start protocol event sequence:

1. The calling office seizes the line by going off-hook.
2. The called end does not immediately return an off-hook acknowledgment once detecting the seizure of the line by the calling office.
3. Instead of returning an off-hook acknowledgment, the on-hook state is maintained until the receive digit register is attached.
4. The called office toggles the off-hook lead for a specific time.

Note This on-hook/off-hook/on-hook sequence constitutes the "wink."

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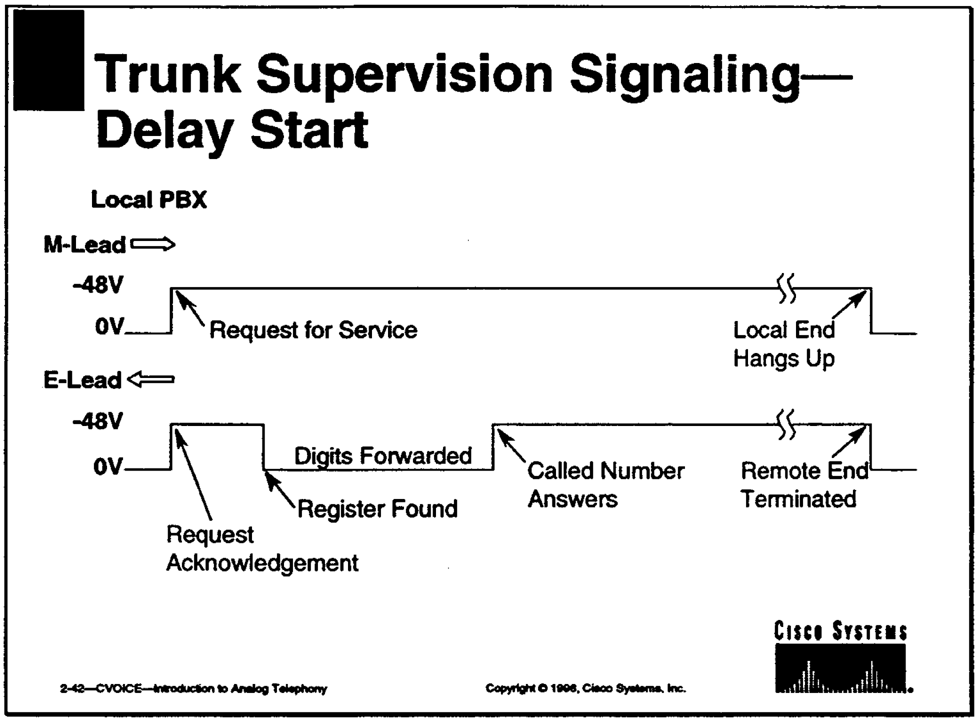
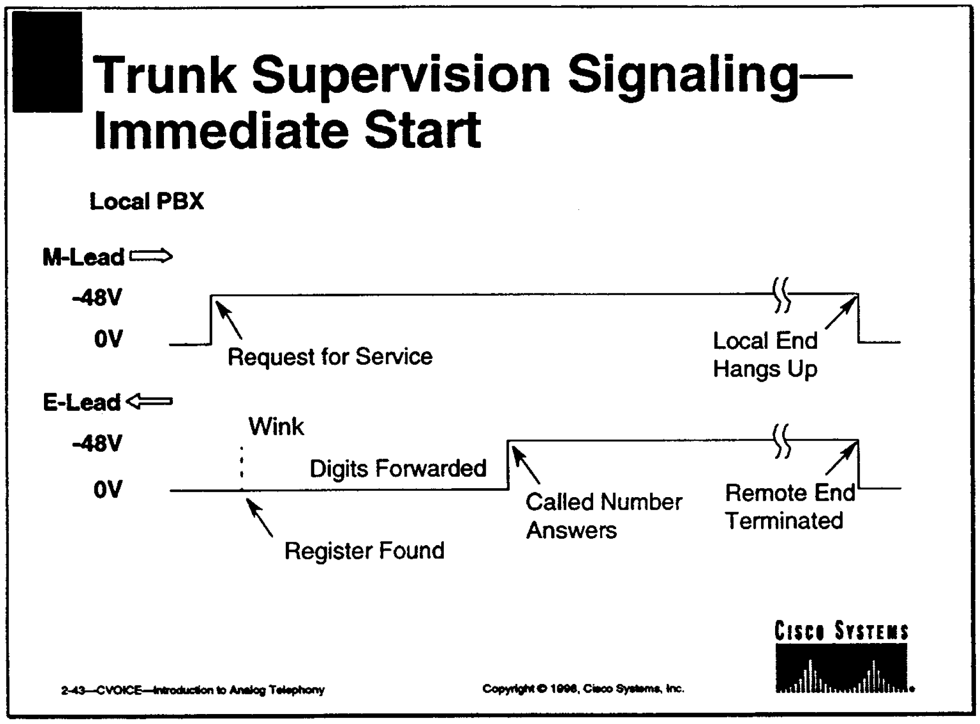
5. The calling office receives the wink and forwards the digits to the remote end.
6. The called party answers the phone.
7. The remote PBX raises the M-lead during the call.



Trunk Supervision Signaling—Delay Start

The following scenario provides a summary of delay start signaling:

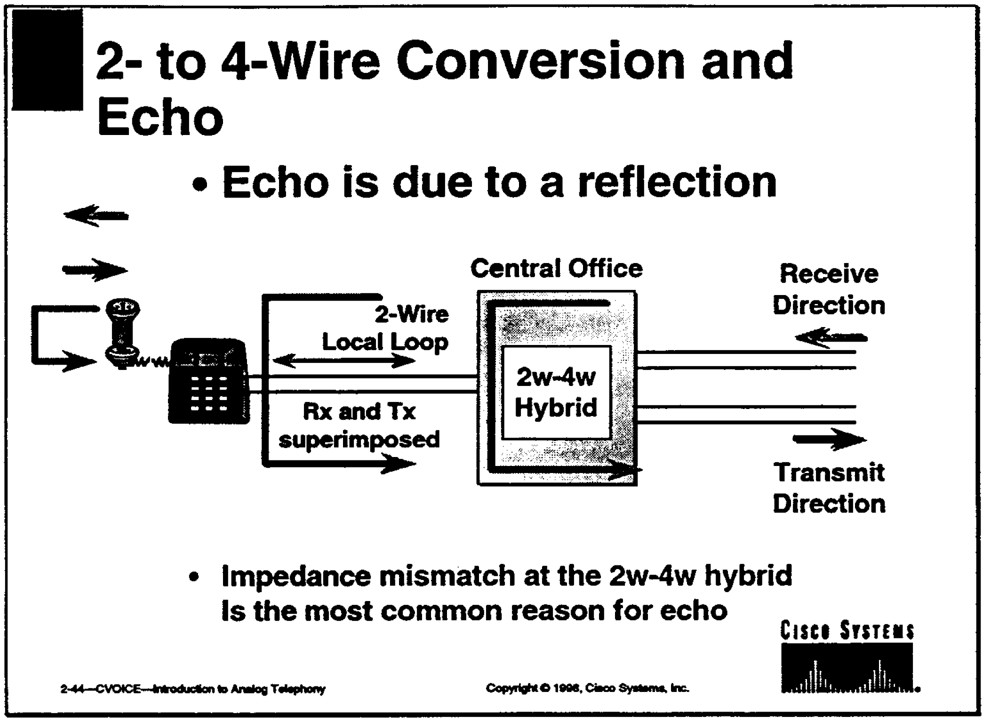
1. When you place a call, your originating switch will go off-hook.
2. The originating switch will look at the status of the remote switch's signal.
3. The originating switch will wait until the remote switch supervision is on-hook.
4. Once the remote is on-hook, the originating switch will output digits.



Trunk Supervision Signaling—Immediate Start

The following scenario provides a summary of the immediate start protocol event sequence:

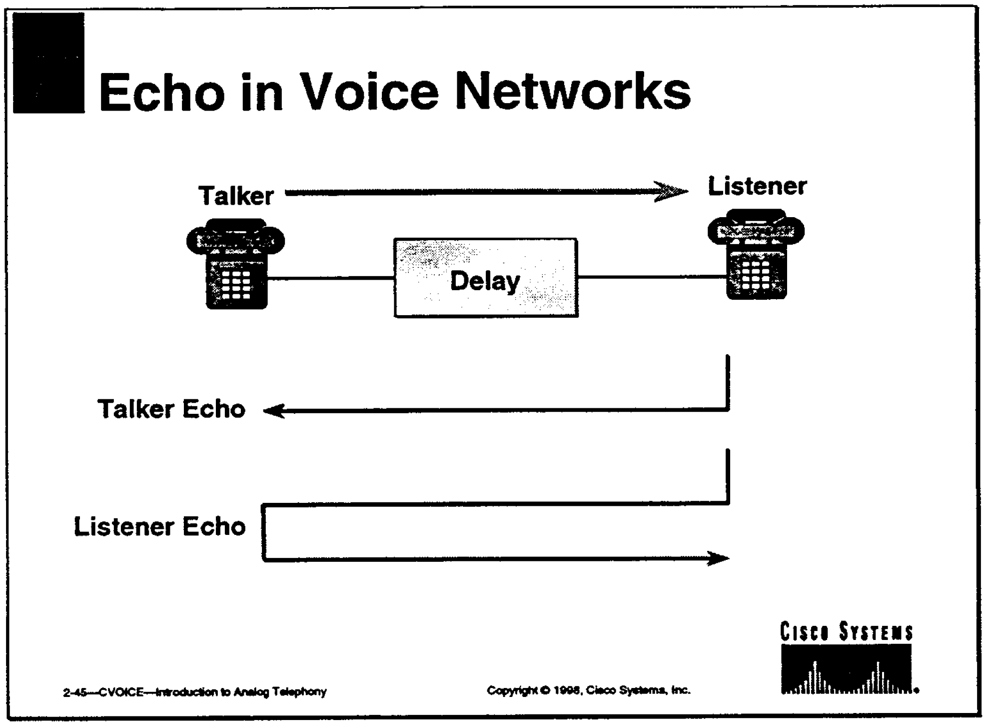
1. Your company's PBX seizes the line by going off-hook.
2. Instead of receiving a double acknowledgment, your local PBX waits a predetermined time and forwards the digits "blindly." The originating switch goes off-hook and maintains the condition for at least 150 ms before outputting digits on the audio path.
3. The remote PBX only acknowledges your PBX after the called party answers the call.



2- to 4-Wire Conversion and Echo

Your local loop is made up of 2 wires. Once it reaches the switch the connection is changed to 4 wires with a 2- to 4-wire hybrid converter so your signal can be transported across the trunks in the network.

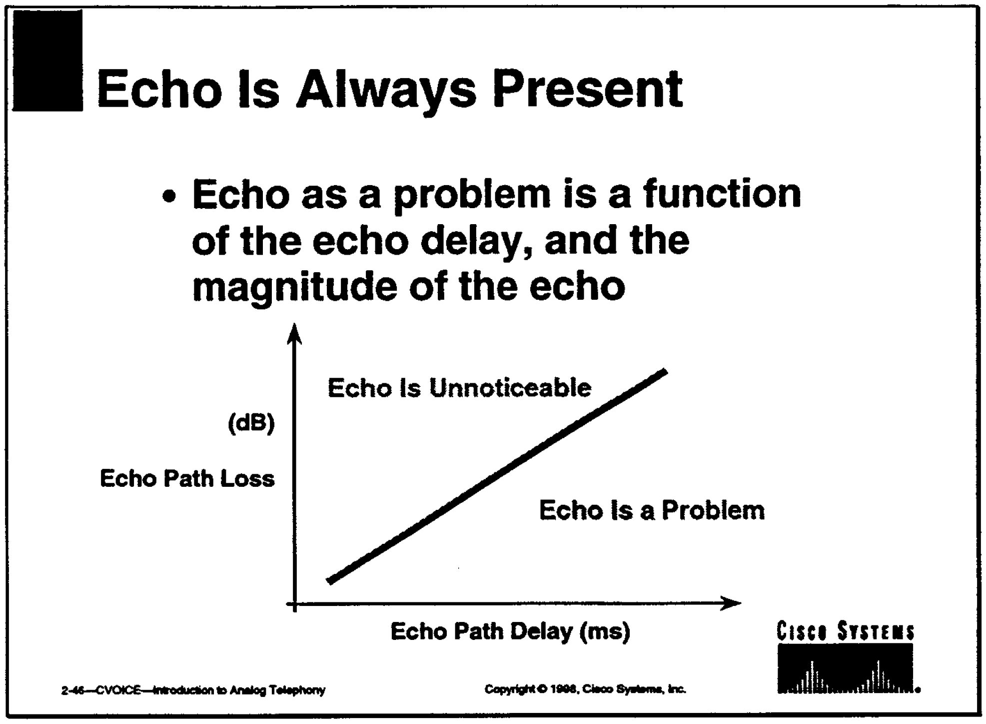
If there is a good impedance match between the lines, the hybrid is said to be balanced with little or no reflected energy. However, if the hybrid is inadequately balanced, and a portion of the transmit voice is reflected back toward the receive side, echo results.



Echo in Voice Networks

Two types of echo exist. They are:

- Talker echo —If you talk and here your voice reflect to you, you are experiencing talker echo. In effect, you hear yourself twice.
- Listener echo —If you are listening to another speak and hear the speaker's voice twice, you are experiencing listener echo.



Echo Is Always Present

Some form of echo is always present. However, echo is a problem if the magnitude or loudness of the echo is high. Echo is also a problem if the delay time between when you speak and when you hear your voice reflected is significant. If you are the listener, echo is a problem if you hear the speaker twice.

Note Everyone's echo tolerance is different. However, echo delay over 50 ms is generally problematic for most people.

If you sense a problem with echo in your telephone network, there are two ways to solve the problem. They are:

- Echo suppression
- Echo cancellation

Echo Suppression

- Suppresses your voice on the return path
- Acts as a noise gate, effectively making communications half-duplex

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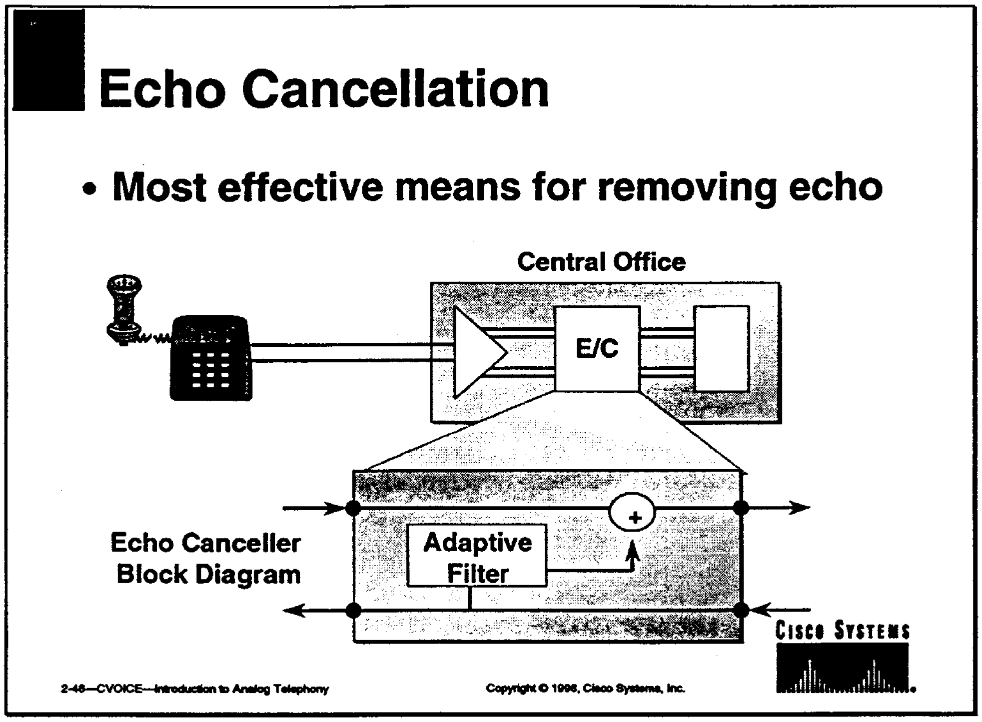
Echo Suppression

Voice is often strong and echo is attenuated. The echo suppressor determines which signals match to you and which signals match to the person you are speaking to, or both. If the echo suppressor determines that the echo is on the return path, the echo suppressor either attenuates or breaks the transmission path. If the echo suppressor determines that both speech and echo are present at the same time from a combination of both parties on the phone, the echo cannot be attenuated without affecting the voice level.



Echo Cancellation

Due to echo suppression's shortcomings in addressing certain echo conflict situations such as the one described in the "Echo Suppression" section of this chapter, a more sophisticated method of eliminating echo is echo cancellation.

Rather than break or attenuate the transmit path as is the case in echo suppression, echo cancellation uses an echo canceller to build a mathematical model of the speech pattern and subtracts it from the transmit path.

Note The echo canceller only removes the echo from one end of the circuit. If the echo is an issue at both ends of the circuit, another separate echo canceller would need to be applied at the other end.

Telephone Call Procedure

When you place a call to another party, the following events occur.

1. When the line is in idle state, it is said to be on-hook and the telephone or PBX opens the 2-wire loop.
2. If you lift the handset off the cradle to place a call, you cause the switch hook to go off-hook and close the loop. Current can now flow through the telephone circuit.
3. The switch will detect the current and return dial tone.
4. When you get dial tone you will request a specific connection by dialing a dial or pressing buttons with a rotary phone or DTMF phone, respectively.
5. The switch will then signal the called party by sending ringing voltage to the called party.
6. The switch will also send an audible ring signal back to you to indicate that it is calling the called party.
7. When the called party lifts the handset, the loop on the called end closes and current flows through the loop. There is now an analog connection.

Written Exercise

Answer the following questions.

Questions

Q1) When you speak to your friend on the phone, you hear yourself speaking twice. What is happening? Why is it happening? List and describe two solutions to the problem.

Q2) Your boss receives a phone bill and complains of paying a local line charge for each telephone in the office. What type of device can you suggest be used to streamline the amount of lines connected to the telephone company? Describe the device's characteristics.

Q3) You are in charge of designing your company's telephone network. Your company has two office sites with PBXs at each site. You have been instructed to directly connect the two sites. Describe what you need to connect the sites.

Q4) You work at a remote office but want all your customers to call you through the main office. Your calls must then be trunked to you. Describe the type of trunk you need between the main and remote offices.

Q5) Your boss complains that often she will pick up the phone at the office and someone will already be on the line even though the phone did not ring. Explain to her why and explain solutions.

Q6) Explain the E&M Type I interface wiring and signaling scheme.

Q7) Explain the wink start protocol process.

Answers

Answers to the written exercise follow:

Q1) When you speak to your friend on the phone, you hear yourself speaking twice. What is happening? Why is it happening? List and describe two solutions to the problem.

Your local loop is made up of 2 wires. Once it reaches the switch the connection is changed to 4 wires with a 2- to 4-wire hybrid converter so your signal can be transported across the trunks in the network. If the hybrid is inadequately balanced, and a portion of the transmit voice is reflected toward the receive side, echo results. You can solve your echo problem with echo suppression or echo cancellation. An echo suppressor determines which voice signals are yours. If it determines that the echo is on the return path, it attenuates the signals.

Q2) Your boss receives a phone bill and complains of paying a local line charge for each telephone in the office. What type of device can you suggest be used to streamline the amount of lines connected to the telephone company? Describe the device's characteristics.

Suggest to your boss that he make the company's telephone system more efficient with the purchase of a PBX. He can connect all the company phones to the PBX and have fewer lines going out to the telephone company.

Characteristics of a PBX are:

- Location is on the customer site.
- Purchase and maintenance is by the PBX owner.
- Separate battery backup to system is generally required.
- The PBX is a connection medium to other customer switches and to the outside world.
- Voice and data switching capabilities are often enabled through digital technology.
- PBXs provide interface options with other equipment (voice mail, etc.)

Q3) You are in charge of designing your company's telephone network. Your company has two office sites with PBXs at each site. You have been instructed to directly connect the two sites. Describe what you need to connect the sites.

You should connect the sites with tie trunks. Generally, tie trunks serve as dedicated circuits that connect PBXs to each other. Subscribers who have PBXs that they want linked will lease tie trunks from the telephone companies on a monthly basis and reduce their cost by avoiding paying for the use of telephone lines, the other option, on a per-call basis.

Q4) You work at a remote office but want all your customers to call you through the main office. Your calls must then be trunked to you. Describe the type of trunk you need between the main and remote offices.

You need FX trunks. An FXO interface will sit at the company's PBX, fooling the PBX into thinking that your telephone is a local phone, with a local number. An FXS interface will be at your site, fooling your telephone into thinking it is connected to a switch. The FXS provides your phone with dial tone.

Q5) Your boss complains that often she will pick up the phone at the office and someone will already be on the line even though the phone did not ring. Explain to her why and explain solutions.

Her PBX uses loop start signaling and both ends are trying to seize the trunk at the same time, causing glare. Although loop start is common, especially on home telephone lines, it can be a problem on high volume trunks. Purchasing a PBX that has signaling methods designed to solve for glare is a solution. Ground start signaling is one such example.

Q6) Explain the E&M Type I interface wiring and signaling scheme.

Type I is a 2-wire E&M signaling type, one wire is the E-lead and the other is the M-lead. It is common in North America. With the Type I interface the 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 tie line equipment, which detects it via a resistive load.

Q7) Explain the wink start protocol process.

The calling office seizes the line by going off-hook. The called end does not immediately return an off-hook acknowledgment once detecting the seizure of the line by the calling office. Instead of returning an off-hook acknowledgment, the on-hook state is maintained until the receive digit register is attached. The called office toggles (winks) the off-hook lead for a specific time. The calling office receives the wink and forwards the digits to the remote end. The called party answers the phone. The remote PBX raises the M-lead during the call.

Lab Exercise: Verifying Analog Voice Ports

On Cisco equipment, it is on the voice ports that you configure the elements you just learned. You can use various **show** commands to verify many of the configurations.

Lab Setup

No lab setup is required.

Exercise: Verify Voice Ports

Use the **show** command you are instructed to use to evaluate the voice ports.

The **show voice port** command displays configuration information about a specific voice port, 1/0/0. From the privileged EXEC mode, use the **show voice port 1/0/0** command to verify the configuration of the voice port.

Correct configuration should look like the following:

```
R1#show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
The Last Interface Down Failure Cause is
Alias is NULL
Noise Regeneration is enabled
Non Linear Processing is enabled
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancel Coverage is set to 16 ms
Connection Mode is normal
Connection Number is
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s

Analog Info Follows:
Region Tone is set for northamerica
Currently processing none
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm

Voice card specific Info Follows:
Signal Type is loopStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
```


Q1) What type of voice port are you using?

Q2) This voice port is on which slot, subunit, and port?

On an FXS voice port, you should have dial tone because FXS supplies the phone with services your telephone company normally would. Pick up the receiver and check for dial tone. Hang up the telephone after confirming that you have dial tone.

Q3) Look at the **show voice port** output again. What is the signal type?

Q4) What is the administrative state of your telephone?

Q5) What is the impedance set at?

Q6) What is the hook status of your telephony device?

Step 3 Now pick up the handset again. This time with the handset off-hook, type **show voice port 1/0/0** again.

Q7) What is the hook status now?

Step 4 Your router has been configured so you can place a call between the two phones on your router. The telephone numbers are 1111 and 2222. Check again for dial tone on each telephone.

Step 5 On your touch tone handset, you can check for DTMF detection. Pick up the telephone with telephone number 1111. Begin to dial the telephone with number 2222. If the dial tone stops, your DTMF address signaling works correctly.

Step 6 Finish dialing the 2222 phone number to complete the call.

Answers

Q1) What type of voice port are you using?

You are using an FXS port.

Q2) This voice port is on which slot, subunit, and port?

The voice port is on slot 1, subslot 0, port 0.

Q3) What is the signal type?

The signal type is loop start.

Q4) What is the administrative state of your telephone?

The administrative state should be UP.

Q5) What is the impedance set at?

The impedance is set at 600r Ohm.

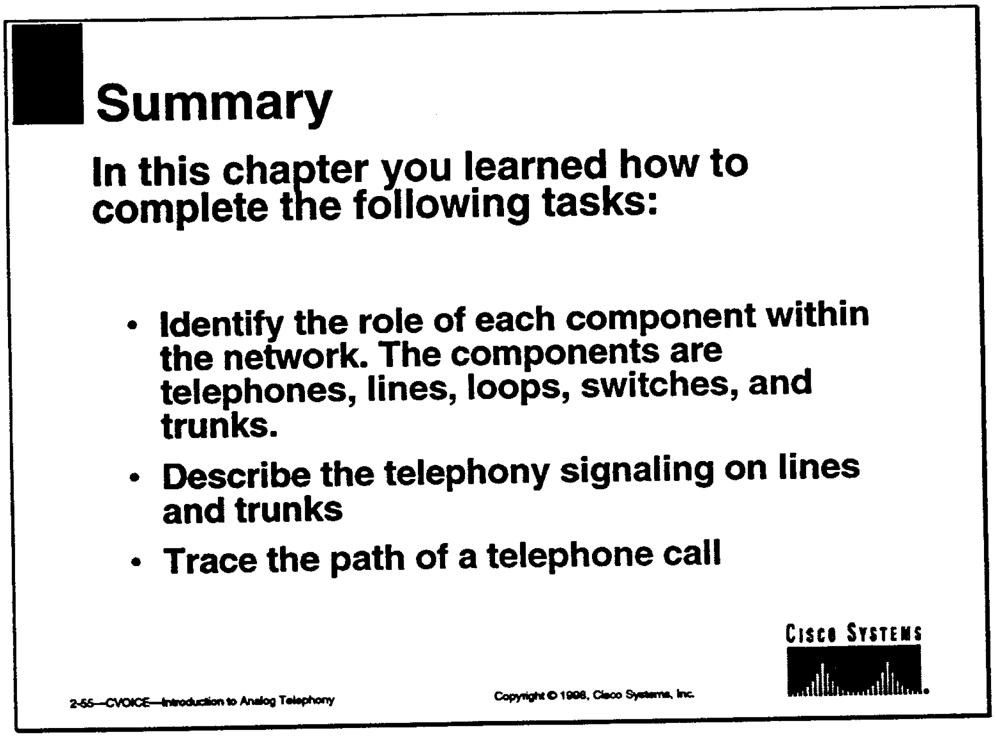
Q6) What is the hook status of your telephony device?

If your handset was on the telephone cradle, the status is on-hook.

Q7) What is the hook status now?

If your handset was off the cradle, the hook status is off-hook.

Summary



Summary
In this chapter you learned how to complete the following tasks:
• Identify the role of each component within the network. The components are telephones, lines, loops, switches, and trunks.
• Describe the telephony signaling on lines and trunks
• Trace the path of a telephone call
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Digital Voice Technology

Objectives

Objectives

- Review, identify, and define digital telephony fundamentals and basics
- Contrast digital and analog signaling
- Identify and contrast the different digital frame formats, signaling formats, and coding

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Objectives (cont.)

- **Compare the various levels of voice quality**
- **Categorize the various types of digital voice compression**
- **Examine ISDN and identify the basic components of this digital architecture**

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3-3—CVOICE—Digital Voice Technology

Chapter Content

The content to support the Chapter 3 learning objectives is primarily a review of digital voice fundamentals and basics. The student should be generally familiar with this material. This module will ensure that the student has at least a basic understanding of digital voice, telecommunications concepts, and fundamentals before getting into the real course material.

Digital Voice Background

In the early 1970s, digital loop carrier technology was brought into use. This new type of carrier was based on digital-technology electronics, with the aim of increasing transmission performance through digital technology. This new system was economically driven, offered the voice-quality upgrade that subscribers required, and was more feature-rich than the old analog carrier system. Digital loop technology was the solution for deploying some of the newer services such as digital data service (DDS), and some of the special analog services as well. Digital proved to be more reliable, easier to install, and far less complex to maintain than analog signaling.

Whereas analog consisted of an analog box deployed on the subscriber side, digital extended beyond its subscriber side configuration of basic telephone service along with DDS and a TR57-type suite of electronics by interfacing with a digital system on the network side that brought about significant pair-gain capabilities of approximately 12 to 1. Because digital signals are regenerated, digital signals do not accumulate noise in the way analog signals do. This is one of the primary reasons for converting signals from analog to digital.

Analog versus Digital Signaling

- **Analog signals require one set of wires (2 or 4) per call**
- **Digital signals require <= 64 kbps**
- **Higher quality/speed line can carry up to 1.544/2.048 Mbps (T1/E1)**
- **Digital uses time-division multiplexing (TDM) for more efficient transmission**

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3-4—CVOICE—Digital Voice Technology

Analog signaling represents information as a continuously variable signal quantity such as voltage. Because analog signaling qualifies information this way, the actual analog signaling can represent an infinite number of states.

Digital signaling, however, represents signaling by very discrete or noncontinuous values represented as a sequence of binary digits (bits), usually as 1s and 0s, that may be represented as the presence of an electrical pulse, (1), or no pulse, (0).

Analog signals can be converted to digital signals by a *codec* (coder, decoder). A codec performs this operation by sampling, quantizing, and encoding the signal. Codecs are used to convert voice-frequency channels to 64-kbps digital signal level 0 (DS0) channels.

Digital signals can be *multiplexed* or combined onto one physical medium, which reduces the number of wires needed to transmit multiple phone calls.

Digitizing Analog Signal

- Sample
- Quantization
- Encode
- Compression (optional)

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Digitizing Analog Signal

In order to convert an analog signal to a digital signal, three steps must be performed:

1. Sample the analog signal regularly.
2. Quantize the sample.
3. Encode the value into 8-bit digital form.

(Optional) Compress the samples to reduce bandwidth (multiplexing).

Digitizing analog voice is analogous to taking movie/video of continuous motion. It is possible to reproduce the motion by taking samples of the motion at a regular interval.

Sampling

According to the Nyquist Theorem, the sampling rate should be two times the highest frequency in order to produce playback that appears neither choppy nor too smooth.

Quantizing

Quantizing consists of a scale made up of 8 major divisions, namely 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 compared between the chords. It is the finer graduations at the origin that result in less distortion for low-level tones.

Encoding

PBX output is a continuous analog voice wave form. Digital T1 digital voice is a snapshot of the wave encoded in ones and zeros.

Compressing

(Optional) Although not essential to convert analog signals to digital, compressing signals reduces bandwidth consumption, and is widely used.

Analog to Digital Conversion Process

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

A band-pass filter is used to restrict voice signals to a 300 Hz to 3400 Hz range in order to avoid any potential aliasing conflicts during encoding or decoding processing.

- **Sampling:** The analog signal is sampled at periodic intervals. The output of the sampling step is a *Pulse Amplitude Modulation* (PAM) signal.
- **Quantizing:** The PAM signal is matched to a segmented scale. The purpose of this step is to "measure" the amplitude (or height) of the PAM signal and to assign an integer number that defines that amplitude.
- **Encoding:** The integer base-10 number is then converted to an 8-bit binary number. The output is an 8-bit word in which each bit may be either a 1 (pulse) or a 0 (no pulse).

This entire process is repeated 8000 times per second for a telephone voice channel service. The most commonly used method of converting analog to digital is *Pulse Code Modulation* (PCM).

The optional fourth step, compression, is used to save bandwidth, thereby allowing more voice calls to be carried over a single channel.

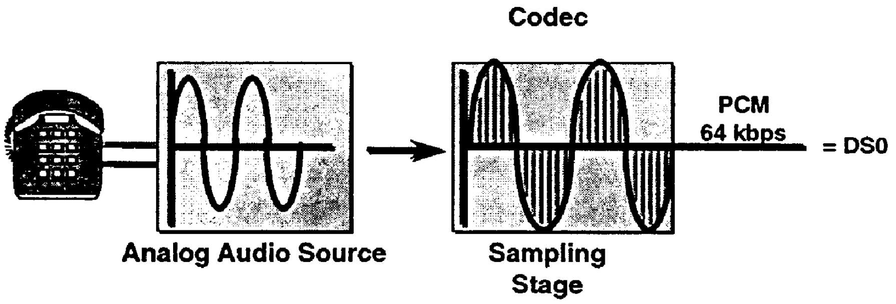
Digital to Analog Conversion Process

After the digital PCM signal is transmitted to the receiving terminal at the far end, it must be converted back to an analog signal.

The process of converting digital signals back to analog signals is described as follows:

- **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.
- **Filtering:** The PAM signal is then passed through a properly designed filter that reconstructs the original analog wave form from its digitally coded counterpart.

Digitizing Voice: Nyquist Theorem



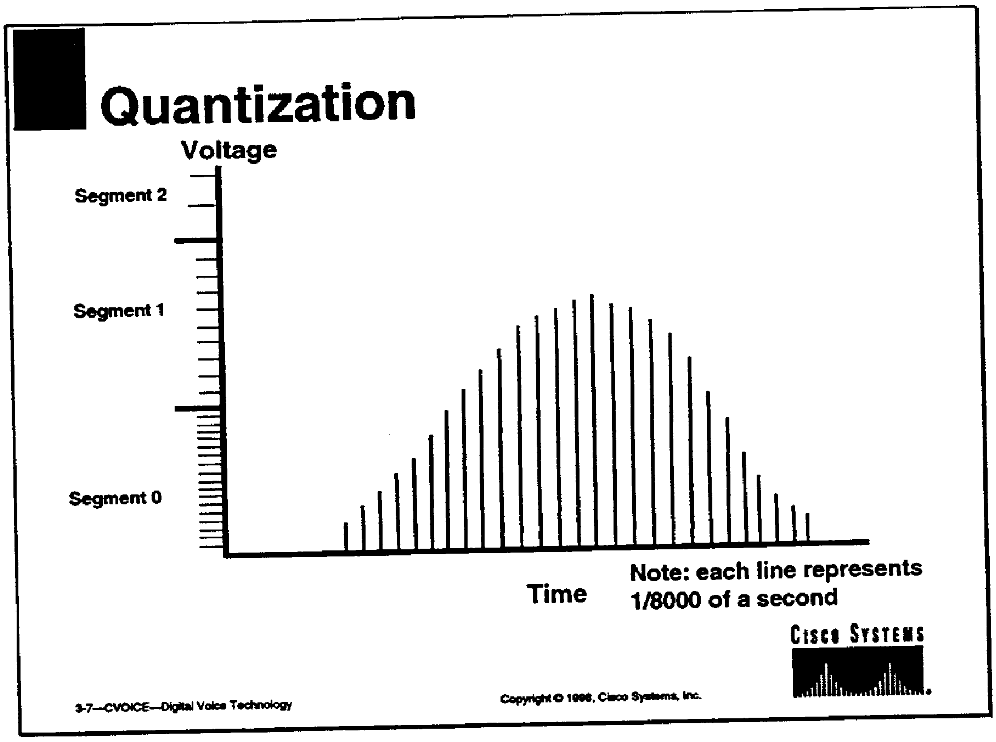
Nyquist Theorem

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

The highest frequency for a voice is 4000 Hz, thus 8000 samples per second (that is, one sample every 125 microseconds).

To calculate the bit rate of digital voice = 2 x 4 kHz x 8 bits per sample

= 64,000 bits per second (64 kbps), that is, a DS0 rate.



Quantization

In the graphic, the x-axis is time and the y-axis is the voltage value (PAM).

Quantization divides the range of amplitude values of an analog signal sample into a set of discrete steps, which are closest in value to the original analog signal. Each step can be given a unique digital code word.

The voltage range is divided into 14 segments (7 positive, 7 negative). Starting with segment 0, each segment has fewer steps than the previous segment, which reduces the noise to signal ratio and makes it uniform. It also closely represents the logarithmic behavior of the human ear. If there is a noise to signal ratio problem, it can be resolved by converting PAM to PCM using a logarithmic scale.

Quantization Technique

- **Linear**
  - **Uniform quantization**
- **Logarithmic quantization**
  - **Companding the signal**
  - **Uniform signal to noise ratio**
  - **Two methods**
    - **A-law (most countries)**
    - **Mu-law (North America, Japan)**

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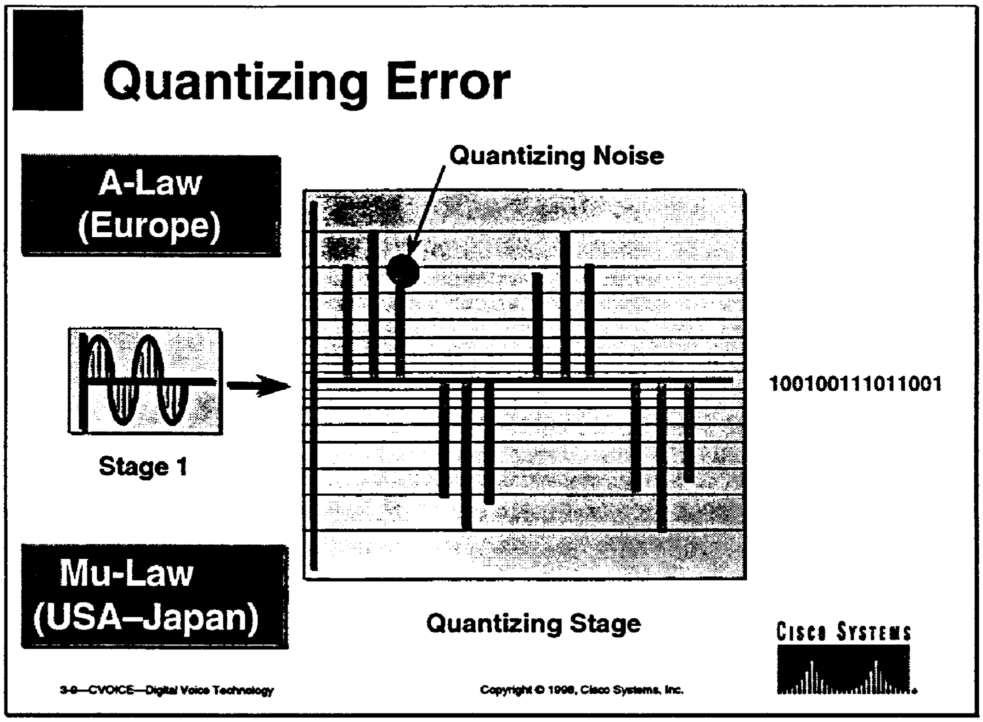
Quantization Technique

Linear sampling of analog signal causes small amplitude signals to have higher noise to signal ratio and therefore poorer quality than larger amplitude signals.

Mu-law and A-law provide a method of reducing this problem by allowing smaller step functions at lower amplitudes and higher ones. Both *compand* the signal. That is, Mu-law and A-law **compress** the signal for transmission, then at the other end **expand** the signal back to its original form, that is, compand the signal.

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

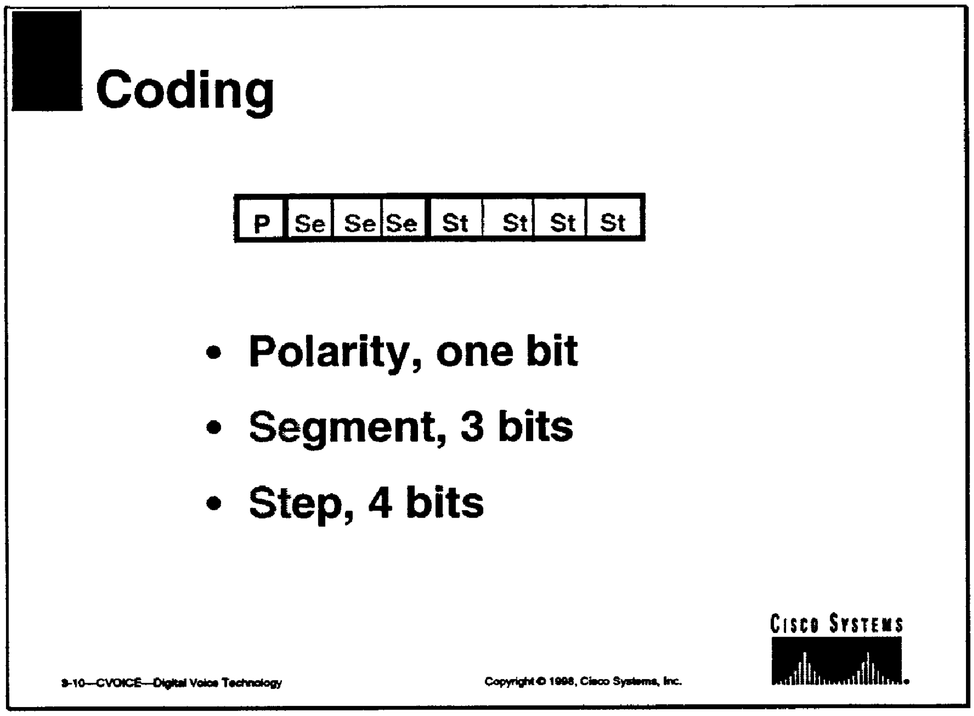


Mu-law and A-law are linear approximations of a logarithmic input/output relationship.

They both generate 64-kbps bit streams using 8-bit code words to segment and to quantize levels within segments.

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

Note When communicating between a Mu-law country and an A-law country, the Mu–law country must change its signaling to accommodate the A-law country.



There are 8000 samples taken per second, which provides an 8-bit code word representation. The 8 bits represent the following:

- Polarity. Polarity represents a positive or negative position above or below the quantizing line.
- Segment. Specifies the segment (voltage) level. There are 16 segments: 8 positive and 8 negative. Each segment is twice the length of the preceding segment.
- Step. Represents the division within the segments, up to 16 steps (gradients or divisions), that is, the step quantizes the segment.

Speech Coding Schemes

- **Wave form coders**
  - **Nonlinear approximation of the actual wave form**
  - **Examples: PCM, ADPCM, Mu-law, A-law**
- **Vocoders**
  - **Synthesized voice**
  - **Examples: LPC, channel, phase**
- **Hybrid coders**
  - **Linear wave form approximation with synthesized voice**
  - **Examples: APC, SELP, CELP**

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Speech Coding Schemes

Terms used in the figure above are defined as follows:

- PCM = Pulse Code Modulation.
- ADPCM = Adaptive Differential Pulse Code Modulation.
- LPC = Linear predictive coding: Precursor to CELP
- Channel = Frequencies divided up into discreet channels, generated at the far end
- Phase = Phase structure analyzed, sent, and generated at the far end
- APC = Adaptive predictive coding: Used together with LPC to come up with CELP
- CELP = Code-excited linear-predictive
- SELP = Self-excited linear-predictive

Wave form coders start with the analog wave form, taking 8000 samples/sec, then determine the most efficient way to code the analog signal for transmission.

Vocoding schemes are very low bit rates (2.4 is possible) but sound very synthetic. In fact, training is necessary to become accustomed to the voice. These are typically military applications where battlefield conditions are unpredictable, yet the synthesized output is always the same. Also, most applications are half-duplex.

Hybrid coders are part of what is called analysis-by-synthesis coding (AbS), which defines all generally used speech coding techniques in the 4.8 kbps to 16 kbps range. Because AbS continuously analyzes and "learns" what to expect a speech wave form should look like in the near term (5 ms) future, hybrid coders are a much higher quality than simple analysis-and-synthesis. The feedback loop allows the codebook to continuously learn.

Voice Compression Benefits/Drawbacks

- **Benefit:**

  Reduce bandwidth consumption

- **Drawbacks:**

  Quantization distortion

  Tandem switching degradation

  Delay (echo)

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Voice Compression Benefits/Drawbacks

Compression is used to reduce bandwidth and storage memory, which in turn reduces the time and cost of transmission. Compression algorithms are optimized for voice.

Quantization delay, tandem switching, and echo are problems inherent to compression. Each of these problems will be addressed later in the course.

Voice Compression Techniques

- **Wave form algorithms**
  - **– PCM: Pulse Code Modulation**
  - **– ADPCM: Adaptive Differential Pulse Code Modulation**
- **Source algorithms**
  - **– LD CELP: Low delay, code-excited linear-processing**
  - **– CSA-CELP: Conjugate structure algebraic code-excited linear-processing**

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Voice Compression Techniques

Voice compression techniques are as follows:

- Wave form algorithms (coders):
  - Sample analog signals 8000 times/second
  - Use predictive differential method to reduce bandwidth
  - Bandwidth reduction highly impacts voice quality
  - Does not take advantage of speech characteristics
- Source algorithms function as follows:
  - Source algorithm coders are called vocoders
  - Vocoders take advantage of speech characteristics
  - Reduce the bandwidth by sending transmitting linear filter setting
  - "Codebook" excitation index. The index is used by receiver to look up a set of excitation values.

Example: Wave Form Compression

- **ADPCM**

  Wave form coding scheme

  Adaptive: automatic companding

  Differential: encode changes between samples only

  ITU standards:

  G.721 rate: 32 kbps = (2 x 4 kHz) x 4 bits/sample

  G.723 rate: 24 kbps = (2 x 4 kHz) x 3 bits/sample

  G.726 rate: 16 kbps = (2 x 4 kHz) x 2 bits/sample

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Example Wave Form Compression

ADPCM is a way of encoding analog voice signals into digital signals by adaptively predicting future encodings by looking at the immediate past.

The adaptive part reduces the number of bits per second that another (more common) method, PCM, requires to encode voice. ADPCM reduces the number of bits required to encode voice signals.

Wave form coders like ADPCM take 8000 samples per second of the analog voice signal and turn them into a linear PCM sample. ADPCM then calculates the predicted value of the next sample from the immediate past sample, and encodes the difference.

The ADPCM process generates 4-bit words, therefore 16 specific bit patterns are generated. The CCITT ADPCM algorithm transmits all 16 possible bit patterns. The ANSI ADPCM algorithm uses 15 of the 16 possible bit patterns (an "all-zeroes" 0000 pattern is not generated).

Example: Source Compression

- **CELP**
  - **Hybrid coding scheme**
- **High quality voice at low bit rates, processor intensive, use of DSPs**
- **G.728: LD CELP—16 kbps**
- **G.729: CSA-CELP—8 kbps** Cisco Routers
  - **G.729a variant—8 kbps, less processor intensive, allows two voice channels encoded per DSP**

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Example: Source Compression

CELP transforms analog voice signals as follows:

- Input to coder is converted from 8-bit PCM to 16-bit linear PCM sample.
- A codebook uses feedback to continuously "learn" and predict the voice wave form.
- Coder is "excited" by a white noise generator.
- Mathematical result (recipe) is sent to far-end decoder for synthesis and generation of voice wave form.

LD CELP is similar to CSA-CELP, except:

- LD CELP uses a smaller codebook and operates at 16 kbps to minimize delay to 2 to 5 ms (no "look-ahead").
- 10-bit codeword is produced from every five samples of speech from the 8-kHz input.
- Four of these 10-bit codewords are called a "subframe," which takes approximately 2.5 ms to encode.
- Two of these "subframes" are combined into a 5-ms block for transmission.

CSA-CELP is a variation of CELP that performs the following functions:

- Codes on 80-byte frames, which takes approximately 10 ms to buffer and process.
- A "look-ahead" of 5 ms is added.
- Noise reduction and pitch-synthesis filtering adds to processing requirements.

G.729 and G.729a Comparison

- Both are ITU standards
- Both are 8 kbps CSA-CELP
- G.729 more complex and processor intensive
- G.729 higher quality than G.729a
- Compression delay the same (10-20 ms)

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G.729 and G.729a Comparison

G.729a Two calls

G.729 One call

There is a fine line between ITU-G.729 and G.729a.

G.729 is the CSA-CELP that Cisco is using for high quality 8-kbps voice in the MC3810, and the UVM in the not too distant future, and also in the 3600 VoIP. G.729 is considered high quality, that is to say that when properly implemented, this 8-kbps compression will sound as good as 32-kbps ADPCM. G.729 is a complex and processor intensive compression algorithm. As a result you can only get one voice channel on a single DSP chip in today's technology. This is not a Cisco situation, but just the current state of DSP technology. It is believed that in the near futre, DSP chips will be able to handle two G.729 voice channels on a single chip.

G.729a, however, is the little brother (or sister) of G.729. It is also an 8-kbps compression, but not quite as high as G.729. It is less complex and can get two channels per DSP chip today. G.729a is not as good as G.729, and is more susceptible to network irregularities (delay, variation, tandeming, etc.).

Compression Techniques Bandwidth Requirement

Standard	Bit Rate (kbps)
G.711, PCM	64
G.726, G.727, ADPCM	16,24,32,40
G.728, LD CELP	16
G.729, CSA-CELP	8
G.729a, CSA-CELP	8

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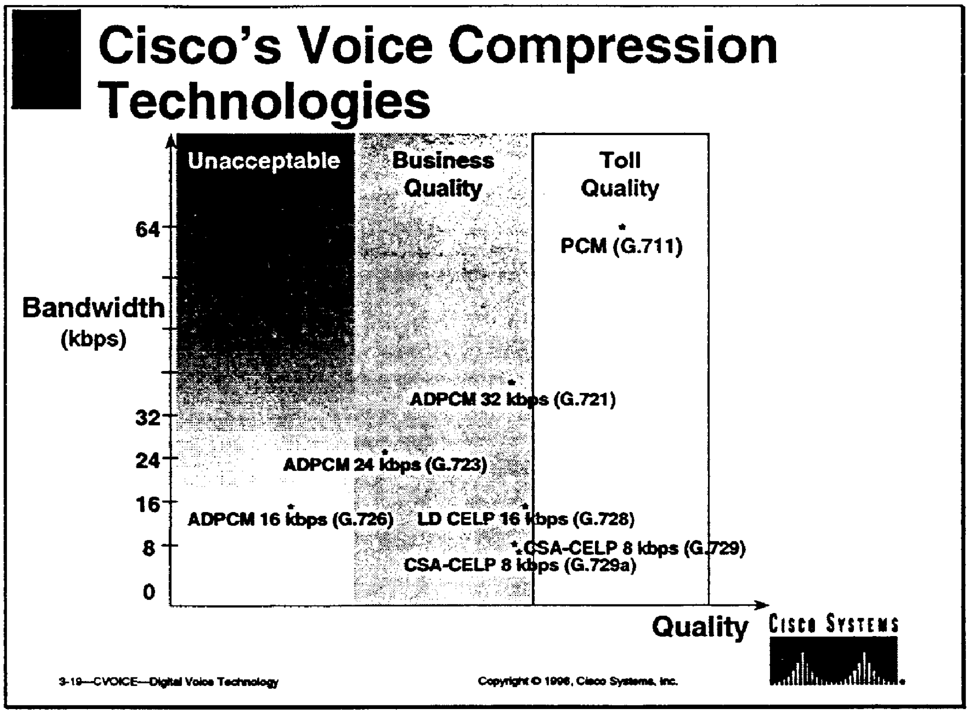
Compression techniques are standardized by CCITT and assigned a number as listed:

- PCM: The amplitude of voice signal is sampled and quantized 8000 times per second. Each sample is then represented by one octet (8 bits) and transmitted. Either A-law or Mu-law is used for sampling to reduce noise/signal ratio.
- ADPCM: In this method the difference between the current sample and its predicted value (made from past sample) is used. This method reduces the bandwidth requirement at the cost of quality of the signal. The sample may be represented by 2, 3, 4, or 5 bits.
- CELP: In all these algorithms, an excitation value and a set of linear-predictive filters (setting) are transmitted. The filter settings transmission are less frequent than excitation values and are sent on as-needed bases.



Cisco Voice Compression Technologies

Digital voice uncompressed is PCM at 64 kbps.

Various methods of voice compression have varying bit rates and subjective "qualities," which are really impairment analysis.

Digital Speech Interpolation (DSI)

- Silence suppression and VAD
- Removal of voice silence
- Examines voice for power, change of power, frequency, and change of frequency
- All factors must indicate voice "fits into the window" before cells are constructed
- Automatically disabled for fax/modem

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Digital Speech Interpolation

Digital Speech Interpolation (DSI) is a type of multiplexing, a way to share bandwidth among a larger number of users than there are circuits for. DSI allocates the silent periods in human speech to active use. Recall that at least 50 percent of a voice conversation is always quiet.

DSI is not good for data transmission because it "clips" the first little bit of every new snippet of conversation, unless you monopolize the channel the whole time by talking incessantly or transmit continuously. If you pause you get clipping as the system drops you and then reconnects you. Clipping can ruin the transmission unless the header knows that DSI is operating and can resend the data.

DSI uses voice activity detection (VAD) and silence suppression.

DSI is similar to statistical multiplexing.

Telephone Voice Quality

- Toll quality
- Transparent quality
- Conversational quality
- Synthetic quality

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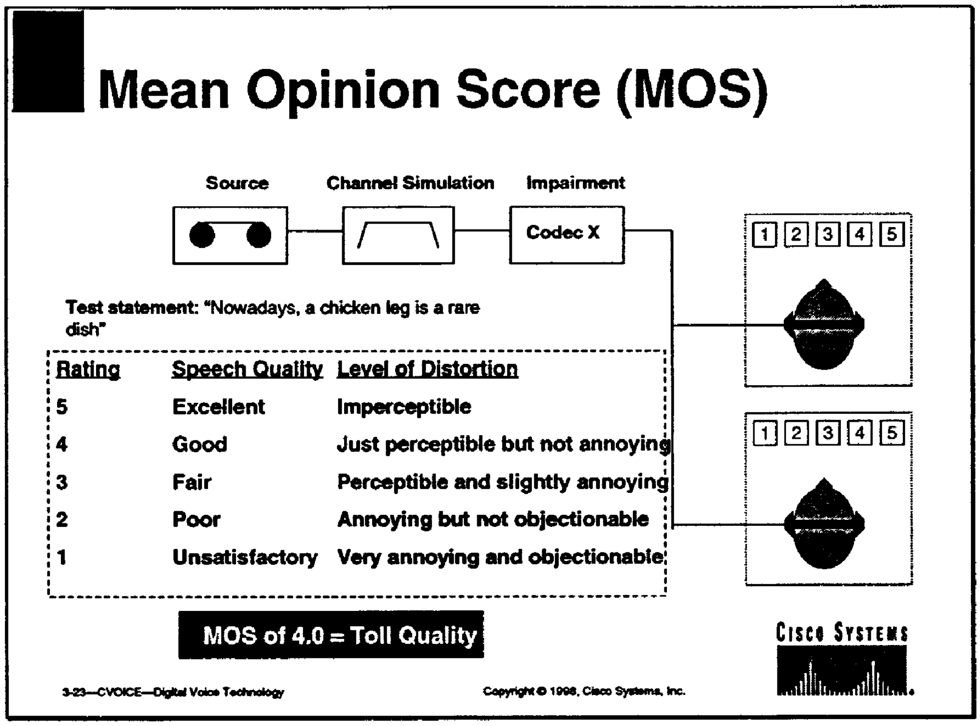
Telephone Voice Quality

Telephone voice quality types are as follows:

- Toll quality. The standard telephone speech signal quality. Bandwidth of 200 Hz to 3200 Hz with a signal/noise ratio (SNR) of > 30 dB and Total Harmonic Distortion (THD) below 2 to 3 percent.
- Transparent quality. Quality similar to that of toll quality. The difference in quality can only be distinguished by direct A/B comparison.
- Conversational quality. Highly intelligible, but with noticeable distortion. Although this type is worse than toll quality, the speaker at the other end can be identified and understood.
- Synthetic quality. More than 80 percent intelligibility, but with degraded quality. The synthetic sound is machine-like, and it may not be possible to identify the far-end speaker.

Voice Quality Measurement



Mean Opinion Score

Mean opinion score (MOS) is a system of grading the voice quality of telephone connections. The MOS is a statistical measurement of voice quality, derived from a large number of subscribers judging the quality of the connection. Graded by humans (very subjective), the range is 1 to 5, where 5 is direct talk.

MOS Rating of Digital Voice

Codec		Bit Rate	MIPS	Comp. Delay (ms)	Framing Size	MOS
G.711	PCM	64	0.34	0.75	0.125	4.1
G.726	ADPCM	32	13	1	0.125	3.85
G.728	LD CELP	16	33	3-5	0.625	3.61
G.729	CSA-CELP	8	20	10	10	3.92
G.729a	CSA-CELP	8	10.5	10	10	3.9
G.723.1	MPMLQ	6.3	16	30	30	3.90
G.723.1	ACELP	5.3	16	30	30	3.8?

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MOS Rating of Digital Voice

Mean opinion score (MOS) testing was conducted by the ITU in order to develop formulas to determine the ratings of the standards (G.721 through G.729).

MOS under Varying Conditions

• Example: G.729	MOS Rating
Average speech level:	3.85
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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MOS under Varying Conditions

Traditional quality standards meet overall loudness rating requirements (G.111), quantization distortion requirements (G.113), and end-to-end delay requirements (G.114).

The new G.113 voice quality standard is as follows:

Itot = Io + Iq + Idte + Idd + Ie
where:

Io: Impairments caused by nonoptimum overall loudness rating OLR or high circuit noise.

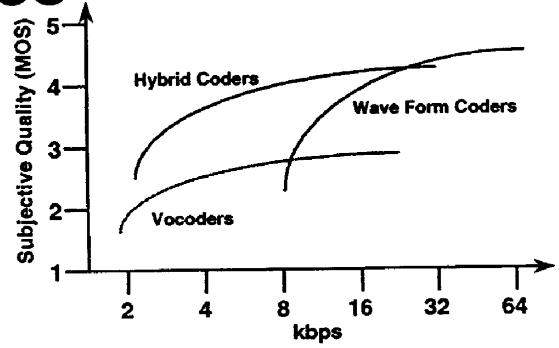
Iq: Impairment caused by PCM-type quantizing distortion.

Idte: Impairments caused by talker echo.

Idd: Speech communication difficulties caused by long one-way transmission times.

Ie: Transmission impairments caused by special equipment in the connection, in particular nonwave form low-bit-rate codecs.

Subjective Impairment Analysis: MOS



Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just Perceptible, Not Annoying
3	Fair	Perceptible and Slightly Annoying
2	Poor	Annoying but Not Objectionable
1	Bad	Very Annoying and Objectionable

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Everyone has the ideal reference for what is considered toll-quality live conversation. Therefore, toll quality on a voice transport system really is the measurement of the impairment. How good does it need to be?

Channel Signaling Types and Frame Formats

Channel Signaling Types

- T1 (defines physical characteristics)
  - AMI/B8ZS (line coding)
  - 100 ohms (impedance)
- DS1 (describes framing characteristics)
  - Frame: 24 DS0s or channels
  - SF (superframe): 12 frames (D4 framing)
  - ESF (Extended Super Frame): 24 frames

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T1/DS1

T1

T1 identifies the physical attributes of a 1.544-Mbps transmission medium.

Alternate Mark Inversion (AMI) is the line code used for T1 (and E1) lines where the "1s" or "marks" on the line alternate between positive polarity and negative polarity.

Bipolar with Eight Zero Substitution (B8ZS) is a T1 line protocol that converts a channel word with eight consecutive zeros into a code that, at the far end, is converted back to eight zeros. It allows 64-kbps clear channel operation while assuring the ones density required on the T1 line.

DS0

Digital Service level 0 is the smallest unit of transmission defined in the hierarchy. It is a circuit or channel of 64,000 bps. A DS0 channel can carry one digital PCM voice call. A total of 24 DS0s will be multiplexed together to form the next level called DS1.

DS1

Digital Service level 1 (DS1) is a circuit of 1.544 Mbps. It is the level of service output from a channel bank. The terms DS1 and T1 are often confused. T1 is actually the facility that the DS1 stream will be transmitted over. A DS1 carries 24 8-bit byte DS0s at 1.544 Mbps. T1 can be offered as a straight "pipe" or circuit at 1.544 Mbps with no defined format of the bits. The pulses on DS1 circuits are in bipolar AMI format.

Channel Signaling Types (cont.)

- **E1 (physical characteristics)**
  - **AMI/HDB3 (line coding)**
  - **75/120 ohms (impedance)**
- **E1 (framing characteristics)**
  - **Frame: 32 channels or E0s**
  - **MF (multiframe): 16 frames**

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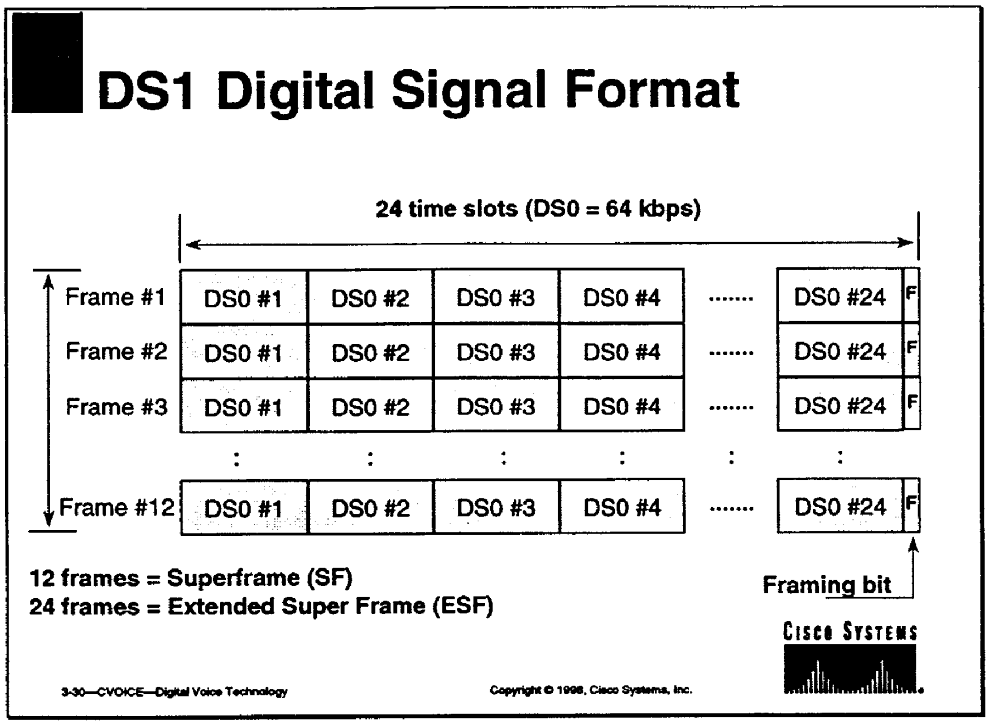
E1

Similar to T1/DS1, E1 defines the European digital transmission format.

E1 consists of 32 64-kbps time slots or channels that make up a transmission rate of 2.048 Mbps. Of these 32 channels, 30 are used for voice and data transmission. Channel 1 (time slot 1) is used for framing purposes. Time slot 16 is used for channel signaling.

AMI is the line code used in both T1 and E1 lines where the "1s" or "marks" on the line alternate between positive polarity and negative polarity.

High Density Bipolar 3-bit (HDB3) is a new line interface for E1, similar to B8ZS for T1, which eliminates patterns with eight or more consecutive zeros. It allows for 64-kbps clear channel operation and still assure the ones density required on the E1 line.



DS1 Digital Signal Format

The DS1 digital signal format characteristics are as follows:

- A DS1 frame is 193 bits long, made up of 8 bits from each of the 24 time slots (DS0s) plus 1 bit for framing. A DS1 repeats every 125 microseconds, resulting in 8000 samples per second. (8 bits x 24 time slots, plus 1 framing bit, x 8000 samples/second = 1.544. Mbps.)
- There are two major framing/format standards for T1:
  - D4 specifies 12 frames in sequence as a *superframe*. D4 framing pattern is 100011011100 and synchronizes within 4 frames D4 superframe.
  - Extended Super Frame (ESF) format is dominant in the public and private networks. Both types of formats retain the basic frame structure of 1 framing bit followed by 192 data bits.
- This DS1 signal format is referred to as the *D4* frame format.
- The 193rd bit of each DS1 frame is used for frame synchronization.

Extended Super Frame

Extended Superframe Format

	S Bits			Bits Used in DS0		Signaling Bit Option			
Frame #	Fe	DL	BC	Traffic	Signaling	T	2	4	16
1	-	m	-	Bits 1-8					
2	-	-	C1	Bits 1-8					
3	-	m	-	Bits 1-8					
4	0	-	-	Bits 1-8					
5	-	m	-	Bits 1-8					
6	-	-	C1	Bits 1-8	Bit 8	-	A	A	A
7	-	m	-	Bits 1-8					
8	0	-	-	Bits 1-8					
9	-	m	-	Bits 1-8					
10	-	-	C1	Bits 1-8					
11	-	m	-	Bits 1-8					
12	1	-	-	Bits 1-8	Bit 8	-	A	B	B
13	-	m	-	Bits 1-8					
14	-	-	C1	Bits 1-8					
15	-	m	-	Bits 1-8					
16	0	-	-	Bits 1-8					
17	-	m	-	Bits 1-8					
18	-	-	C1	Bits 1-8	Bit 8	-	A	A	C
19	-	m	-	Bits 1-8					
20	1	-	-	Bits 1-8					
21	-	m	-	Bits 1-8					
22	-	-	C1	Bits 1-8					
23	-	m	-	Bits 1-8					
24	1	-	-	Bits 1-8	Bit 8	-	A	B	D

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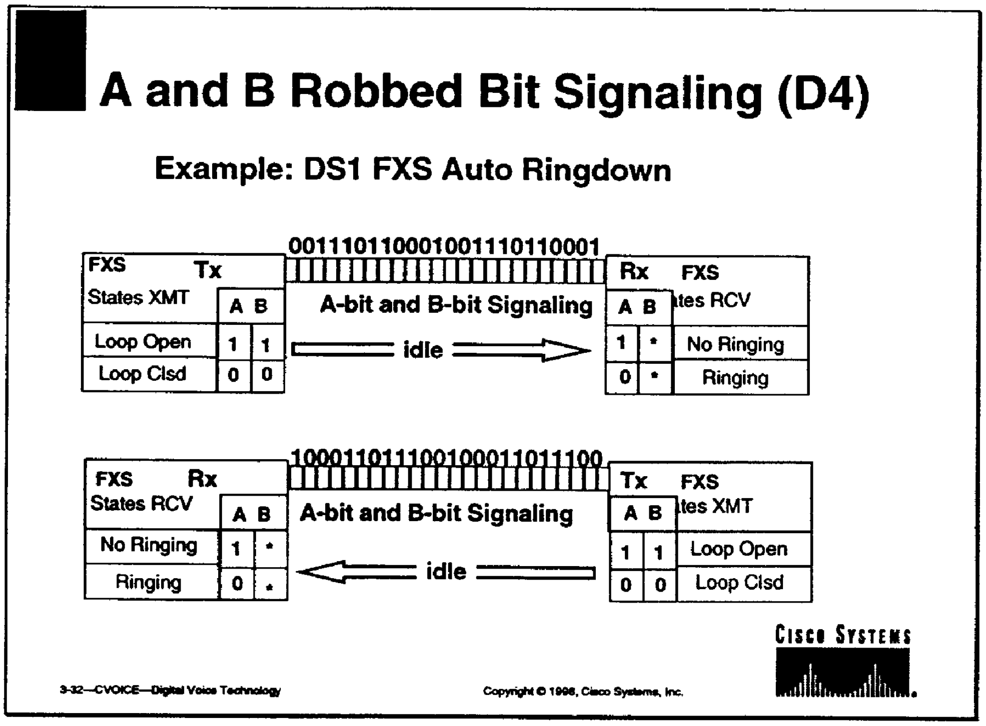
Extended Super Frame

The ESF format is as follows:

- 24 frames in a row, in the same time cycle of 125 microseconds, with framing, and a cyclical redundancy check (CRC).
- ESF framing pattern is 001011, in frames 4, 8, 12, 16, 20, and 24. This is represented by the FE (framing) column in the figure above. Note the pattern: 001011.
- The BC column is for block checking, and the DL column is for data link.
- Channel associated signaling (CAS) robs the least significant bit (LSB) of every byte in frames 6, 12, 18, and 24 for ABCD bits. ABCD signaling option providing additional control/signaling information. ABCD signaling can represent 16 different signaling states or control information.

Note

Fe = Extending Framing (sequence 001011)
Dl = 4 kbps Data Link (message bits "m")
BC = Block Check Field (check bits C1-C6)
Option T = Transparent (bit 8 for traffic)
Option 2 = 2-State Signaling (Channel A)
Option 4 = 4-State Signaling (Channel A, B)
Option 16 = 16-State Signaling (Channels A,B,C, D)

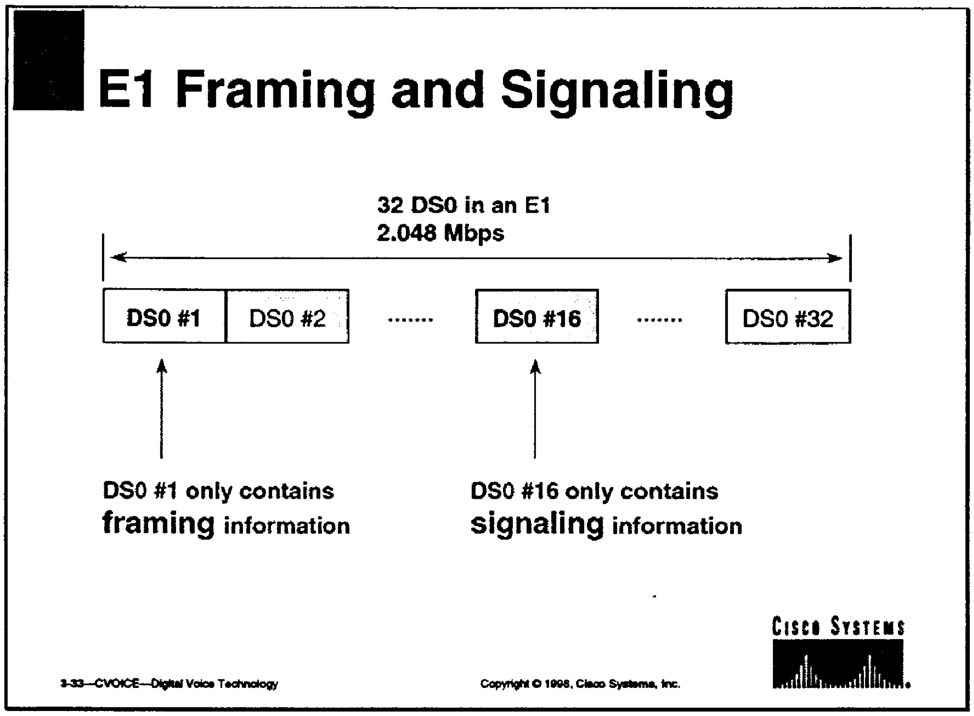


A and B Robbed Bit Signaling (D4)

A and B bits provide near- and far-end off-hook indication for control signaling.

A superframe uses *robbed bit signaling* in frame 6 and 12 for control signaling. These two robbed bits are identified as "A" bit and "B" bit, or *A and B bit signaling*. The two bits can represent different signaling states or control features (on/off hook, idle, busy, ringing, and addressing). The robbed bits are the least significant bit (LSB) from an 8-bit word. The effect on quality is minimal.

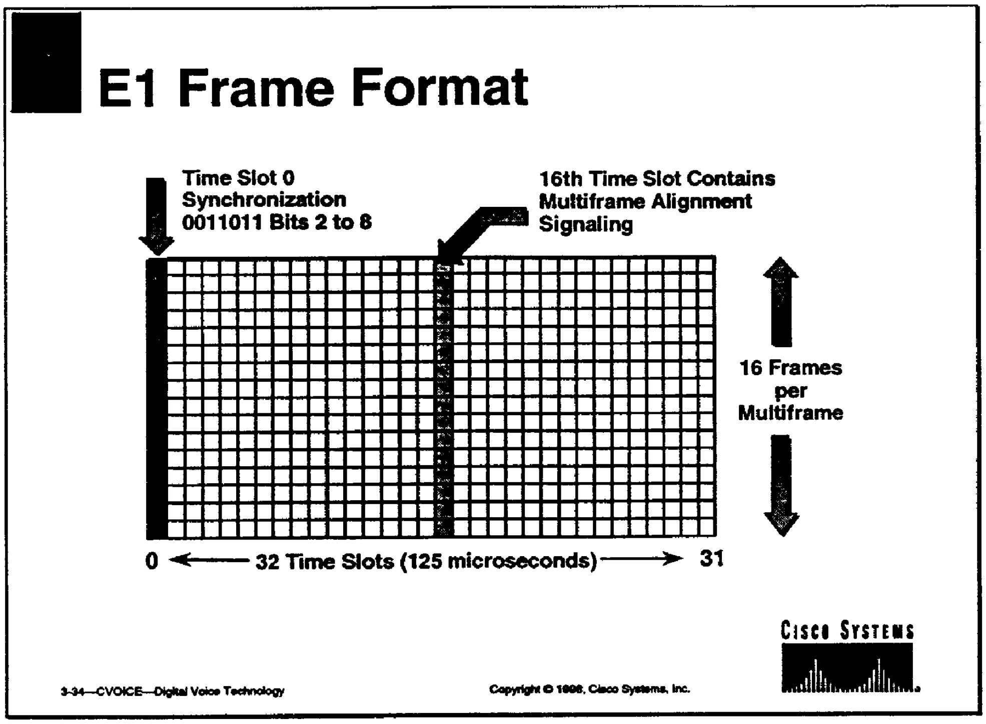
ESF also uses robbed bit signaling (in frames 6, 12, 18, and 24), which yields ABCD signaling options, providing additional control/signaling information. The lost bits are the least significant bits. Their effect on the quality is insignificant.



E1 Framing and Signaling

In E1 framing and signaling, 32 channels or time slots are available, 30 are used for voice and data. Time slot 1 is used for framing information. Time slot 16 is used for signaling for all the other time slots.

Note Remember: The DS0s are numbered 1 through 32, while the time slots of the E1 frame are numbered 0 through 31.



E1 Frame Format

In E1 frame format, 32 time slots make up a frame. 16 E1 frames make up a *multiframe.*

The time slots are numbered 0 though 31.

- Time slot 0 carries only framing information.
- Time slot 16 carries only signaling information for all the other time slots.
- The other 30 time slots carry voice and data traffic.

E1 Line Coding

- **Unipolar**
  - **NRZ: Nonreturn to Zero**
  - **RZ: Return to Zero**
- **Bipolar**
  - **AMI**
  - **HDB3**
    - » **Violation pulse if more than three zeros**
    - » **Highest quality and most efficient**

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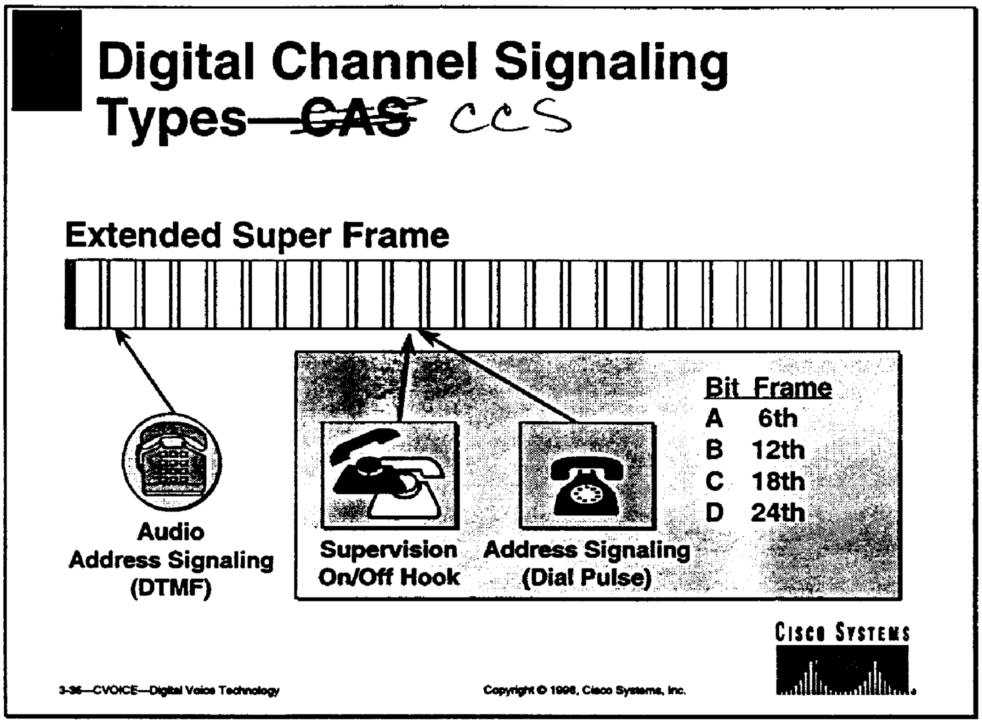
E1 Line Coding

In unipolar coding, ones are represented by positive voltage (3 volts).

In bipolar, ones are represented by alternate positive and negative voltage (3, -3), which balances the voltage potential on the circuit and improves the quality of the signal.

HDB3 eliminates timing, which may be caused by consecutive zeros. It adds a violation pulse after each three zeros, which are removed by the receiving device.

Digital Channel Signaling Types (CAS and CCS)



Channel Associated Signaling

Channel associated signaling (CAS) sends its signal for call setup in the same channel as a voice call.

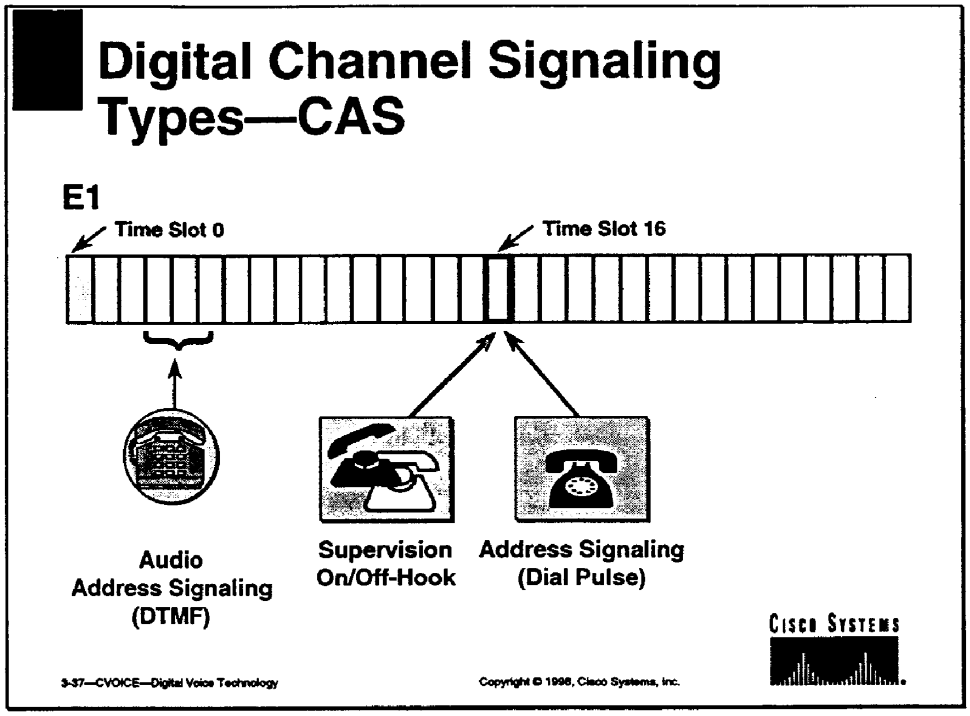
A and B bits are used to map E&M leads and provide call supervision (on-hook and off-hook).

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

Dial (pulse) must be carried with A/B bit signaling.

DTMF (tone) can be carried in-band in the actual audio path.

A CAS signaling example is as follows: T1/DS1.



Common Channel Signaling

Common channel signaling (CCS) differs from CAS signaling in that there is a separate channel used for call setup.

For DTMF (tone), dialing is carried in the audio path.

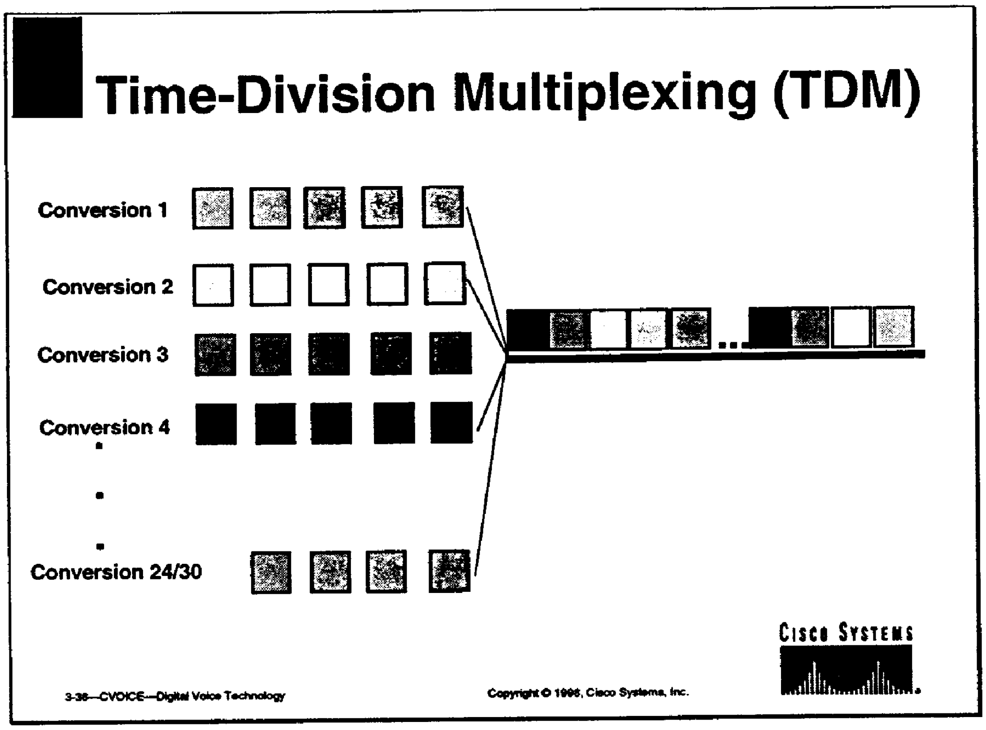
Supervision and address signaling (pulse) are carried in a separate channel (time slot 16).

CCS signaling examples are as follows:

- E1
- ISDN
- DPNSS
- Q.SIG
- SS7

Note

ISDN: Integrated services, digital network (discussed later in this chapter).
DPNSS: Digital private network signaling system
Q.SIG and signaling system 7 (SS7) will be discussed later.



Time-Division Multiplexing (TDM)

TDM is a technique for transmitting a number of separate voice signals simultaneously over one communications medium by quickly interleaving a piece of each signal one after another. Information from each data channel is allocated bandwidth based on preassigned time slots, regardless of whether there is data to transmit.

TDM samples each voice conversation, interleaves the samples, sends them on their way, then reconstructs the several conversations at the far end. There are different ways of sampling. One method samples 8 bits (1 byte) of each conversation (called word interleaving).

Another method is to sample just 1 bit (called bit interleaving).

Digital Telephony—T1 and E1/J1

	T1 (ITU-T G.733)	E1/J1 (ITU-T G.732)
Sampling Frequency	8 kHz	8 kHz
Channel Bit Rate	DS0—64 kbps	DS0—64 kbps
Time Slots per Frame	24	32
Channels per Frame	24	30
Bits per Frame	24 x 8 + 1 = 193	32 x 8 = 256
Framing	D4/Superframe (12) Extended Super Frame (24)	E1: Multiframe (16) J1: CRV in bit 1 of frame
Framing Indicator	193rd bit of frame	2.048 kbps word of 7 bits in the 0 channel of odd frames
System Bit Rate	8000 x 193 = 1.544 Mbps	8000 x 256 = 2.048 Mbps
Signaling	"Robbed bit" channel associated signaling D4/Superframe — Extended Super Frame LSB/channel — LSB/channel Frame 6 and 12 — Frames 6, 12, 18, 24	E1: CCS in time slot 16 CAS in time slot 16—2 channels every other frame J1: time slot 0

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The figure shows a one-page, quick reference summary table of digital telephony for the three major digital telephony systems in use.

Note J1 denotes the digital transmission system for Japan.

Digital Telephony—Synchronization

- **Bit synchronization**
  - Primary reference source
  - Ones density (except for J1/CMI)
- **Time slot synchronization**
  - Bits/byte/channel
- **Frame alignment**
  - Basic rule
  - 193rd bit pattern

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Digital Telephony—Synchronization

Bit synchronization operates the transmitter and receiver at same bit rate so bits are not lost.

Ones density is no more than 15 zeros in a row on a transmission. The most important ones density must be at least 12.5 percent of the transmission.

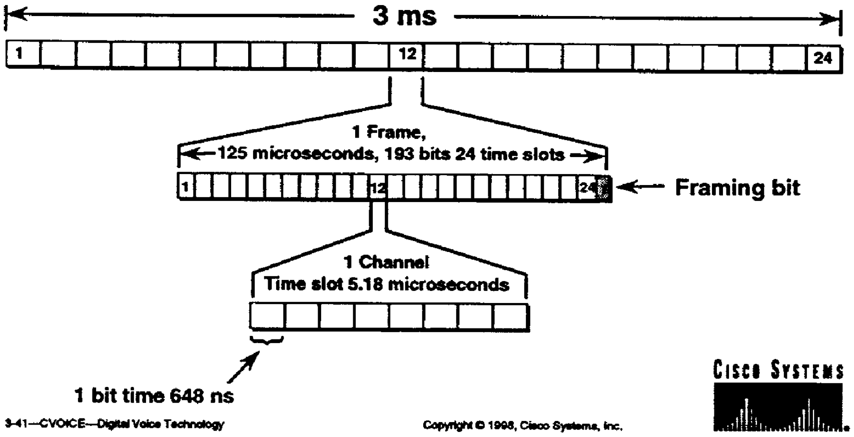
For a voice PCM network, robbed bit signaling renders the 8th bit in all channels unreliable, whether it is used for signaling or not. So the 8th bit is discarded or ignored and forced to a 1 state for data transmission, causing 56-kbps channels.

Time slot synchronization is the phase alignment between the transmitter and receiver so the time slots are lined up for information retrieval for each bit, byte, and channel.

Frame level synchronization operates by phase alignment between the transmitter and receiver so the time slots are lined up for information retrieval by frame. This is where the 193rd bit helps in frame alignment.

Digital Telephony—Synchronization (cont.)

One DS1 ESF



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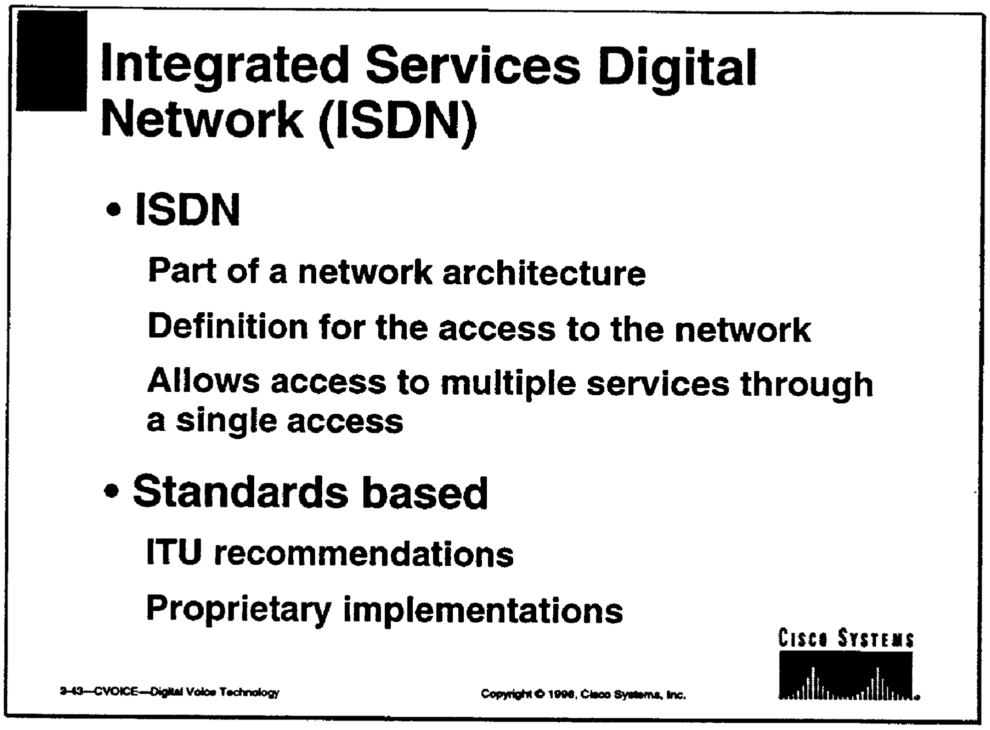
To help explain the critical nature of network synchronization in a T1 system, consider that the receiver must be able to detect each bit as it appears on the link interface at the receiver.

The extremely small slices of time to build just one DS1 ESF are as follows:

- Each bit is only 648 ns in duration.
- Each channel time slot (DS0) is only 5.18 microseconds in duration.
- One frame, made up of 24 channel time slots (DS0s) plus a framing bit, is only 125 microseconds in duration.
- One ESF made up of 24 frames is only 3 milliseconds (ms) in duration.



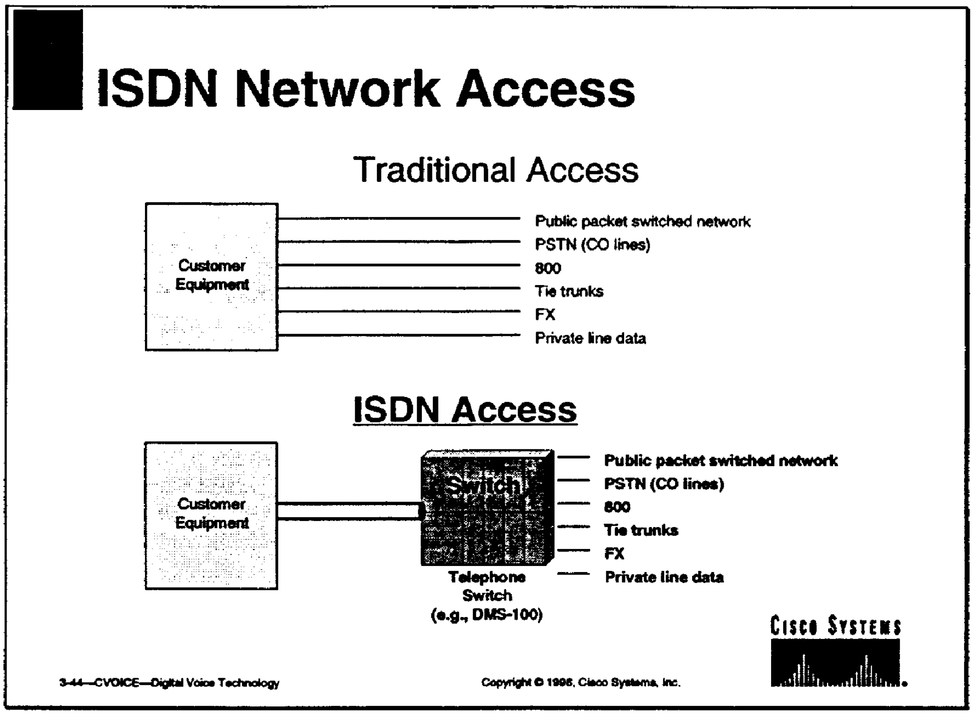
Integrated Services Digital Network (ISDN)

ISDN is an access specification to a network.

The benefits of ISDN are:

- Standards-based voice features such as call forward
- Standards-based network voice features: NRAG; network reason display
- Standards-based enhanced "dialup" capabilities: SW-384K, Group IV fax, audio channels

There are also proprietary implementations of ISDN.



Access features of ISDN include:

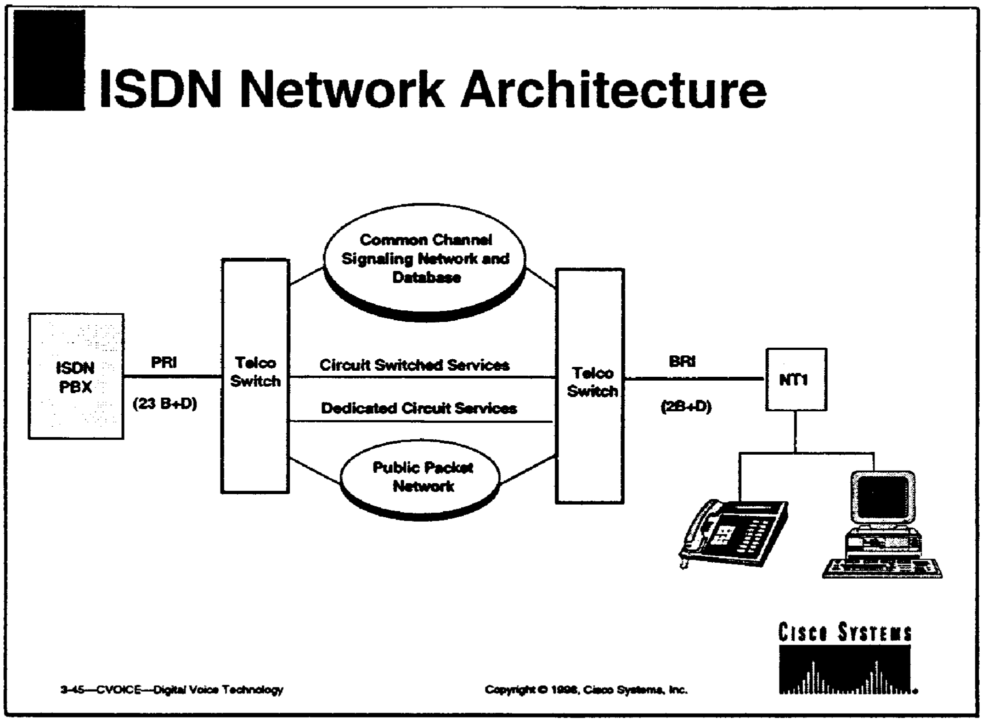
- Fast call setup over all-digital network
- More bandwidth to the desktop
- More efficient use of access trunks reduces number of access trunks required and trunk holding time
- Potential for combining voice and data accesses reduces number required
- CCS network provides a standard out-of-band signaling
- Standards-based for voice, data, and signaling

The PSTN includes such services as direct inward dial (DID), direct outward dial (DOD), and wide-area telephone service (800 WATS).

Calls to and from PSTN include local, direct distance dial (DDD), 800, 900, and 976.

Traffic is routed to and from a PBX to a telco switch over ISDN access. The D channel advises the PBX, or telco switch, of the call information. Traffic types are not tied to a specific B channel. Customers subscribe to "virtual" connections in the telco switch.

FX terminates on a telco switch instead of a PBX.



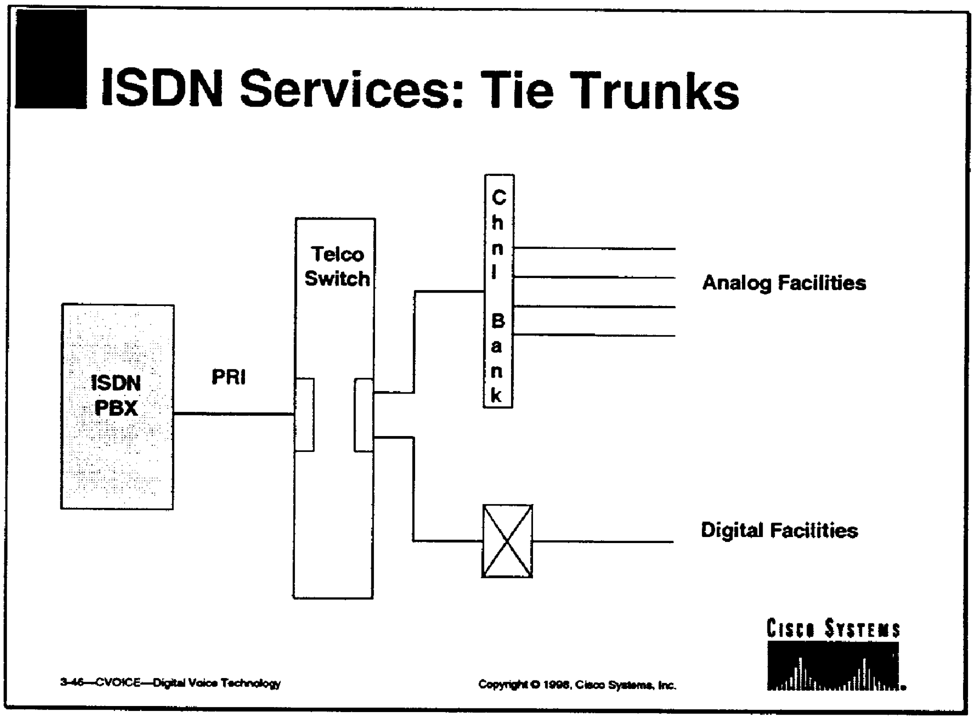
ISDN Network Architecture

The bearer channel (B channel) carries information (voice, data, video, etc.) 64 kbps (DS0).

The signaling channel (D channel) carries instructions between customer equipment and network, and carries information. The D channel can also carry packet switch data (X.25) for the public packet switched network at 16 kbps or 64 kbps.

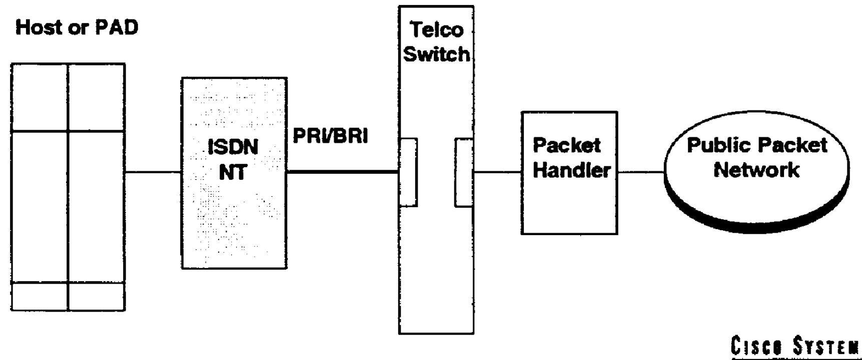
Basic Rate Interface (BRI) is designed to operate using the average local copper pair. It is represented as: 2 B + D (where 2 x 64 kbps + 16 kbps = 144 kbps (not including overhead)).

Primary Rate Interface (PRI) is designed to operate using DS1. It has an optional backup D channel. It is represented as: 23 B + D (where 23 x 64 kbps + 64 kbps = 1.536 Mbps (not including overhead)). In Europe, the PRI channel is 30 B + D.



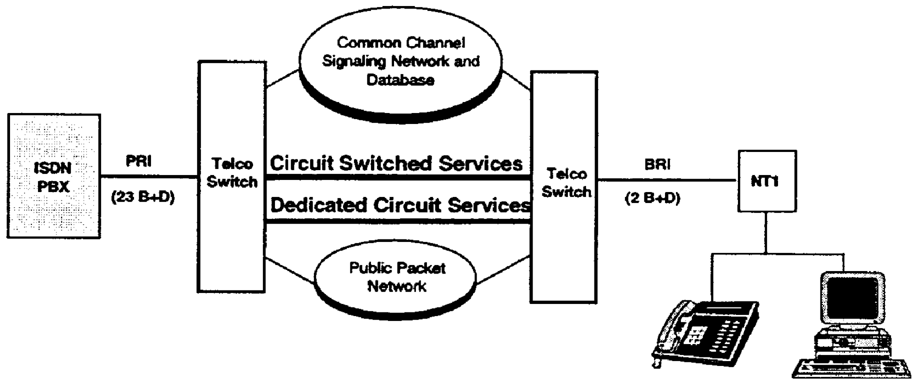
Tie trunks terminate on the telco switch. Using the D channel, calls to and from a PBX define the call as a tie trunk call. Call signaling is converted to in-band signaling or analog.

ISDN Services: Packet-Switched Data



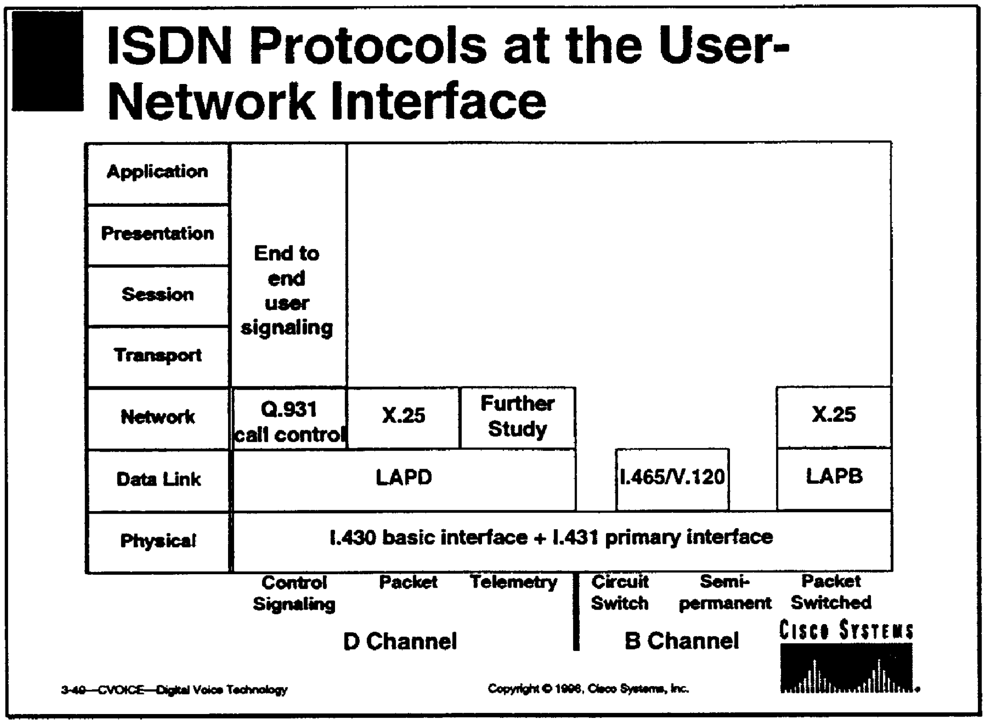
When using the B channel, connection to packet network is nailed up. The D channel can provide dynamic access to a packet network.

ISDN Services: Circuit Switched and Dedicated Data



Circuit switched data specifications permit DS0, 2 x DS0, 6 x DS0, and 24 x DS0 switched data. They are excellent for applications requiring temporary bandwidth

For dedicated data, the B channels can be "nailed up" to provide n x DS0 access to dedicated networks.



This figure shows the ISDN protocols in effect at the user-network interface for both the D channel and B channel.

Layer 3 (Q.930/931) Messages

- Call establishment
- Call information phase
- Call clearing phase
- Miscellaneous

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Layer 3 (Q.930/931) Messages

Layer 3, Q.930/931, uses a standard set of messages to communicate. These standard commands cover the following areas: call establishment, call information, call clearing, and other miscellaneous services.

- Call establishment is used to initially set up a call. Messages traverse between the user and the network and vice versa. Call establishment includes the following functions: Alerting, Call proceeding, Connect, Connect acknowledgement, Progress, Setup, and Setup acknowledgement.
- Call information phase is sent between user and the network after the call is established. Possible uses might allow a user to suspend and then resume a call. The following occur in the call information phase: Hold, Hold acknowledgement, Hold reject, Resume, Resume acknowledgement, Resume reject, Retrieve, Retrieve acknowledgement, Retrieve reject, Suspend, Suspend acknowledgement, Suspend reject, and User information.
- Call clearing terminates a call. The following events occur in the call clearing phase: Disconnect, Release, Release complete, Restart, Restart acknowledgement.
- Various miscellaneous messages negotiate network features (supplementary services). Some of the miscellaneous services include Congestion control, 'Facility, Information, Notify, Register, Status, and Status inquiry.

Q.SIG Protocol

Layer				
Layer 4-7	ROSE: Remote Operation Service Elements ACSE: Association Control Service Elements			End-to-end protocol network transparent
Network	In progress			Q.SIG procedures for supplementary svcs
	ISO 11582, ETS300, 239 ECMA165			Q.SIG generic functional procedures
	ISO 11574, ETS200 171/172, EDMA142/143			Q.SIG basic call
Link Layer	ECMA141, ETS300 402			Interface Dependent Protocols
Physical	Basic Rate I.430	Primary Rate I.431		
Media	Copper	Copper	Optical	

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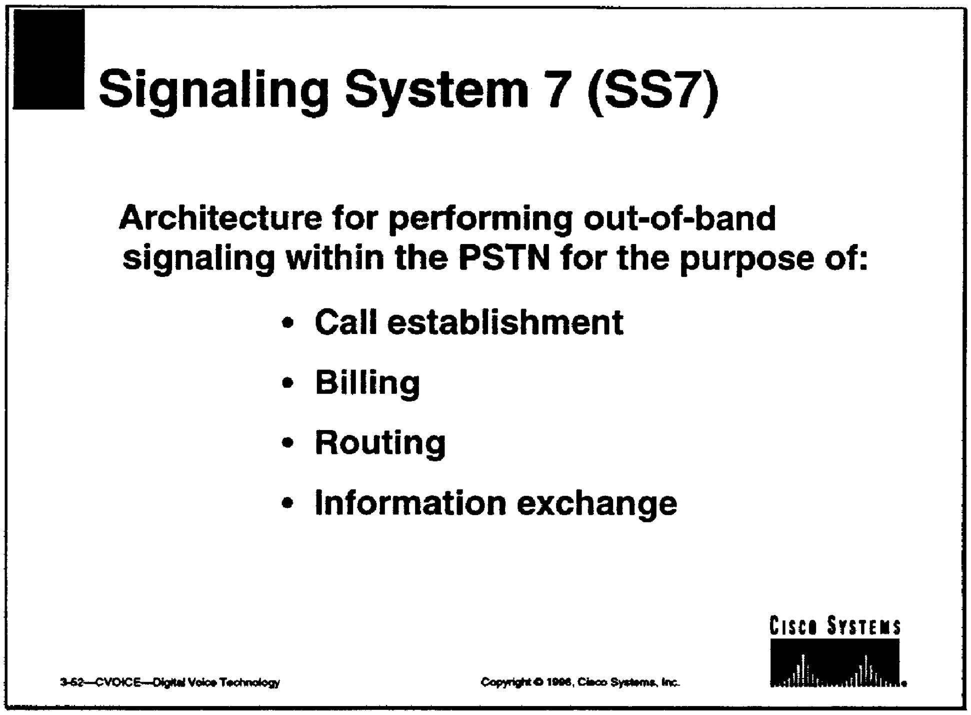
Q.SIG Protocol

Q.SIG is based on ISDN's protocols for signaling (Q.931/932), but differs in that Q.SIG also covers internetwork signaling.

Q.SIG covers:

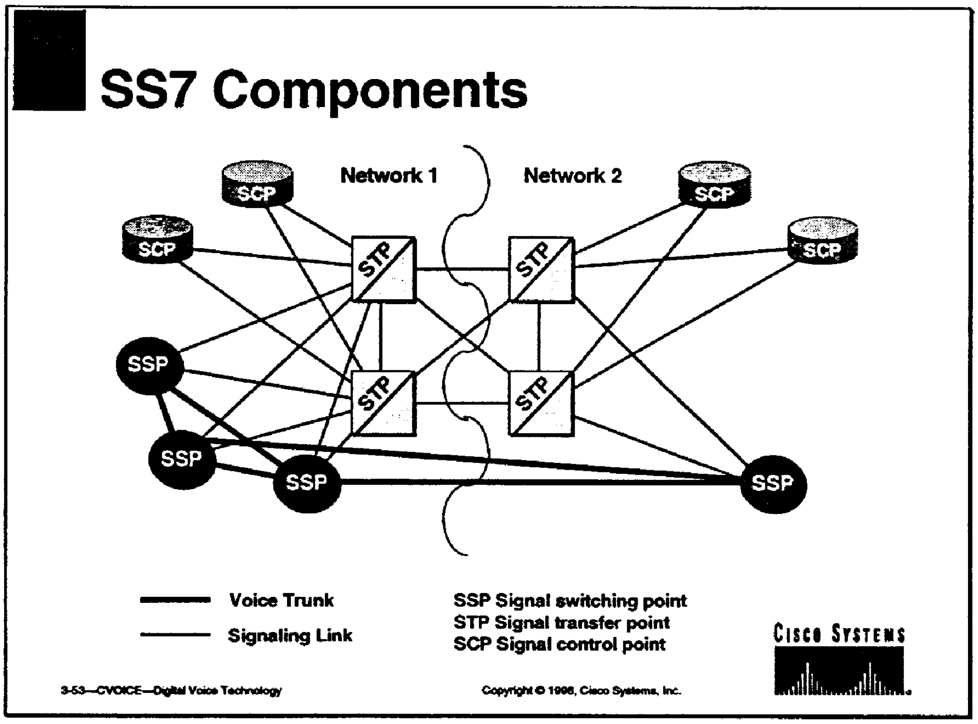
- Peer-to-peer connections between switching entities
- Protocol extensions for transit nodes
- Protocol extensions for transparent transport of signaling between end nodes



Signaling System 7

Signaling system 7 (SS7) was created by CCITT in 1981. The primary functions and benefits of SS7 are as follows:

- Fast call setup by high-speed circuit-switched connections.
- PBX transaction capabilities, (i.e., call forwarding, call waiting, call screening, call transfer) extended to the entire network.
- Dedicated control channel for all signaling functions.
- Only one set of signaling facilities is needed for each associated trunk group.
- Information such as address digits can be transferred directly between control elements.
- SS7 is replacing the older common channel interoffice signaling (CCIS).
- 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.
- The channel used for common channel signaling need not be associated with any particular trunk group.

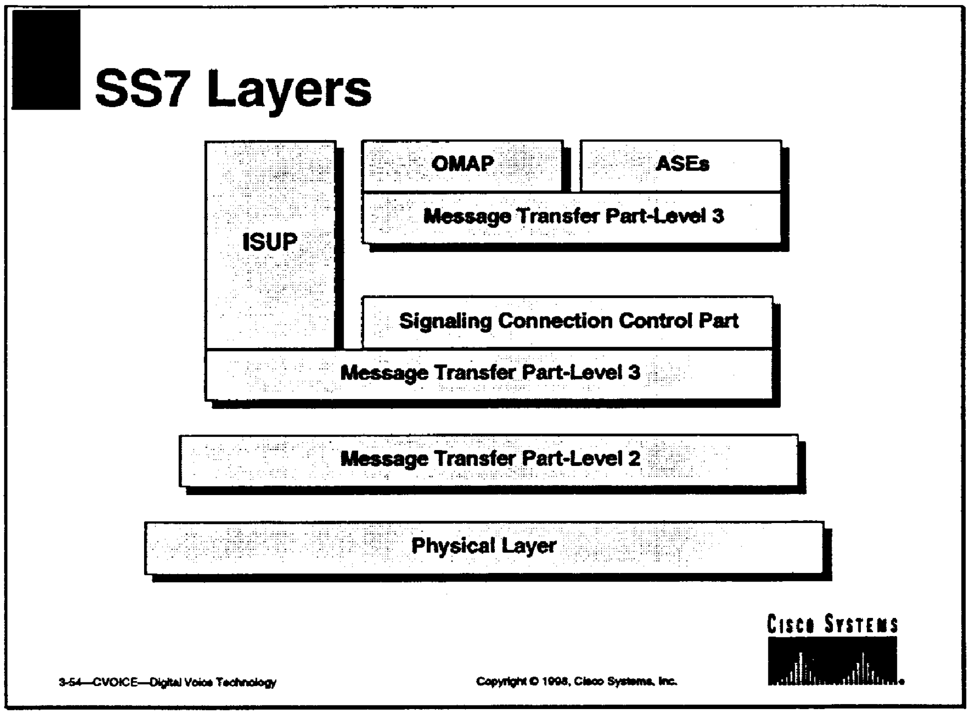


Signal transfer points are the packet switches of the signaling network. They receive and route signaling messages to the proper destination. They are used together in redundant pairs.

Signal control points are databases used in the network for advanced call processing capabilities. Examples of such services include 800 number mapping and credit card usage. They are usually deployed in pairs, but not always.

SSPs are telephone switches (end office or tandem). They generally originate, terminate, or switch voice calls. Each SSP has a link to each of the mated STPs. All SS7 communications are sent out over these links.

Mated STPs are connected to other mated STPs via four links (or sets of links). These links are referred to as a quad.



The physical layer uses DS0 channels and carries raw signaling at a rate of 56 kbps or 64 kbps.

Message transfer part, level 2 (MTP-L2) provides link layer functionality between two endpoints. Capabilities include error checking, flow control, and sequence checking.

Message transfer part, level 3 (MTP-L3) provides network layer functionality. It ensures that messages can be delivered across the SS7 network. Node addressing, routing, alternate routing, and congestion control are all part of this level.

Signaling connection control part (SCCP) takes it one step further. It allows for communications to applications that exist in nodes. Examples of subsystems include 800 call processing, calling card processing, and advanced intelligent network functions.

SS7 Messages and Protocols

- **ISDN User Part (ISUP)**
- **Transaction Capabilities Application Part (TCAP)**
- **Operations, Maintenance, and Administration Part (OMAP)**
- **Application Service Elements (ASEs)**

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ISUP defines the establishment and tearing down of voice and data calls. ISUP is used for both ISDN and non-ISDN calls. The North American version of SS7 uses MTP to transport messages

TCAP defines the messages and protocol used to communicate between applications in nodes. It is used for database services such as 800 and calling card. Because messages must be delivered to individual applications, SSCP must be involved.

OMAP include those capabilities used for validating routing tables and diagnosing link troubles. Messages use both MTP and SSCP.

SS7 Message Formats

FLAG	BSN/BIB	FSN/FIB	Length	Check

Fill-In Signal Unit

1 to 2 bytes

FLAG	BSN/BIB	FSN/FIB	Length	Status	Check

Link Status Signal Unit

8 to 272 bytes

FLAG	BSN/BIB	FSN/FIB	Length	Svc Info	Sig Info	Check

Message Signal Unit

BSN/BIB - Backward Sequence Number/Backward Indicator Bit
FSN/FIB - Forward Sequence Number/Forward Indicator Bit
Length - Number of octets between itself and the check sum
Svc Info - Is used to indicate the type of signaling message that follows

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The Service Info. field is an 8-bit field that is broken down into the following:

- Four bits for the type of information contained in the Signaling Info. field. The values and what they represent follow:

  0 (Signaling Network Mgmt)

  1 (Signaling Network Testing and Maintenance)

  3 (Signaling Connection Control Part)

  5 (ISDN User Part (ISUP))

- Two bits are used to indicate whether the message is intended for use in a national or international network.
- Two bits are used (United States) to indicate priority, with three being the highest. Thy only apply to congestion and not transmittal priority.

The first portion of the Signaling Info. field is the routing label and is made up of seven octets:

- Three to identify the destination point code.
- Three for the originating point code.
- One for the signaling link selected.

Point codes consist of a three-part identifier: network number, cluster number, and member number.

Summary

Summary

- Review, identify, and define digital telephony fundamentals and basics
- Contrast digital and analog signaling
- Identify and contrast the different digital frame formats, signaling formats, and coding

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Summary (cont.)

- Compare the various levels of voice quality
- Categorize the various types of digital voice compression
- Examine ISDN and identify the basic components of this digital architecture

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Written Exercise

Questions

Answer the following questions.

Q1) What are the four steps in digitizing analog signals?

Q2) Describe the differences between digital and analog signaling.

Q3) List the two main types of compression algorithms. List at least three attributes of each type of algorithm.

Q4) In DS1 framing format, how many frames are there in a superframe? How many frames in an Extended Super Frame (ESF)?

Q5) How does CAS and CCS perform supervisory signaling?

Q6) What is the Nyquest Theorem and how does it apply to digitization?

Q7) List the four types of telephone voice quality. Which one is the most desirable, and why?

Q8) What are some of the features of ISDN?

Q9) What are some of the features of SS7?

Answers

Answers to the written questions follow:

Q1) What are the four steps in digitizing analog signals?

Sample

Quantization

Encode

Compression (optional)

Q2) Describe the differences between digital and analog signaling.

Analog: Analog signal require one set of wires (2 or 4) per call

Digital: Digital signals require <= 64 kbps

Higher quality/speed line can carry up to 1.544/2.048 Mbps (T1/E1)

Digital uses time-division multiplexing (TDM) for more efficient transmission

Q3) List the two main types of voice compression algorithms. List at least three attributes of each type of algorithm.

Wave form algorithms, and source algorithms.

Wave form algorithms:

- Sample analog signals 8000 times/second
- Use predictive differential method to reduce bandwidth
- Bandwidth reduction highly impacts voice quality
- Does not take advantage of speech characteristics

Source algorithms:

- Source algorithm coders are called vocoders
- Vocoders take advantage of speech characteristics
- Reduce the bandwidth by sending transmitting linear filter setting
- "Codebook" excitation index. The index is used by receiver to look up a set of excitation values.

Q4) In DS1 framing format, how many frames are there in a superframe? How many frames in an Extended Super Frame (ESF)?

There are 12 frames in a superframe. ESF is made up of 24 frames.

Q5) How do CAS and CCS perform supervisory signaling?

Channel associated signaling (CAS) sends its signal for call setup in the same channel as a voice call. Common channel signaling (CCS) differs from CAS signaling in that there is a separate channel used for call setup.

Q6) What is the Nyquest Theorem and how does it apply to digitization?

Nyquist Theorem: When a signal is sampled at regular intervals (two samples per cycle), at a rate at least twice the highest frequency in the channel, the samples will contain sufficient information to allow an accurate reconstruction of the signal at the receiver. (Note: The highest frequency for a voice is 4000 Hz, thus 8000 samples per second are taken.)

Q7) List the four types of telephone voice quality. Which one is the most desirable, and why?

Toll quality, transparent quality, conversational quality, synthetic quality.

Toll quality is the most desirable; the standard telephone speech signal quality.

Q8) What are some of the features of ISDN?

- Fast call setup over all-digital network
- More bandwidth to the desktop
- More efficient use of access trunks reduces number of access trunks required and trunk holding time
- Potential for combining voice and data accesses reduces number required
- CCS network provides a standard out-of-band signaling
- Standards-based for voice, data, and signaling

Q9) What are some of the features of SS7?

- Fast call setup by high-speed circuit-switched connections.
- PBX transaction capabilities, (i.e., call forwarding, call waiting, call screening, call transfer) extended to the entire network.
- Dedicated control channel for all signaling functions.
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- SS7 is replacing the older common channel interoffice signaling (CCIS).
- 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.
- The channel used for common channel signaling need not be associated with any particular trunk group.

4

Cisco Voice Products

Objectives

Objectives

Upon completion of this chapter, you will be able to perform the following tasks:

- **Given a Cisco 2600 router and other necessary components, set up and verify the hardware connections necessary to transport voice over an IP network**
- **Given a Cisco 3600 router and other necessary components, set up and verify the hardware connections necessary to transport voice over an IP network**
- **Given a Cisco MC3810 router and other necessary components, set up and verify the hardware connections necessary to transport voice over Frame Relay or ATM**
- **Given a Cisco AS5300 router and other necessary components, set up and verify the hardware connections necessary to transport voice over IP**

Chapter Content

This chapter specifically tells you how to set up your router to make a voice call.

The chapter outline follows:

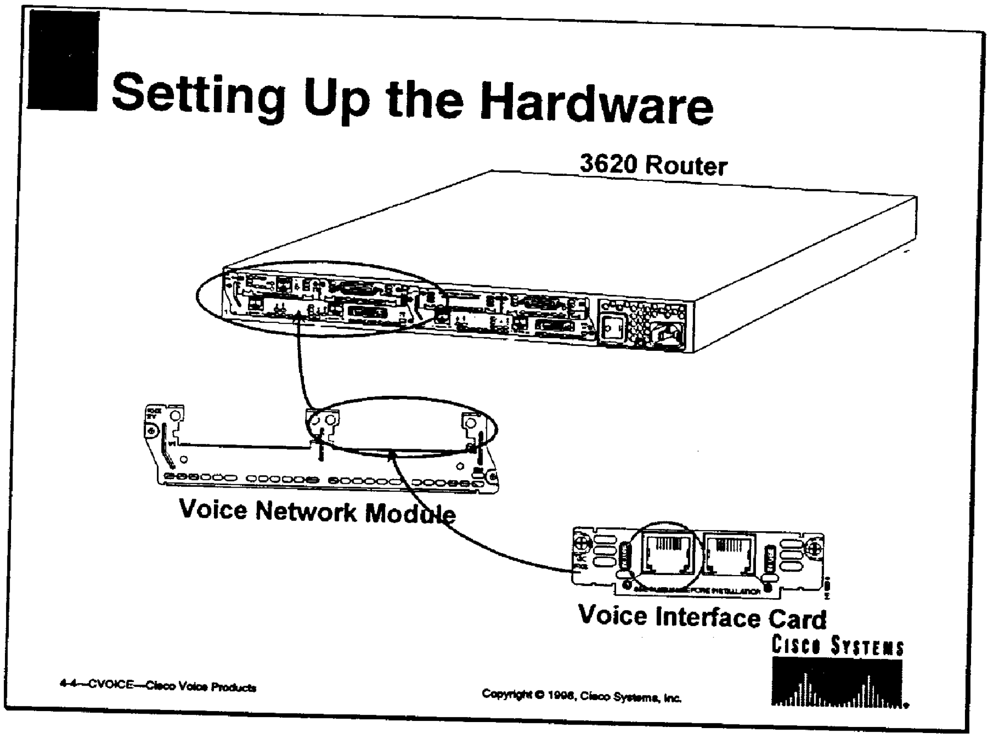
- Introduction
- Setting Up Your 2600 and 3600 Router for Voice over IP
- Setting Up Your MC3810 Router for Voice over Frame Relay or ATM
- Setting Up Your AS5300 Access Server for Voice over IP
- Summary

Introduction

Cisco is the proven leader in developing and transporting time-sensitive data across WANs. With the 2600, 3600, MC3810, or AS5300 family of routers, you can now run telephony over your existing data network.

Cisco voice products offer the following benefits:

- Using the data network for direct voice and data connections within an enterprise
- Reuse of WAN network to transport voice traffic



Setting Up Your 2600 and 3600 Router for Voice over IP (VoIP)

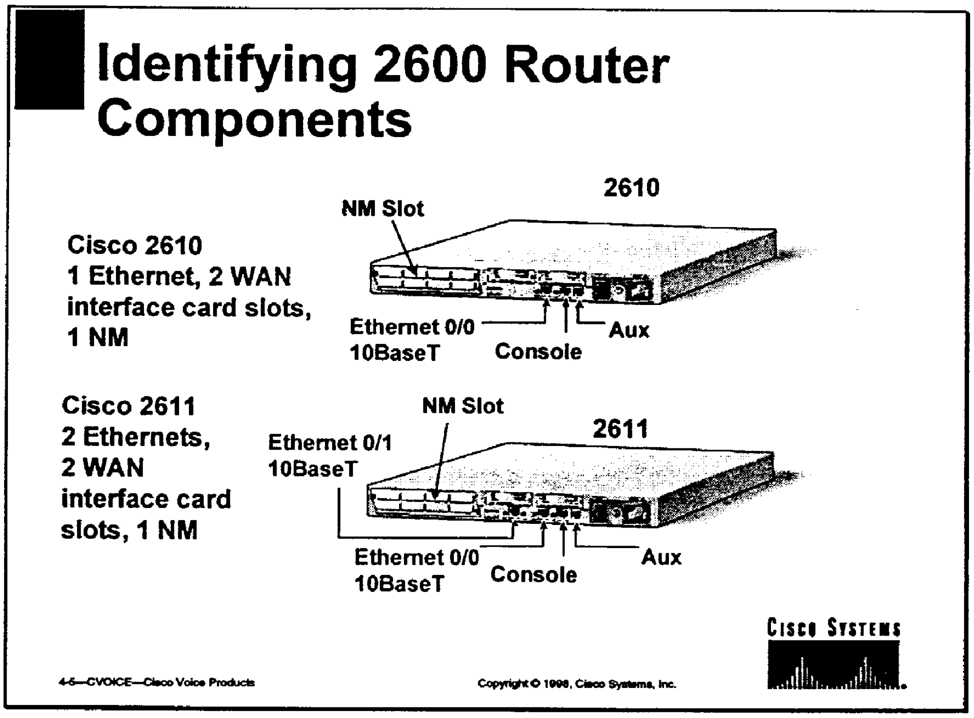
You can use your 2600 or 3600 router to transport your voice calls across an IP network. However, you must install additional hardware components into your router to transport voice messages. This section highlights the necessary components and the necessary installation procedures.

To make VoIP calls you need to install two hardware devices into the router:

- Voice Network Module (VNM)—This device converts and compresses digital voice signals into voice packets that can be transmitted in an IP environment.
- Voice Interface Card (VIC)—This device provides the connection from the telephony device to the router. It also converts analog signals into a digital format.

This section instructs you how to make proper connections. The section topics are:

- Identifying 2600 Router Components
- Identifying 3600 Router Components
- Selecting the Voice Network Module
- Selecting the Voice Interface Card
- Voice Port Addressing on Your VIC
- Router, VNM, and VIC Assembly
- Verifying Hardware Installation
- Installing VNM and VIC Procedure



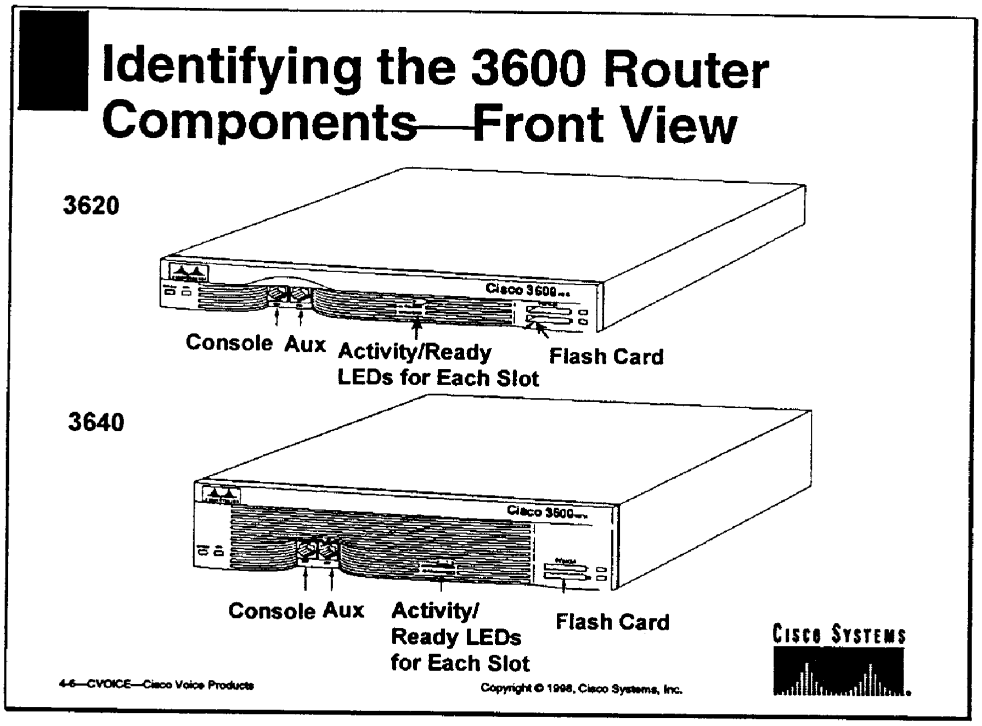
Identifying 2600 Router Components

Within the 2600 family of routers, you may currently select between the 2610 and 2611 family of routers.

The 2610 router includes the following features:

- 40-MHz MPC 860 RISC processor
- One network module slot
- Two WAN interface card slots
- One Ethernet port, 10BaseT only, no AUI
- One internal Flash SIMM slot (maximum 16 MB Flash)
- Two internal DRAM 100-pin DIMM slots (maximum 64-MB DRAM)
- Console and auxiliary ports with maximum speed of 115.2 kbps
- 47-watt AC or DC power supply, or external redundant power supply

The 2611 router includes the same features as the 2610. It also includes one additional Ethernet port.



Identifying 3600 Router Components

The Cisco 3620 router, a member of the Cisco 3600 series of routers, is a two-slot modular access router whose LAN and WAN connections can be configured by means of interchangeable network modules and WAN interface cards. The modular design of the router provides flexibility, allowing you to configure the router to your needs and to reconfigure it if your needs change.

Note The 3640 router works like the 3620 but is larger. It includes four module slots from 0 to 3.

The 3620 router includes the following features:

- 80-MHz IDT RISC processor
- Two network module slots
- Two PCMCIA slots for Flash memory
- Two internal Flash SIMM slots (maximum 16-MB flash)
- Four internal DRAM SIMM slots (maximum 64-MB DRAM)
- Console and auxiliary ports with maximum speed of 115.2 kbps
- 60-watt AC or DC power supply, or external

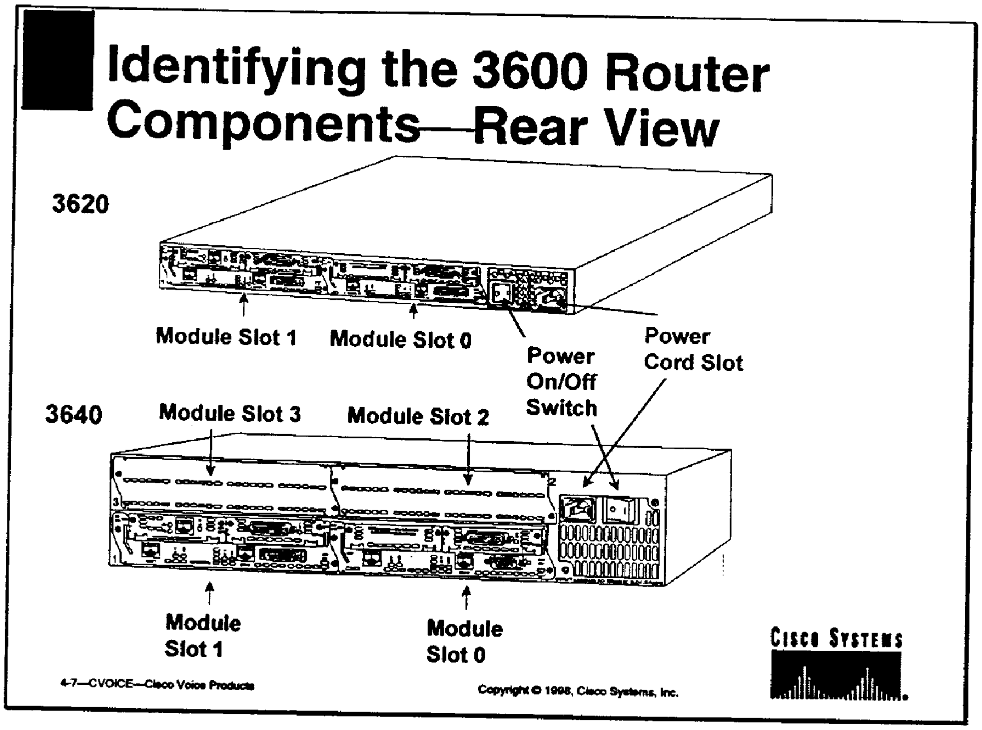
The 3640 router includes the following options:

- 100-MHz IDT RISC processor
- Four network module slots
- Two PCMCIA slots for Flash memory

- Two internal Flash SIMM slots (maximum 16-MB Flash)
- Four internal DRAM SIMM slots (maximum 128-MB DRAM)
- Console and auxiliary ports with maximum speed of 115.2 kbps
- 140-watt AC or DC power supply, or external

The visible features on the front of each router are as follows:

- Console port
- Auxiliary port
- Activity/ready LED lights for each module slot
- PC card slot



The back of the router includes the following:

- Connector for the power cord
- Power on/off switch
- Two chassis slots used to mount network modules if you have a 3620 router
- Four chassis slots used to mount network modules if you have a 3640 router

On the 3620, module slot 0 is on the right, near the power supply, and slot 1 is on the left. These slot numbers correspond to the two sets of LEDs on the front panel of the chassis. The slot number is used as part of the identification of the network interfaces installed in the router. One slot should have a WAN network module installed to provide the connection to the IP WAN. The other will be used for the VNM.

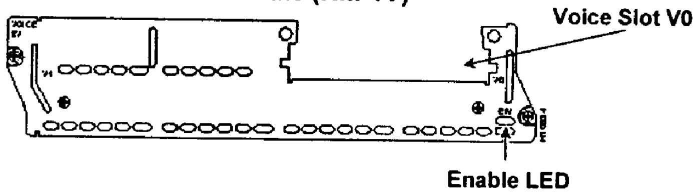
The 3640 routers have two more module slots for a total of four. Facing the router, module slot 0 is on the bottom right, slot 1 is on the bottom left, slot 2 is on the upper right, and slot 3 is on the upper left.

Reference:

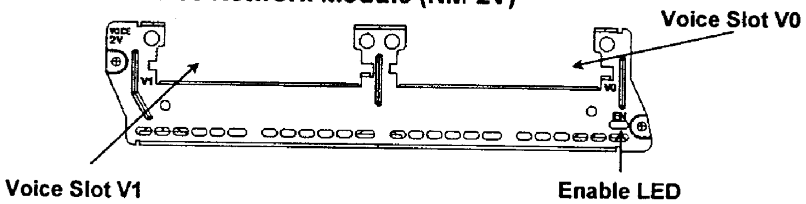
This course does not provide a tutorial on additional, available module interfaces and WAN interface cards. For more information on available options, refer to the *Cisco 3600 Router Installation and Configuration Guide* that came with your router.

Selecting the Voice Network Module

One-Slot Voice Network Module (NM-1V)



Two-Slot Voice Network Module (NM-2V)



CISCO SYSTEMS

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Selecting the Voice Network Module

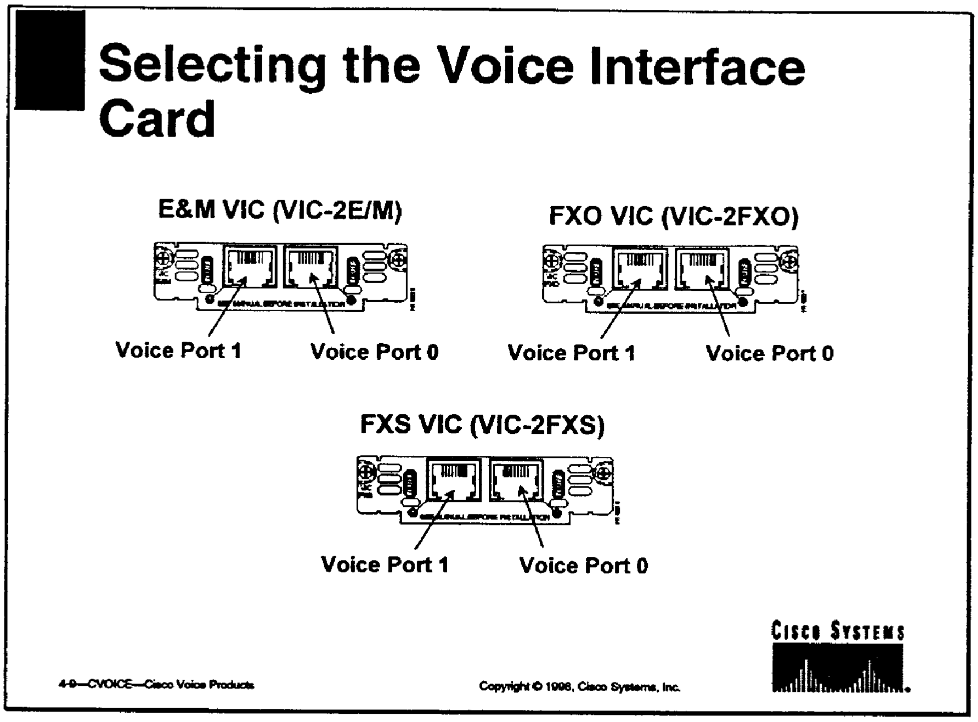
Cisco manufactures two VNMs designed to convert telephone voice signals into a form that can be transmitted over an IP network. You must install at least one of these on your 2600 or 3600 series router to transport voice traffic. The modules are:

- One-slot VNM (Product Number: NM-1V)—You can install one VIC into a one-slot VNM. The single VIC slot is slot 0.
- Two-slot VNM (Product Number: NM-2V)—You can install two VICs into a two-slot VNM. Facing the rear of the router, the VIC slots are slots 0 and 1 from right to left, respectively.

Because each VIC interfaces with up to two telephony devices, a one-slot and two-slot VNM can support a maximum of two and four telephony devices, respectively. Device selection depends upon the amount of telephony devices you wish to connect to the router.

Each VNM has an enable LED that indicates that the module has passed its self-tests and is available to the router.

Caution *Each router comes with panels covering the module slots, called module blank filler panels. For safety and proper ventilation, remove the filler panels only if you install a module into the slot. Keep this slot covered with the blank filler panel when modules are not installed.*



Selecting the Voice Interface Card

In addition to the VNMs, at least one VIC must be installed into the VNM to process analog voice for transport over an IP network. VICs provide the connection to the telephone equipment or telephone network. Three VICs, each providing two voice ports, are available:

- E&M VIC (Product Number: VIC-2E/M)—Used for PBX to PBX tie line connections.
- FXO VIC (Product Number: VIC-2FXO)—Used for connections to central office equipment.
- FXS VIC (Product Number: VIC-2FXS)—Used for direct connections to telephony devices.

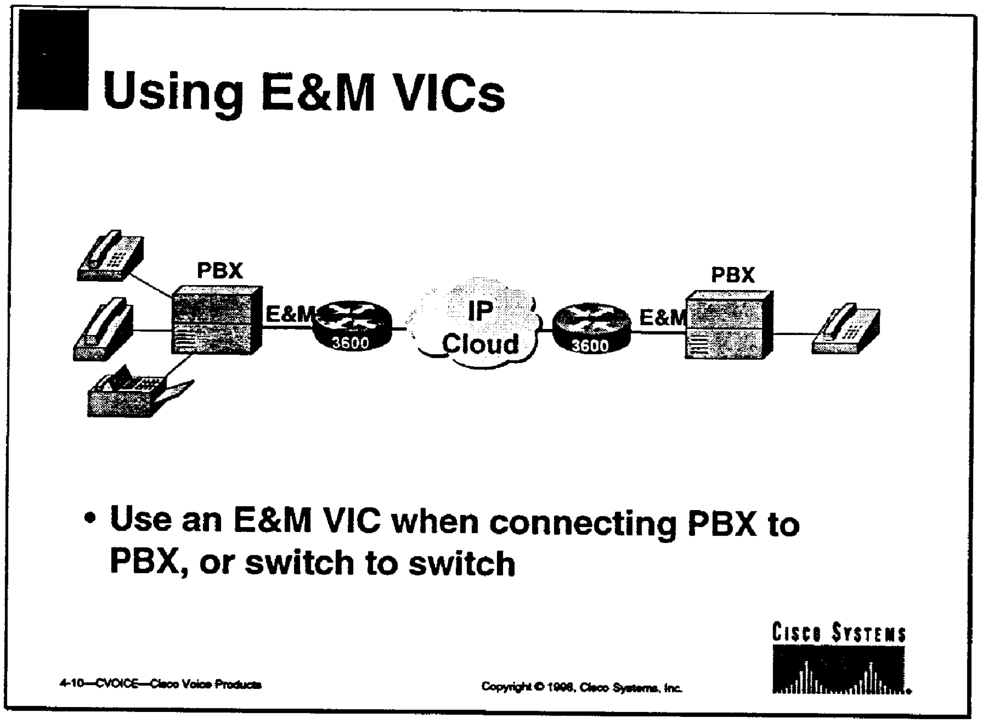
Each VIC includes two voice ports. Each port is an actual RJ style connection in which you plug a telephone, fax, or PBX. Regardless of the VIC you use, the voice ports are called V0 and V1 from right to left, respectively.

Each VIC has an enable LED that corresponds to each voice port. This LED will have an amber flash while the router and VIC are being powered up and initialized. After initialization the light turns off. When your telephony device is properly installed, and you operate it (i.e., pick up the telephone receiver to make a call), the LED light will turn on and turn green to specify that the voice port is in operation. An amber signal designates that the voice port is not in operation.

Caution *Although VICs physically resemble WAN interface cards, which install in a Cisco 3600 series two-slot network module to provide WAN interfaces, VICs and WAN interface cards are not interchangeable. VICs cannot be installed in a two-slot network module, and WAN interface cards cannot be installed in a VNM. For information on available WAN interface cards, refer to the installation and configuration guide that accompanied your router.*

Caution *Although VICs physically resemble WAN interface cards, which install in a Cisco 2600 WAN interface slot to provide WAN interfaces, VICs and WAN interface cards are not interchangeable. VICs cannot be installed in a two-slot network module, and WAN interface cards cannot be installed in a VNM. For information on available WAN interface cards, refer to the installation and configuration guide that accompanied your router.*

Caution *Similar to the router, the VNM comes with panels covering the VIC slots, also called blank filler panels. For safety and proper ventilation, remove the filler panels only if you install a VIC into the slot. Keep this slot covered with the blank filler panel when VICs are not installed.*



Using E&M VICs

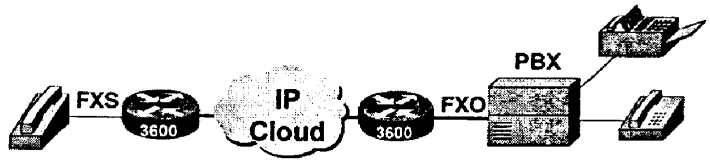
E&M—Earth and Magneto or Ear and Mouth—is a trunking arrangement generally used for:

- Tie lines between telephone switches (PBX)
- Telephone switch-to-network connections for off-premise extensions (OPX)
- Centrex connections between a PBX and a CO

Cisco's E&M interfaces use RJ-48 connectors that allow connections to PBX trunk lines (tie lines). The figure above shows a typical application with E&M interfaces.

Note An RJ-45 connector will plug into an RJ-48 interface as well.

Using FXO VICs



- **Use a FXO VIC when connecting to the station side of a PBX or directly to the CO**
- **FXOs allow for off-premise connections**

Cisco Systems

4-11—CVOICE—Cisco Voice Products

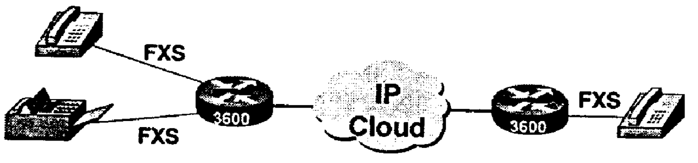
Using FXO VICs

FXO interface connects local calls to:

- The telephone company's central office
- PBX

Cisco's FXO interfaces use RJ-11 connectors, the same interfaces you will see on telephony devices, which connect directly to the PSTN central office or a PBX. This interface is useful for OPX applications. The figure above shows a typical application with an FXO interface.

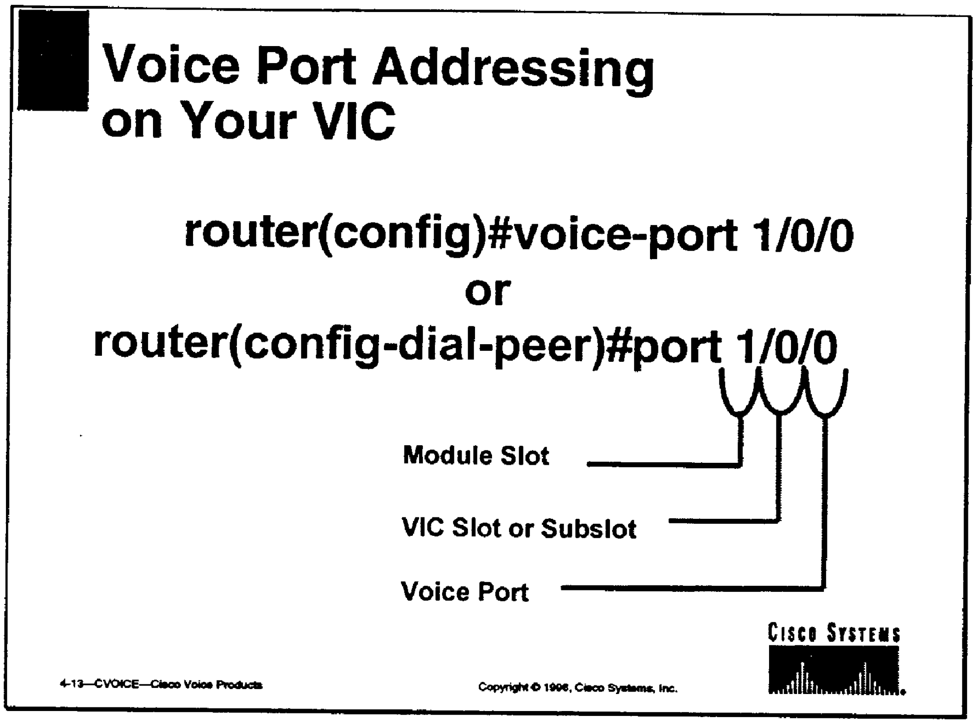
Using FXS VICs



- **Use an FXS VIC when connecting directly to a standard phone or fax machine**

Using FXS VICs

FXS is an interface that connects directly to a standard telephone, fax machine, or similar device and supplies ring 48 voltage, line power, and dial tone. Cisco's FXS interfaces use RJ-11 connectors that allow connections to basic telephone service equipment, key sets, and PBXs. The figure above shows a typical application with an FXS interface.



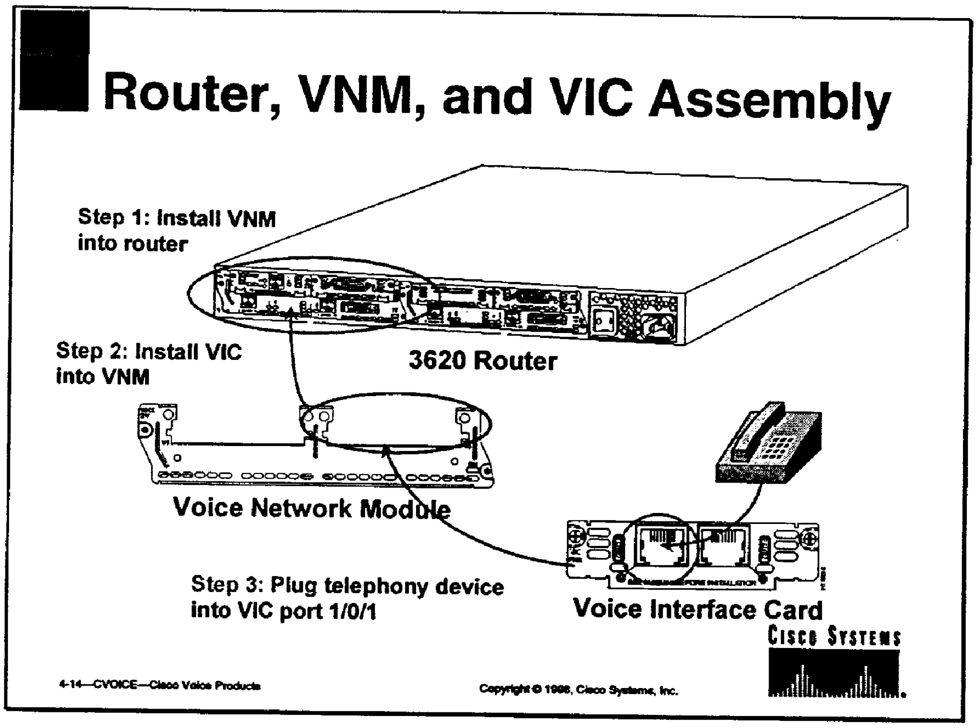
Voice Port Addressing on Your VIC

You must be able to correctly identify each voice port on the VIC. Cisco IOS™ software commands like the **voice-port** and **port** commands identify each port with a three-part specification. Each part is delimited with a forward slash (/):

- First is the module slot on the router.
- Second is the VIC slot, also known as the subslot.
- Third is the actual voice port for the telephony device.

For example, if a telephone was plugged into voice port 1/0/1 on your 2600 or 3600 router, it would be in module slot 1, VIC slot 0, and voice port 1.

The VIC you select depends on the way the network is configured. Each card is used for specific applications. Review the following interface descriptions and applications to determine the VIC(s) that best suit your needs.



Router, VNM, and VIC Assembly

The 2600 or 3600 router, VNM, VIC, and telephony device fit together as depicted in the figure above.

Verifying Hardware Installation

When the router's power switch is in the ON position, check that the LED light corresponding to the VNM slot you installed is on. If the light is on, you most likely installed the VNM and VIC properly. If the light is off, turn off the machine and check that the VNMs and VICs are securely positioned. You may need to remove them and reinstall the hardware.

Note FXS VICs provide dial tone. If you are using an FXS VIC, check for dial tone on the telephony device by lifting the receiver and listening for dial tone.

Installing VNM and VIC Procedure

The following tools and procedure describe how to install the VNM and VIC.

Required Tools and Equipment

You need the following tools and equipment to install a VIC into a Cisco 3600 series router:

- VIC
- VNM
- Number 1 Phillips screwdriver or small flat-blade screwdriver

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Caution *When installing the VNM and VIC you should use an ESD-preventive wrist strap to prevent electrical shock.*

Procedure

Complete the following steps to install the VNMs and VICs:

WARNING *Warning means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. For a complete list of all precautionary warnings, refer to the* Installation and Configuration Guide *and the* Regulatory Compliance and Safety Information *document that accompanied the router.*

Step 1 Turn OFF electrical power to the router. However, to channel ESD voltages to ground, do not unplug the power cable. Remove all network interface cables, including telephone cables, from the rear panel.

Note Your power cord has three prongs that plug into the power source. The round prong is designed to channel the ESD voltages to ground.

WARNING *The following warning applies to routers that use a DC power supply: Before performing any of the following procedures, ensure that power is removed from the DC circuit. To ensure that all power is OFF, locate the circuit breaker on the panel board that services the DC circuit, switch the circuit breaker to the OFF position, and tape the switch handle of the circuit breaker in the OFF position.*

Step 2 Remove the blank filler panel from the module slot where you plan to install the module.

Step 3 Align the VNM with the guides in the chassis and slide it gently into the slot.

Step 4 Push the module into place until you feel its edge connect securely with the connector on the motherboard.

Step 5 Fasten the module's captive mounting screws into the holes in the chassis.

Step 6 Remove the blank filler panel from the slot in the VNM where you plan to install the card.

Step 7 Align the card with the guides in the VNM and slide it gently into the VIC slot.

Step 8 Push the card into place until you feel its edge connect securely with the connector in the VNM.

Step 9 Fasten the card's captive mounting screws into the holes in the VNM faceplate.

Step 10 Install the network interface cables.

Step 11 Install the telephone, fax, or PBX cord into the appropriate router voice port.

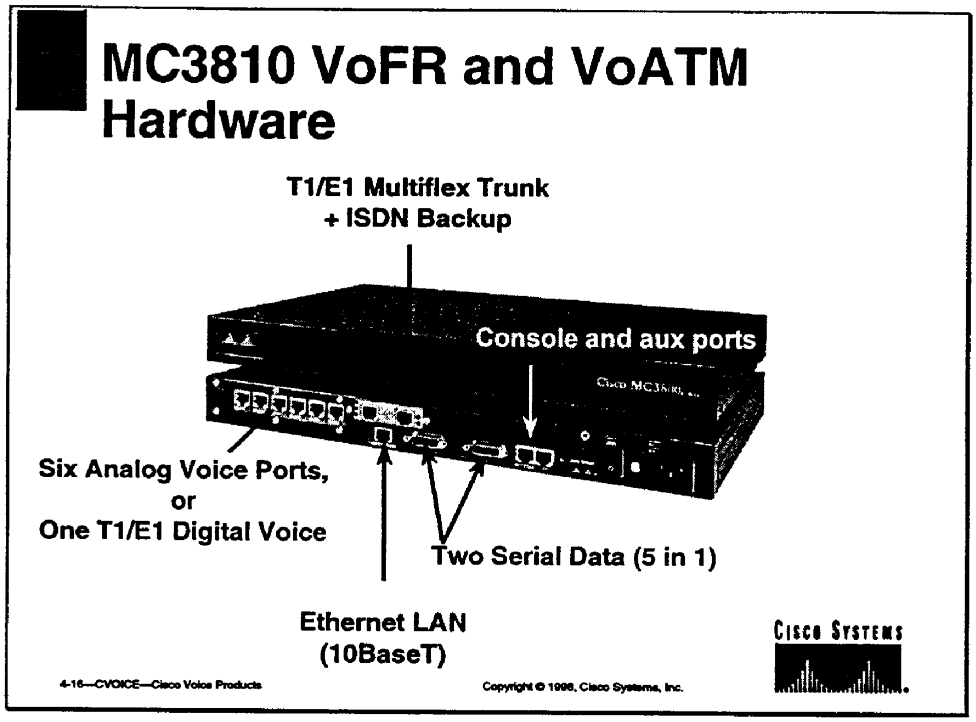
Step 12 Power ON the router.

Step 13 Verify proper installation by checking the LED lights.

WARNING *The following warning applies to routers that use a DC power supply: After wiring the DC power supply, remove the tape from the circuit breaker switch handle and reinstate power by moving the handle of the circuit breaker to the ON position.*

WARNING *This equipment is intended to be grounded. Ensure that the host is connected to earth ground during normal use.*

Setting Up Your MC3810 for VoFR and VoATM Hardware



MC3810 for VoFR and VoATM Hardware

The MC3810 uses an enhanced Cisco 2500 series router to provide Vvoice over ATM (VoATM), Frame Relay (VoFR), Voice over S/HDLC (Cisco encapsulation), as well as data and video over Frame Relay, ATM, or S/HDLC.

Voice traffic is sent to the Multiflex Trunk (MFT) for transport over an ATM WAN trunk or Frame Relay network.

The Cisco MC3810 is supplied with various standard hardware chassis configurations. The chassis are manufactured with different sets of functional modules to provide specific functional capability. Many specific configurations are possible, but they are all variations of the basic categcries listed in the following table.

MC3810 Chassis Options

Functional Modules	Installed	Service Types Supported
Base chassis	None or optional MFT	Data, video
Analog chassis	AVM, VCM6, optional MFT	Data, video, analog voice
Digital chassis	DVM, VCM6 (1 or 2), optional MFT	Data, video, digital voice

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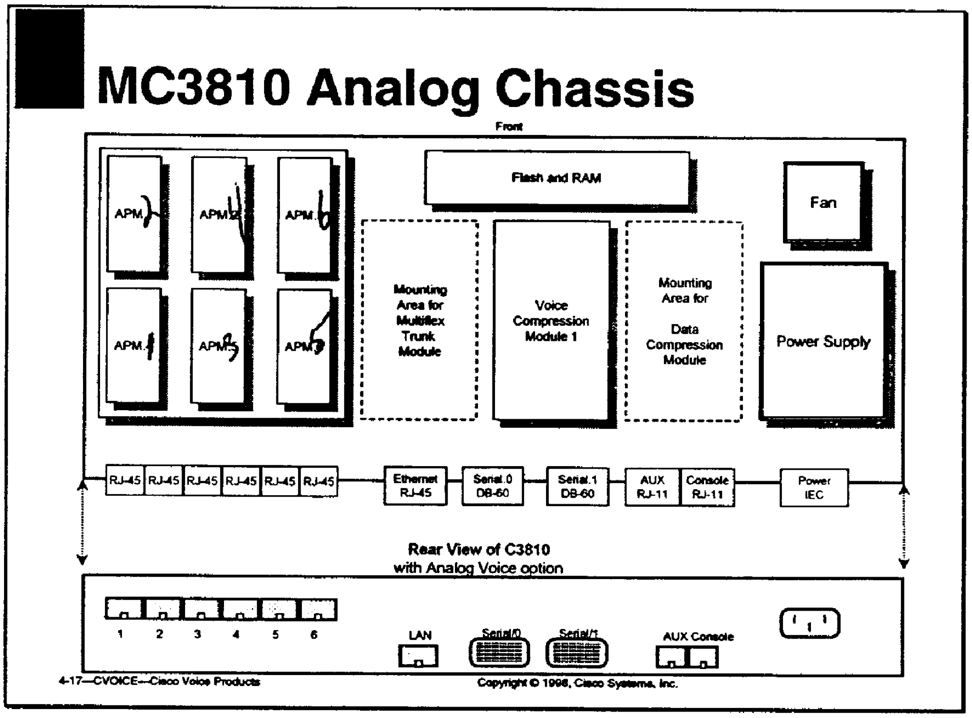
Note Because the base chassis does not support Voice over ATM, it is ***not*** covered in this chapter.



MC3810 Analog Chassis and Related Modules

The analog voice chassis, or system control board, can route Ethernet, S/HDLC, Frame Relay, video data, and compressed voice applications. An analog personality (APM) module is added for a specific application requirement. The optional MFT adds the capability for Nx64 PCM voice, compressed Voice over Frame Relay, and compressed Voice over ATM.

Analog Voice Module

The AVM plugs in to the system control board and provides six analog voice interfaces via RJ-45 connectors. The interfaces may be used with analog telephones, key systems, or PBXs. The AVM provides the plug-in mounting point for up to six APMs. To activate an interface, the user must install the desired style of APM (APM-FXS, APM-FXO, or APM-E&M) on the AVM.

The AVM can be configured for either T1 (Mu-law) or E1 (A-law).

Analog Personality Module

APMs are plug-in daughter cards that provide specific services to a specific port on the AVM. APMs give analog voice its unique "personality" or style of connection type such as:

- Battery
- Dial tone or ringing
- Power (gain or attenuation)
- Signaling (FXS, FXO, E&M)

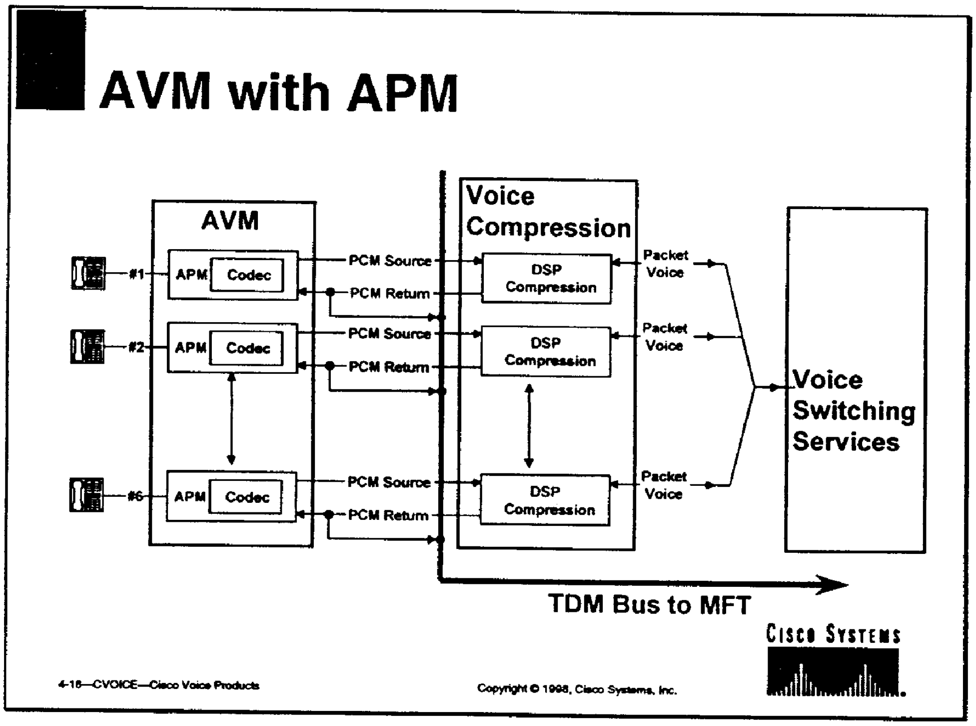
Cisco IOS software is used to configure the functional components on the AVM and VCM6, for a particular FXS, FXO, or E&M module.

Function and Acronyms of the Various APMs

APM signaling modules are mounted on the AVM. They provide interfaces to the user via the RJ-45 jacks on the AVM. The APMs provide the necessary interface circuitry as well as the codec that digitizes the voice into PCM samples.

The acronyms and functions of the various APMs are as follows:

- FXS Personality Module. An APM (APM-FXS) provides connection for a station (i.e., telephone, fax, key system), to services normally provided by a CO or PBX. In addition, FXS cards provide battery, that is, they provide battery to emulate CO-type connections to PBX trunk lines. FXS modules also provide dial tone. FXS points at a station or telephone.
- FXO Personality Module. An APM (APM-FXO) provides a service connection normally coming from a CO to a PBX or from one PBX to another. APM-FXO modules connect to a CO trunk. They provide a CO-type interface connecting to loop start/ground start CO trunks to the station-side of a PBX. FXO cards receive battery. FXO points at a CO or PBX.
- E&M Personality Module. AN APM (APM-E&M) provides service connection normally used for PBX-to-PBX, CO-to-CO, or switch-to-switch tie-line interconnection. As a general rule, E&M (Types I-V) connect to analog line cards on PBXs. E&M refers to the leads used to send and receive on-hook or off-hook status.



The AVM/APM combination provides the following features:

- Six ports of FXS, FXO, or E&M (in any combination)
- Integrated talk battery and ring generator
- Adjustable transmit and receive levels
- 2-wire FXS/FXO voice interface
- 2- and 4-wire E&M interface
- Wink start, immediate start, and delayed start
- Software configurable ground start, loop start, or battery reverse signaling
- Software configurable A-law or Mu-law PCM encoding
- Software configurable impedance (the value is country dependent)

Note If the MFT is used for the network trunk, and ATM is the trunk protocol, then the serial ports may be grouped into logical DS0 groups for AAL1 connections.

A block diagram of the AVM and APM is shown in the previous figure. A voice channel is associated with each APM. Voice signals are digitized into PCM encoded voice samples by the codec on the APM. The codec may be configured for either Mu-law (T1) or A-law (E1) PCM coding. The PCM samples are then passed to the voice compression services in the Cisco MC3810.

Note that the PCM source from the AVM to the compression service does not attach to the TDM bus that connects to the TDM services on the MFT, which means that analog voice ports cannot be directly connected in PCM format to the MFT. However, the PCM return path may be connected to the TDM bus.

Voice Compression Module (VCM)

VCM6 is a stand-off mounted functional module with six each surface-mounted DSPs (three on top, three underneath) that compresses voice traffic after it has been encoded by the AVM's codecs, and multiplexes it in to the ATM (or S/HDLC, or Frame Relay) data stream.

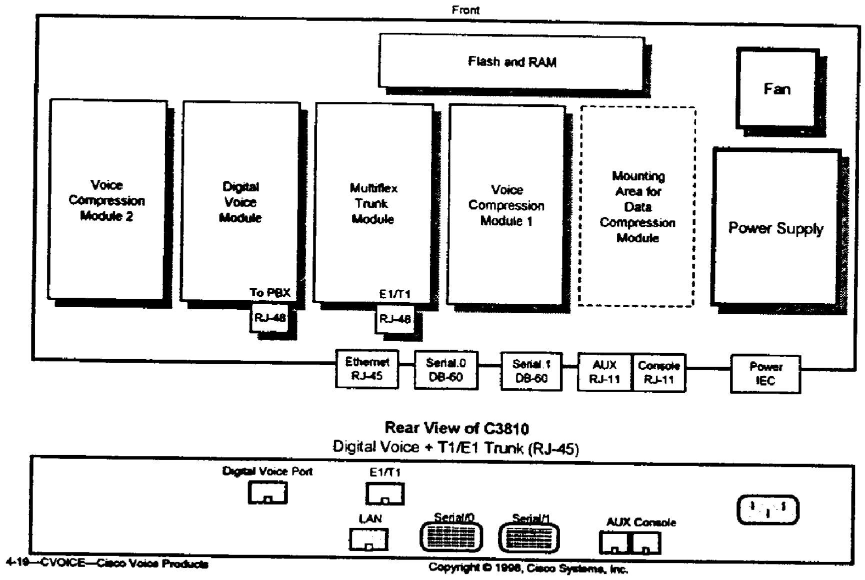
Digital Signal Processor

The DSP's primary function is to compress/decompress the 64-kbps PCM voice traffic into/from either an 8-kbps G.729 or G.729 Annex A CSA-CELP format.

Each DSP can process two G.729a calls, but G.729 or fax traffic requires a full DSP for processing.

Cisco IOS software configures the DSP's coding function on a per-call basis. Codec functionality is globally configured via ROM Monitor.

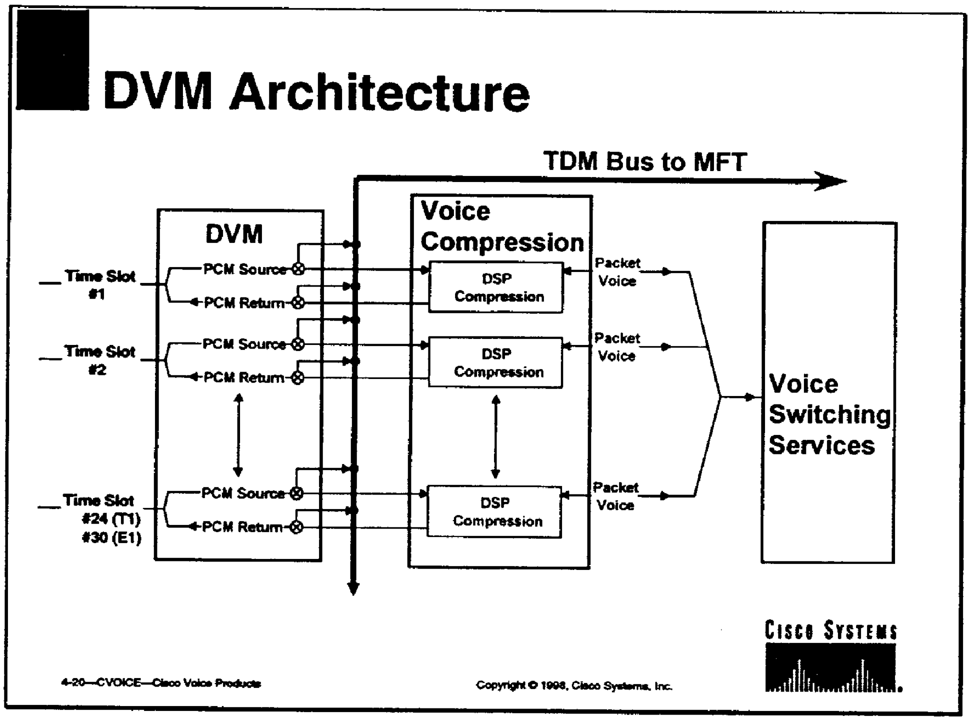
MC3810 Digital Chassis



MC3810 Digital Chassis

The digital chassis provides the user with digital voice interface capabilities. The chassis has slots for a digital voice module (DVM), which enables T1/E1 connections to a digital PBX, compression modules, and an MFT trunk.

The MFT provides for voice and data services over a single WAN interface running at T1/E1 speeds.



Digital Voice Architecture

The DVM provides connectivity to digital PBXs or channel banks on a T1/E1 interface for up to 24 digital phone lines. Signaling types supported on per-channel basis include FXS, FXO, E&M, and CAS (Mercury).

The MC3810 will work with PBXs conforming to the standard specification, EIA/TIA-464-A (February 1989), for PBX Switching Equipment. Channel associated signaling (CAS) is supported on the DVM. Both North American T1 CAS (ABCD bits robbed from each channels) and European CAS (Mercury, ABCD bits from each channel are passed in time slot 16) are supported.

The DVM provides the following features:

- Single T1/E1 (up to 24 digital voice channels)
- CAS
- T1 CAS and European CAS
- FXS, FXO, or E&M
- Wink start, immediate start, and delayed start
- Software configurable ground start, loop start, or battery reverse signaling

Depending on the configuration used, channels from a DVM may be directed to either the voice compression services or the TDM bus. The figure above shows the DVM and how it operates with the TDM bus if the MFT is installed.

Caution *If the MFT is used for the network trunk, and ATM is the trunk protocol, then the DVM channels must be directed to the voice compression services.*

Note It is not possible to configure a MC3810 with both analog and digital voice modules.

Voice Compression Module

DVM requires the use of a voice compression module (VCM), referred to as a V6M6. This compression module is for 6 or 12 voice channels, depending on the voice compression coding algorithm. It supports up to six fax channels. The module has three DSP chips.

Voice compression services are provided by TMS320LC542 fixed-point DSPs from Texas Instruments. Each DSP can run either two channels of G.729a CSA-CELP, or one channel of G.729 CSA-CELP. The G.729a CSA-CELP is less computationally intensive than the G.729 version, and therefore two voice channels may be accommodated on a single DSP.

The Cisco MC3810 is capable of performing either 8 or 16 ms of echo cancellation on a single voice channel.

A maximum of 24 time slots of voice may be configured on the DVM for G.729a CSA-CELP and PCM. A maximum of 12 voice channels of G.729 may be configured on the DVM. The CSA-CELP coders will perform A-law and Mu-law conversion as required.

Fax relay requires the resources of an entire DSP so a maximum of 12 voice channels of fax relay are supported. When fax tones are detected the fax relay function begins automatically. The table below summarizes the maximum number of voice channels under different configuration scenarios.

Maximum Number of Active Voice Channels

Voice Coding	Maximum Voice Channels in T1 Systems	Maximum Voice Channels in E1 Systems
G.711 PCM, 64 kbps	24	30
G.729 CSA-CELP, 8 kbps	12	12
G.729a CSA-CELP, 8 kbps	24	24
Fax Relay	12	12

MC3810 WAN Trunk Options

- **Universal I/O serial**
  - **Frame Relay**
  - **HDLC**
  - **Nx64 kbps to 2 Mbps**
- **Multiflex Trunk (MFT)**
  - **Frame Relay**
  - **ATM over T1/E1**
  - **HDLC**
  - **TDM (voice/video/data)**
  - **Xx64 kbps speeds to E1**

4-21—CVOICE—Cisco Voice Products

Cisco Systems

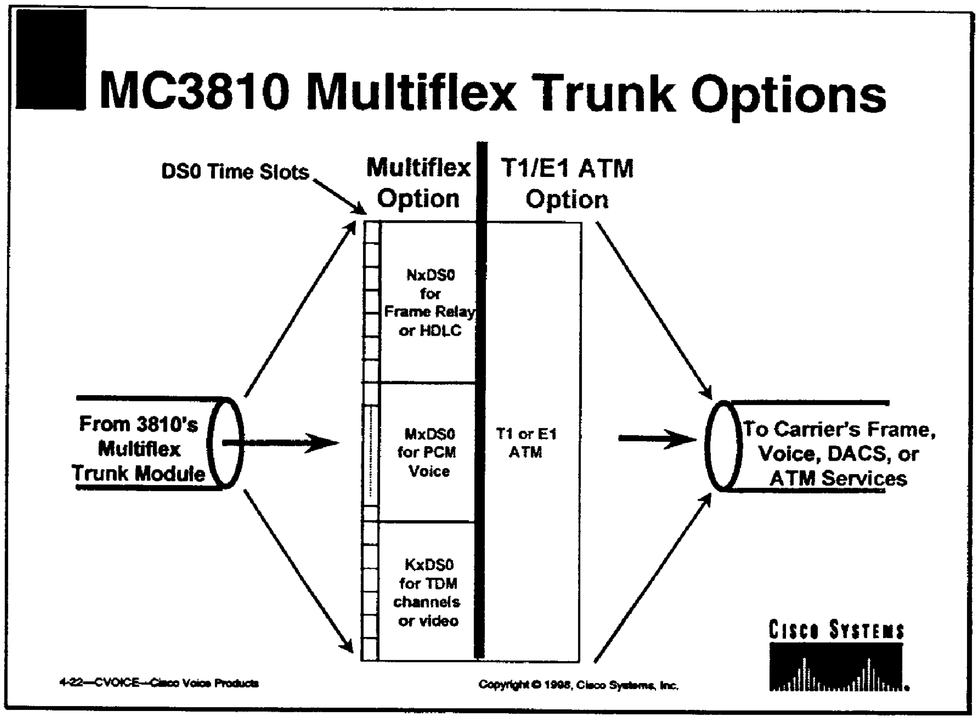
MC3810 WAN Trunk Options

There are two WAN trunk options:

- Two universal I/O serial interfaces
- Multiflex trunk option

The two universal I/O serial interfaces (also known as a 5-in-1 serial port with DB-60) connections can handle voice and data for Frame Relay and HDLC, or run in "transparent" mode.

The MFT has all the features listed, and is discussed in more detail on the next figure.



MC3810 Multiflex Trunk Options

The MFT provides the user with a multiservice, T1/E1 trunk with built-in, long-haul CSU/DSU. The MFT is software configurable to support either ANSI T1.403 (T1) or ITU G.703 (E1). It supports connectivity to ATM, Frame Relay, or leased-line carrier services. To connect to a T1 ATM network, you need to use a T1/E1 trunk interface. The MFT provides an RJ-48 connector jack for the network interface and a T1.403-compliant, onboard CSU/DSU to support a T1 trunk.

The MFT may be ordered with either RJ-48 or BNC connections, though not all versions will be available at first release. The MFT is always a DTE, and it derives network timing (clock) and distributes it to the universal I/O serial (UIO) and DVM.

The MFT has two WAN trunk options, multiflex mode, T1/E1 ATM mode.

The features of the MFT in multiflex mode are as follows:

- Frame Relay, HDLC, PPP, IP

  Compressed voice

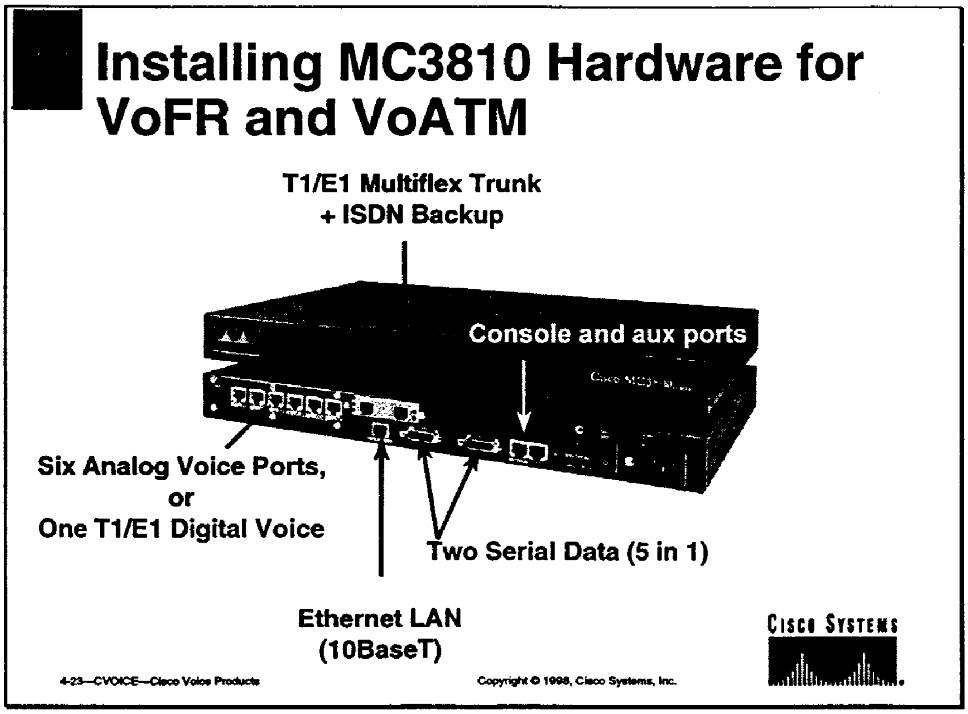
- MxDS0 for PCM voice (TDM)
- DACS channels

  Bit serial data from UIO

  Video or data

The MFT T1/E1 ATM WAN services are as follows:

- Data and video in structured AAL1 format
- Compressed voice or data in AAL5 format



Installing the MC3810 Router

This section provides an overview of the MC3810 hardware setup and interface cabling options for running Voice over Frame Relay or Voice over ATM.

Before you begin installing your MC3810, make sure you have the following equipment handy:

- Screwdriver as required for attaching brackets to rack or wall
- Phillips screwdriver for attaching brackets to MC3810
- Mounting brackets and screws for 24-inch rack, if required
- Interface cables as required
- Wire stripper and torque screwdriver for DC version with internal power supply

In addition, you might need the following external equipment:

- Channel service unit/data service unit (CSU/DSU) for the serial interfaces
- Ethernet hub
- Modem for remote configuration
- Console terminal, or personal computer with terminal emulation software
- Cisco Redundant Power System (RPS) if used

Note Before you can set up the hardware and cables, or configure the Cisco IOS software in the MC3810 to use Voice over ATM, you must first establish a working Frame Relay or ATM network. For more information about configuring Frame Relay or ATM, refer to the *Cisco IOS Wide Area Networking Configuration Guide*. The CVOICE course assumes you have already configured your Frame Relay or ATM backbone network, completed your company's dial plan, and established a working telephony network based on your company's dial plan.

Cabling Requirements for the MC3810

The Cisco MC3810, cables, Documentation CD-ROM, printed publications, and any optional equipment you ordered might be shipped in more than one container. When you unpack each shipping container, check the packing list to ensure that you received all of the following items:

- Cisco MC3810 multiservice concentrator
- Power cord, 6-foot (1.8-meter) (AC models only)
- Console cable (RJ-45 connectors, light blue)
- RJ-45-to-DB-25 female DTE adapter (labeled TERMINAL)
- RJ-45-to-DB-25 male DCE adapter (labeled MODEM)
- RJ-45-to-DB-9 female DTE adapter (labeled TERMINAL)
- Mounting brackets for 19-inch rack (one pair) with screws for attaching to chassis
- Optional equipment (network interface cables, mounting brackets for 24-inch rack, spares)
- Cisco Information Packet (warranty package)
- Documentation CD-ROM and optional printed publications, as specified by your order

Setting Up Your MC3810 Router for VoFR or VoATM

To set up, install, and configure the MC3810 for VoFR or VoATM communications, complete the following steps:

Step 1 Connect the input power (AC or DC and including redundant power system).

Step 2 Connect the MC3810 console port to either an ASCII terminal or a PC running terminal emulation software. (Or, a modem to the auxiliary port, if you plan to configure the MC3810 remotely.)

Step 3 ASCII terminal: connect the light-blue cable from the console port (light blue) to the female DTE adapter.

Step 4 Plug the female DTE adapter into the DB-25 I/O port on the terminal.

Step 5 Set terminal for 9600 baud, 8 data bits, 1 stop bit, no parity, and no flow control.

Step 6 PC: plug the adapter into a DB-9 serial port on the PC.

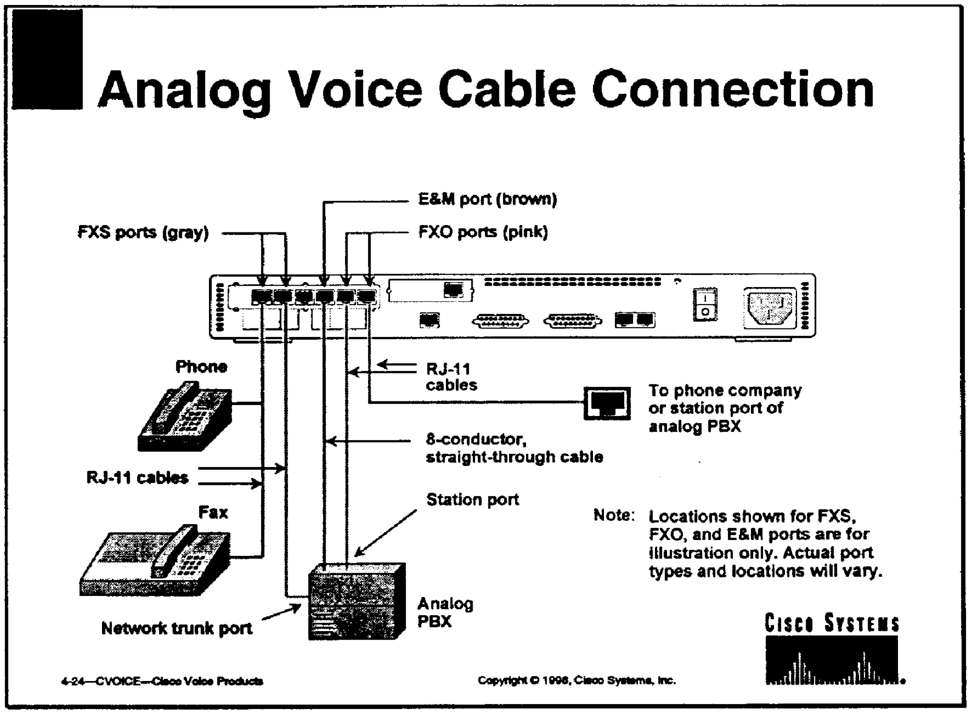
Step 7 Aux port: connect the light-blue cable from the auxiliary port (black) to the male DCE adapter.

Step 8 Plug the male DCE adapter into the DB-25 port on the modem.

Step 9 Configure the modem to match the transmission speed of the auxiliary port (default is 9600 baud), and set the hardware flow control for DCD (Data Carrier Detect) and DTR (Data Terminal Ready) operation.

Step 10 The baud rate for the auxiliary (and console) port can be configured in software for 4800, 1200, 2400, 19200, 38400, 57600, and 115200.

Step 11 Connect the cable from the Ethernet 0 port (yellow) to an available port on the Ethernet hub (if required).

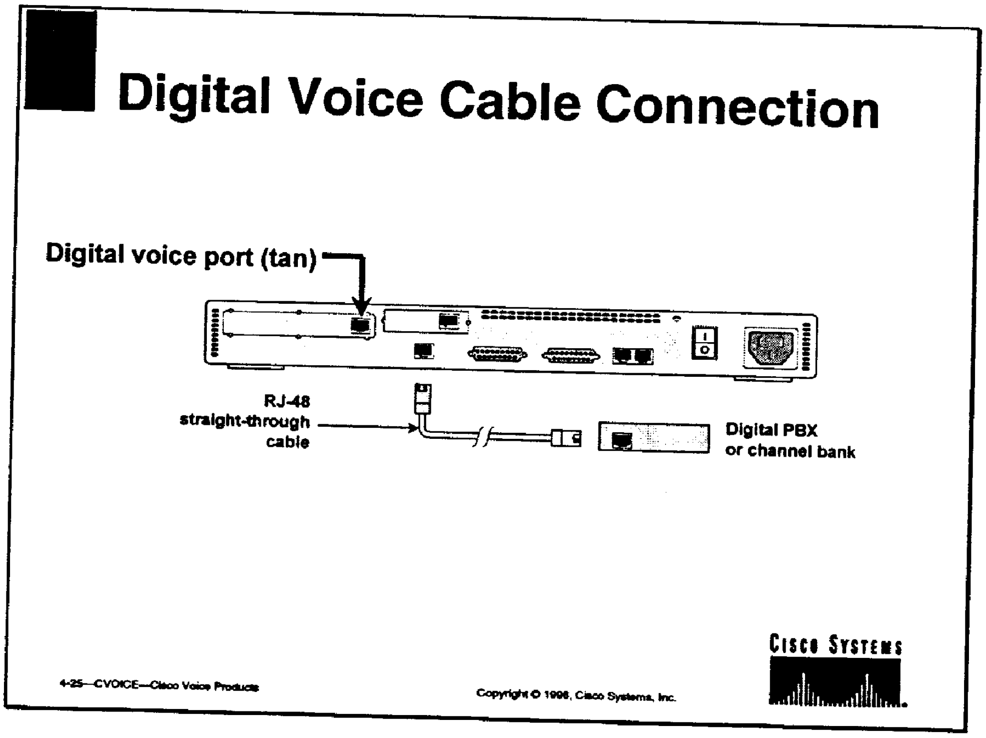


Step 12 If you are making analog voice connections, connect the appropriate cable from an analog voice port to the equipment or line:

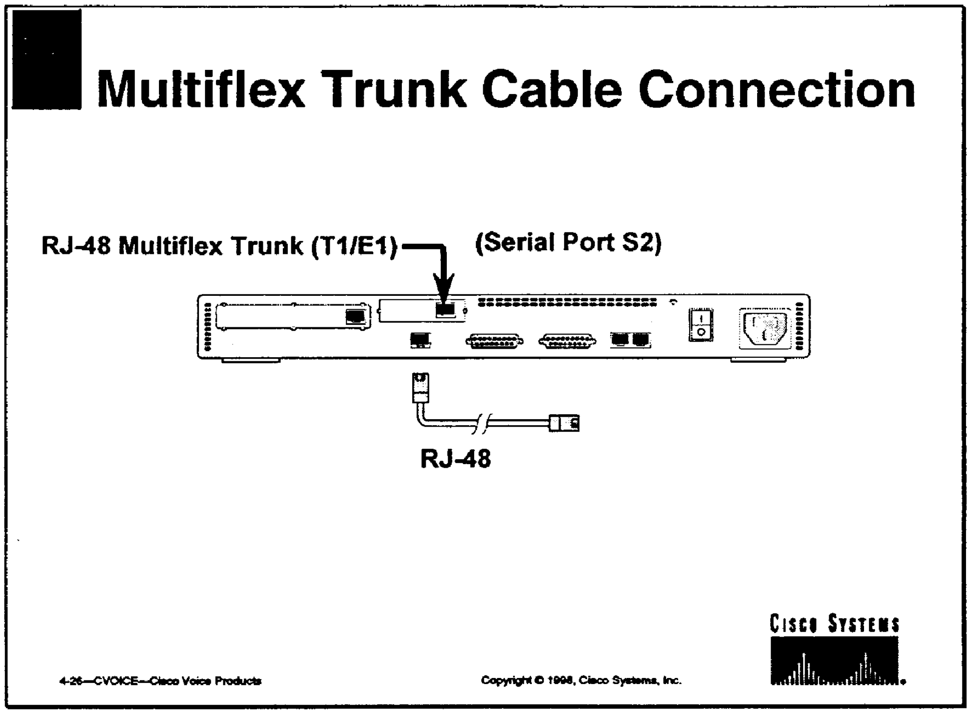
- FXS port (gray) —To telephone or fax equipment or network trunk port of analog PBX
- FXO port (pink) —To central office line or to station port of analog PBX
- E&M port (brown) —To analog PBX

Use the following table to reference the cables you will be installing. The table identifies the proper cable color, connector type, and which port on the MC3810 connects to which interface.

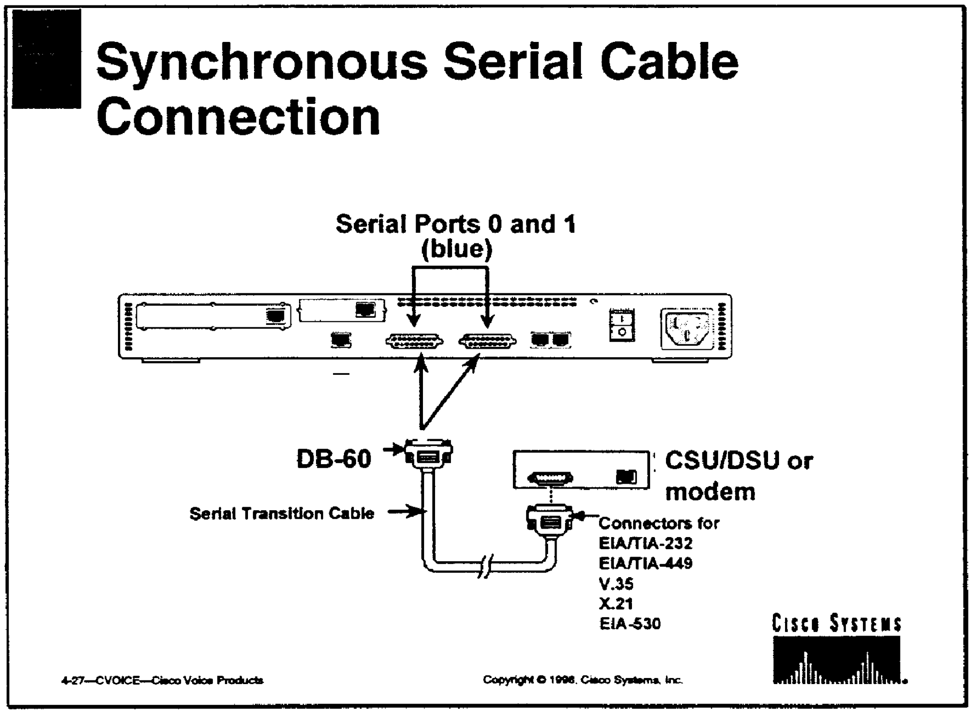
MC3810 Port	Port Color	Connector	Interface to
Console cable	Light blue	RJ-45	Console port
Aux port	Black	RJ-45	Modem
Analog voice—FXS	Gray	RJ-11	Analog phone or fax
Analog voice—FXO	Pink	RJ-11	Telephone central office
Analog voice—E&M	Brown	RJ-1CX	Analog PBX
T1/E1 digital voice	Tan	RJ-48	Digital PBX
T1/E1 trunk—(MFT)	Light green	RJ-48	T1/E1 trunk (telco demarc)
Universal IO serial 2	Dark blue	DB-60	Serial Trunk
Ethernet	Yellow	10BaseT RJ-45	Ethernet hub



Step 13 If you are making digital voice connections, connect the RJ-48 straight-through cable from the MC3810's T1/E1 digital voice port to a PBX or channel bank.



Step 14 To make an MFT cable connection, connect the RJ-48 cable from the T1/E1 trunk port (marked T1/E1 on light green label) to the RJ-48 jack in the network demarcation device (telco demarc).



Step 15 To make a synchronous serial connection, connect the appropriate serial interface cable (DB-60) from the serial port you are using (0 or 1) to the CSU/DSU or modem. Both serial ports are color coded dark blue. The serial interface cable

must match the signaling protocol being used. Connections may be made for EIA/TIA-232, -449, 530, V.35, or X.21.

Verifying the MC3810 Hardware Installation

To verify MC3810 hardware installation, perform the following steps:

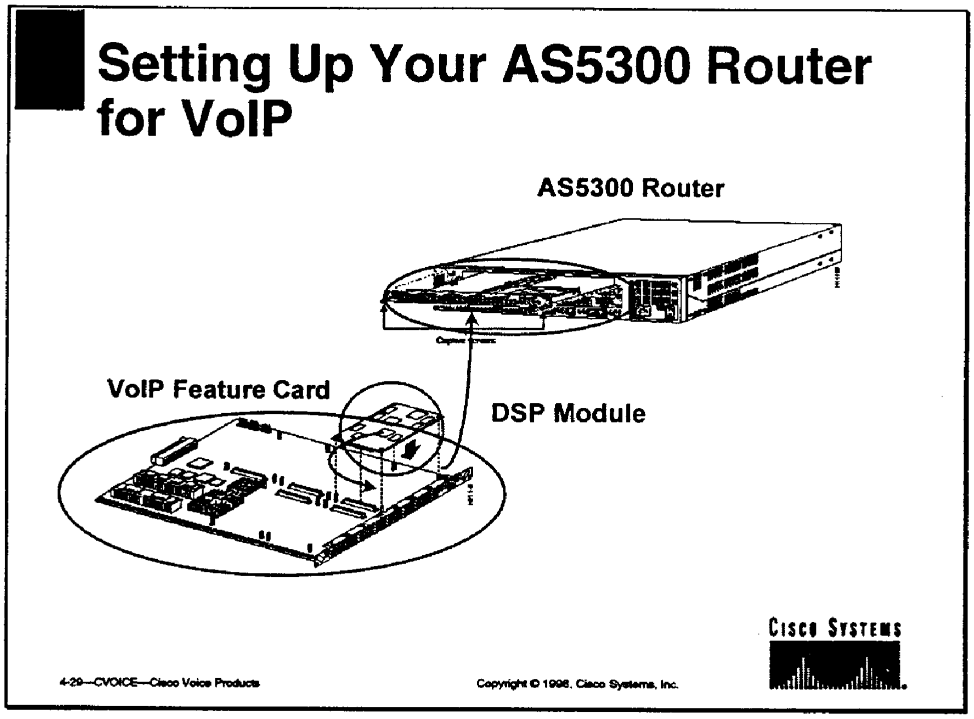
Step 1 Flip the I/O (On-Off) switch to the "O" (On) position so that the router's power LED indicator emits a green color, indicating successful initiation of the MC3810. Make sure all interface cabling LEDs emit a green color, indicating successful connectivity to the network.

Step 2 Using a PC laptop-configured with Telnet or Hyper Term software attached to the MC3810 using the standard rollover console cable, perform basic initial router configuration using Cisco IOS software.

Step 3 Verify and troubleshoot setup, configuration, and connectivity as appropriate to achieve a stable VoFR or VoATM connection using the MC3810. If you have not been able to successfully install the MC3810 hardware and cables, consult the table below for troubleshooting and correcting any problems that may have occurred.

Troubleshooting MC3810 Startup Procedures

Symptom	Possible Cause	Corrective Action
Power LED and fan are off	Power source switched off Faulty power cable Faulty power source Faulty internal power supply	Switch power source on Check/replace power cable Check/correct input power Contact Cisco Technical Assistance Center (TAC) or your Cisco reseller
Power LED on; fan off	Faulty Cisco MC3810	Contact Cisco TAC or your Cisco reseller
Power LED off; fan on	Faulty Cisco MC3810	Contact Cisco TAC or your Cisco reseller
No initialization response from Cisco MC3810	Faulty modem console terminal Faulty cabling to terminal Faulty Cisco MC3810	Check/replace modem/terminal Check/replace cable Contact Cisco TAC or your Cisco reseller
Unit shuts off after operating for some time	Overheating Faulty Cisco MC3810	Check ventilation Contact Cisco TAC or your Cisco reseller
Console screen display freezes	Console fault Software error Faulty Cisco MC3810	Reset/replace console Repeat power-up procedure Contact Cisco TAC or your Cisco reseller



Setting Up Your AS5300 Router for Voice over IP

Your Cisco AS5300 has two primary applications for VoIP:

- The AS5300 provides a central-site telephony termination facility for VoIP traffic.
- The AS5300 provides a telephone company gateway for Internet telephone traffic.

To make VoIP calls you need to install a VoIP feature card with DSP modules into the router as follows:

- VoIP feature card (AS53-CC-VOX)—A voice processing card that resides in one of the slots in the Cisco AS5300 universal access server.

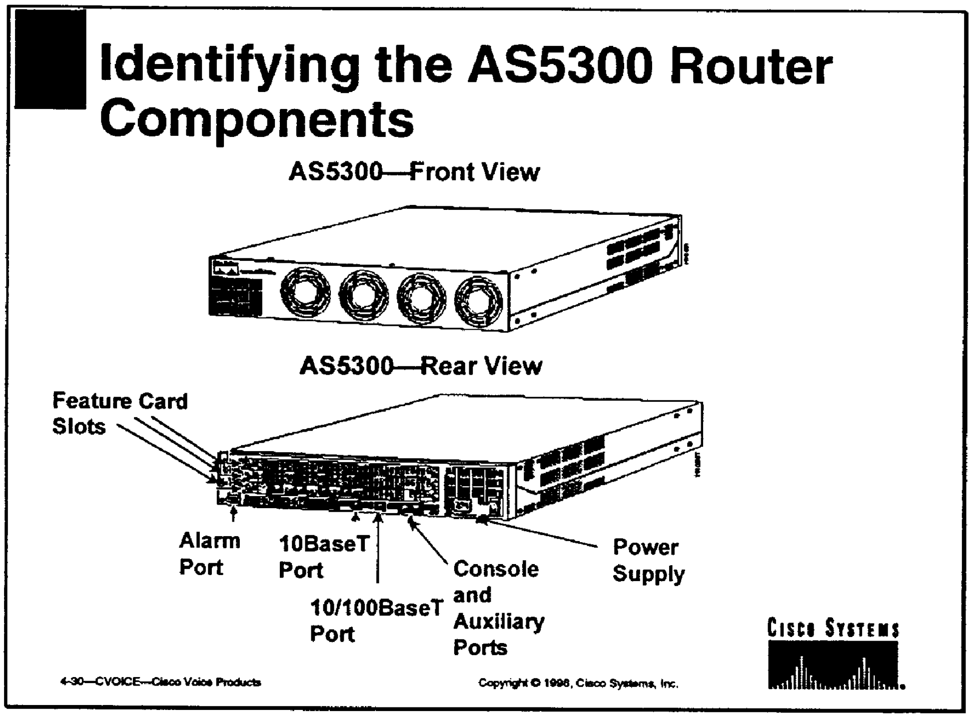
Note Up to five DSP modules (DSPM) can be installed onto the VoIP feature card to perform voice processing for up to 30 ISDN B channels.

- DSP module (AS53-6VOX)—A DSP module provides voice compression and packetization services to the VoIP feature card.

This section instructs you how to make the proper hardware connections. The section topics are:

- Identifying the AS5300 Router Components
- Selecting the VoIP Feature Card
- Selecting the DSP Module
- Router, Feature Card, and DSP Module Assembly
- Verifying Hardware Installation
- Installing the Feature Card and DSP Procedure

Reference: This section only instructs you on the additional components you must install to transport VoIP. For information on installing other feature cards, including the quad TI/Primary Rate Interface (PRI) or quad E1/PRI feature cards, refer to the *Installation and Configuration Guide* that accompanied your router.



Identifying the AS5300 Router Components

The AS5300 router includes the following features:

- One 19-inch modular chassis with a high-speed backplane and three slots for a variety of feature cards

---

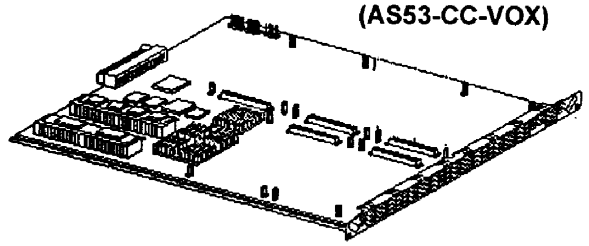
Caution *If the access server is configured with fewer than three feature cards, make sure that a blank slot cover is installed over each open slot to ensure proper airflow inside the chassis.*

Note One slot is used for the quad T1/PRI or quad E1/PRI feature cards. The remaining two slots are available for the VoIP feature card.

- Quad T1/PRI or quad E1/PRI feature cards
- Two Ethernet LAN ports: 10BaseT and 10/100BaseT selectable
- One console port for local administrative access
- One auxiliary port for remote administrative access
- An integral AC or DC power supply

Selecting the VoIP Feature Card

VoIP Feature Card
(AS53-CC-VOX)



- **Used for voice processing**
- **Up to two VoIP feature cards can be installed on a AS5300**

4-31—CVOICE—Cisco Voice Products

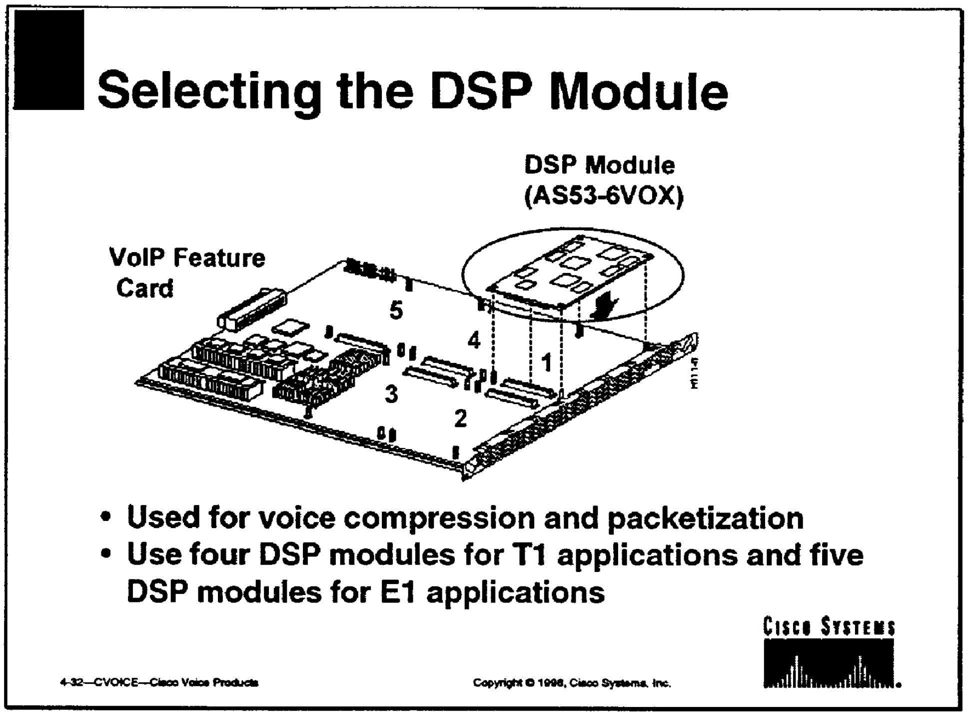
Selecting the VoIP Feature Card

Each AS5300 can accept two voice/fax feature cards and can scale up to 48 or 60 voice connections within a single chassis if you are using a T1 or E1, respectively. The amount you choose depends on the number of connections you desire.

Note T1s are used in North America and E1s are used elsewhere.

The VoIP feature card is a voice processing card that resides in one of the slots in the Cisco AS5300 universal access server. The features of the card are as follows:

- CPU: 4700 mips, 100 MHz
- Support chipset: GT-64010 system controller
- DRAM: Standard 72-pin SIMM (4, 8, 16 MB)
- Flash: Cisco proprietary Flash 80-pin SIMM
- Five DSP module sockets



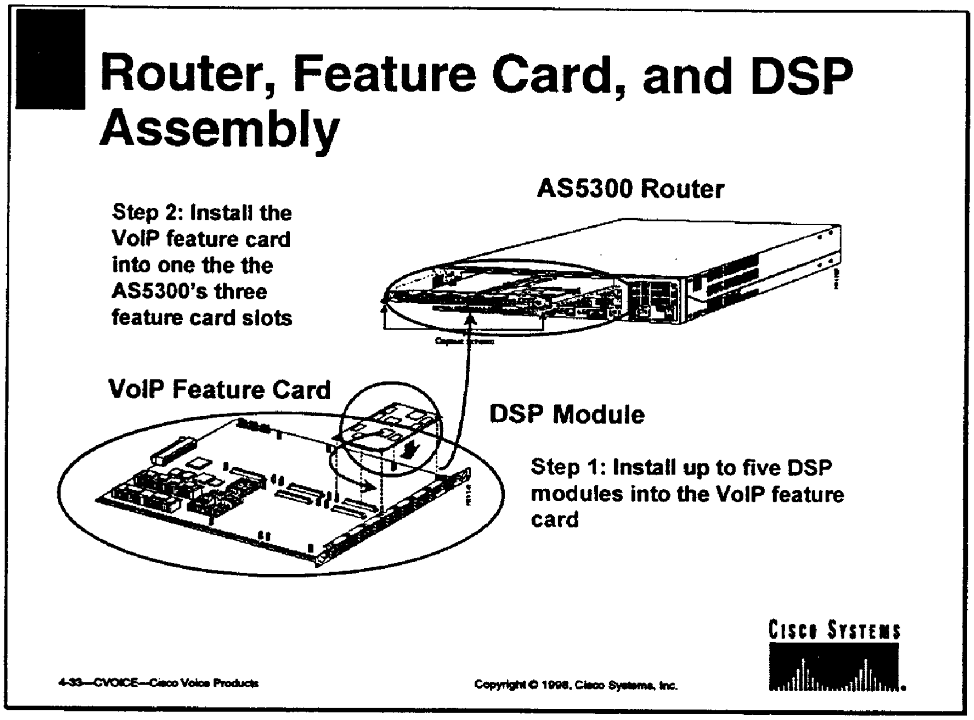
Selecting the DSP Module

The DSP module provides voice compression and packetization services to the VoIP feature card in a configurable and expandable fashion.

The features of the DSP module are as follows:

- Six DSPs per DSP module
- Uses TI TMS320C542 50 MHz
- DSP SRAM: 120 Kwords (16 bit)

Note If you are running T1 applications, use 4 DSP modules for a total of 24 DSPs. If you are running E1 applications, use 5 DSPs for a total of 30 DSPs.



Router, Feature Card, and DSP Module Assembly

The AS5300 router, VoIP feature card, and DSP modules fit as depicted in the figure above.

Verifying Hardware Installation

The LEDs on the front panel of the VoIP feature card indicate the current operating condition of the DSP modules installed on the card. You can observe the LEDs, note any fault condition, and then contact your system administrator or a customer service representative, if necessary.

If the activity LED (ACT) is flickering, there is call activity on the DSP modules. If there is no call activity on the DSP module, the ACT LED will be off.

Note Individual DSP modules do not include LEDs.

When you first power up your AS5300 router, the board OK LED (OK) will flash. When your VoIP feature card has passed initial power-up diagnostics tests and is operating normally, the OK LED light remains on. If the OK LED is off, a fault condition has occurred.

If the LED lights indicate a problem, power off the router and verify that you properly installed each part correctly.

Caution *The VoIP feature cards are not hot-swappable (that is, you cannot remove or install them when the power to the access server is ON). Be sure to turn OFF the power to the access server before installing or removing carrier cards. Failure to do so can damage the access server. Comply with all warnings and cautions in the* Installation and Configuration Guide *that accompanied your router.*

Installing the Feature Card and DSP Procedure

The following procedure describes how to install the VoIP feature card and DSP module.

Required Tools and Equipment

You need the following tools and equipment to install a VoIP feature card and DSP module into a Cisco AS5300 router:

- Cisco AS5300 universal access server
- Feature card removal tool
- Medium-sized Phillips screwdriver
- ESD-preventive wrist strap
- ESD-preventive mat

WARNING *Warning means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents. For a complete list of all precautionary warnings, refer to the* Installation and Configuration Guide *and the* Regulatory Compliance and Safety Information *document that accompanied the router.*

Step 1 Turn OFF electrical power to the access server.

Note Your power cord has three prongs that plug into the power source. The round prong is designed to channel the ESD voltages to ground.

WARNING *The following warning applies to routers that use a DC power supply: Before performing any of the following procedures, ensure that power is removed from the DC circuit. To ensure that all power is OFF, locate the circuit breaker on the panel board that services the DC circuit, switch the circuit breaker to the OFF position, and tape the switch handle of the circuit breaker in the OFF position.*

Step 2 To install the DSP module, mate the socket on the DSP module with one of the five DSP module sockets.

Step 3 Press all four corners onto their respective standoffs.

Step 4 Repeat steps 2 and 3 for each DSP module.

Step 5 To install the VoIP feature card into the access server, figure the VoIP feature card into one of the server's feature card slots until the VoIP feature card touches the backplane connector.

Step 6 Align the captive screws with their holes, and then seat the VoIP feature card completely.

Step 7 Tighten the two captive screws to secure the VoIP feature card to the chassis.

Caution *If the access server is configured with fewer than three feature cards, make sure that a blank slot cover is installed over each open slot to ensure proper airflow inside the chassis.*

Summary

Summary

In this chapter you learned how to complete the following tasks:

- **Set up a Cisco 2600 and verify the hardware connections necessary to transport voice over an IP network**
- **Set up a Cisco 3600 and verify the hardware connections necessary to transport voice over an IP network**
- **Set up a Cisco MC3810 and verify the hardware connections necessary to transport voice over Frame Relay or ATM**
- **Set up a Cisco AS5300 and verify the hardware connections necessary to transport voice over IP**

CISCO SYSTEMS

4-34—CVOICE—Cisco Voice Products

5

Configuring Voice Ports and Dial Peers for Voice

Objectives

Objectives

Upon completion of this chapter, you will be able to perform the following tasks:

- Given an IP, Frame Relay, or ATM network, configure and verify the voice ports so you properly interface to the attached telephony devices
- Given a Cisco router capable of transporting voice, configure and verify POTS dial peers so you can place calls to others connected to your router
- Given a VoIP, VoFR, or VoATM network, configure and verify VoIP, VoFR, or VoATM dial peers so you can place calls to others connected to your router

5-2—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Cisco Systems

Chapter Content

This chapter specifically tells you how to configure the voice ports and dial peers on your 2600, 3600, and AS5300 for VoIP or the 3800 for VoFR or VoATM.

The chapter outline follows:

- Introduction
- Configuring and Verifying Voice Ports
- Configuring and Verifying Dial Peers
- Lab Exercise
- Summary

Introduction

In order to place an end-to-end call, you must configure your voice ports accurately so that proper signaling occurs between the telephony device and the router. You must also configure you router to establish logical voice connections with dial peers. These two topics will be covered in this chapter.

Configuring and Verifying Voice Ports

The voice ports provide the physical connection between your telephony device and the router. You must select and properly configure the voice ports that suit your needs.

This section instructs you on the following:

- Configuring Voice Ports
- Basic Voice Port Settings
- Configuring FXO Voice Ports
- Configuring FXS Voice Ports
- Configuring E&M Voice Ports
- General Voice Port Tuning Options
- Verifying and Troubleshooting Voice Ports
- Configuring and Verifying Voice Port Procedure

The Cisco 2600 3600 and MC3810 products support three different kinds of analog voice ports that were discussed in Chapter 4:

- E&M
- FXO
- FXS

The Cisco IOS™configuration commands you specify for each analog voice port will depend largely on how your network is configured. However, under most circumstances, the default **voice-port** command values are adequate to configure FXO and FXS ports to transport voice data over your existing IP network. Because of the inherent complexities involved with PBX networks, E&M ports might need specific voice port values configured, depending on the specifications of the devices in your telephony network.

The MC3810 and AS5300 support digital applications.

Reference: This section lists many of the voice-port configuration commands available for voice capable routers. Commands and command features vary slightly from application to application and router to router. For information specific to your router and application, reference the *Command Reference Guide* applicable to your product.

Configuring Voice Ports

Router(config)#

voice-port *slot-number/subunit-number/port*

- **Enters voice port configuration mode on your 2600 or 3600 router**

Router(config)#

voice-port *slot/port*

- **Enters voice port configuration mode on your MC3810 router**

Router(config)#

voice-port *controller number*:**D**

- **Enters voice port configuration mode on your AS5300 access server**

CISCO SYSTEMS

5-6—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Configuring Voice Ports

Use the **voice-port** global configuration command to enter the voice port configuration mode and configure a specific voice port. The **voice-port** configuration commands are nested so that all subsequent commands affect only the specified voice port unless otherwise noted.

The following table describes port specifications:

voice-port Command for the 2600 or 3600 Router

slot-number/	Specifies the slot number in the Cisco router where the VNM is installed.
subunit-number/	Specifies the subunit on the VNM where the VIC is installed.
port	Specifies the voice port.

Reference: Reference the "Voice Port Addressing on Your VIC" section in Chapter 4 for an understanding of proper addressing.

To access the voice-port configuration mode for port 0, located on subunit 0 on a ,VIC in module slot 1, for example, enter **voice-port 1/0/0**.

The slot numbering on the MC3810 follows:

voice-port Command for the MC3810 Router

slot/	Specifies the slot number in the Cisco router.
Port	Specifies the voice port.

Note The slot number for voice ports on the Cisco MC3810 is always 1. Valid analog port numbers are 1-6 and valid digital numbers are 1-24 for a T1 application and 1-15 and 17-31 on an E1 application. There is no port 0 for voice ports.

The syntax to configure the voice ports on the AS5300 follows:

voice-port Command for the AS5300

controller number:D	Specifies the T1 or E1 controller and D channel associated with ISDN PRI.

Many **voice-port** configuration command options exist once in the voice-port configuration mode. The following sections describe the commands.

The **voice-port** command options are broken down in the following format:

- Basic Voice Port Settings
- Configuring FXO Voice Ports
- Configuring FXS Voice Ports
- Configuring E&M Voice Ports
- General Voice Port Tuning Options
- Verifying and Troubleshooting Voice Ports

Basic Voice Port Settings

Router(config-voiceport)#

codec { g729r8 | g729ar8 }

- **Sets the codec compression mode**

Router(config-voiceport)#

connection { plar | tie-line } *string*

- **Specifies a connection mode for your voice port**

Router(config-voiceport)#

dial-type { dtmf | pulse }

- **Specifies the address signaling for out-dialing**

No pulse Dial on 3810 & 5300

Cisco Systems

5-6—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Basic Voice Port Settings

You may configure the following basic setting on your voice port.

The **codec {g729r8 | g729ar8}** command configures the voice-port compression mode on the MC3810. The **g729ar8** value is the default and is recommended.

Note When configuring your 2600, 3600, or AS5300 router for VoIP, you will configure the codec on the dial peer. See Chapter 8 for correct VoIP codec configuration.

The g729ar8 compression mode can support a maximum of 24 simultaneously active on-net voice calls, while the g729r8 value can only support a maximum of 12. Both compression modes have a nominal data rate of 8 kbps.

The **connection {plar | tie-line}** *string* command allows you to specify a connection mode for a specified voice port. The following table describes the syntax.

plar	Specifies a Private Line Automatic Ringdown (PLAR) connection. PLAR is handled by associating a peer directly with an interface; when an interface goes off-hook, the peer is used to set up the second call leg and conference them together without the caller needing to dial any digits.
tie-line	Specifies a tie-line connection to a PBX. (Only available on the MC3810.)
string	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.

The **dial-type {dtmf | pulse}** command specifies the type of out-dialing for voice port interfaces. The **pulse** option specifies a pulse dialer, while **dtmf** specifies a touch-tone dialer.

Note Only the **dtmf** option is available on the MC3810. The command is not available on the AS530 because it only allows DTMF.

Basic Voice Port Settings (cont.)

Router(config-voiceport)#

cptone *country*

- **Sets a voice call progress tone locale**

Router(config-voiceport)#

description *string*

- **Describes what your voice port connects to**

Router(config-voiceport)#

no shutdown

- **Activates your voice port**

CISCO SYSTEMS

5-7—CVOICE—Configuring Voice Ports and Dial Peers for Voice

The **cptone** *country* command sets the call progress tone. Using this command, configure the voice port for the local territory's call progress tone setting. The call progress tone setting determines the settings for dial tone, busy tone, and ring-back tone. The call progress tone is different from the DTMF tone.

Reference: The default for this command is northamerica. For a list of supported countries, refer to the *Command Reference Guide* specific to your router.

The **description** *string* command allows you to include a string description of that to which the voice port is connected. *string* is a character string describing port connections.

After you have set the parameters for a voice port, you will need to activate the voice port with the **no shutdown** command.

Note If you do not use a voice port, shut it down with the **shutdown** command.

Configuring FXO Voice Ports

Router(config-voiceport)#

signal { loop-start | ground-start }

- **Sets the signaling type**

Router(config-voiceport)#

impedance { 600r | 600c | 900r | 900c | complex1 | complex2 }

- **Specifies the terminating impedance on an FXO port**

Router(config-voiceport)#

ring number *number*

- **Specifies the number of rings before closing a connection**

CISCO SYSTEMS

5-9—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Configuring FXO Voice Ports

When you have an analog voice port you may choose to use an FXO interface. The default **voice-port** commands work most of the time.

Note This section is not applicable on the AS5300 access server.

However, you may wish to confirm the type of signaling you should use by consulting your telephone vendor. Once you know the type of signaling to use, you can specify the proper signaling with the **signal** command. Signal command options for FXO FXS interfaces follow.

signal Command

loop-start	Specifies loop start signaling. Used for FXO and FXS interfaces. With loop start signaling, only one side of a connection can hang up. This is the default setting for FXO and FXS voice ports. You should find out from your telecommunications vendor if you are using loop start or ground start.
ground-start	Specifies ground start signaling. Used for FXO and FXS interfaces. Ground start allows both sides of a connection to place a call and to hang up.

Note Configuring the **signal** command for an FXO voice port will change the signal value for both voice ports on a Cisco 2600 or 3600 VIC.

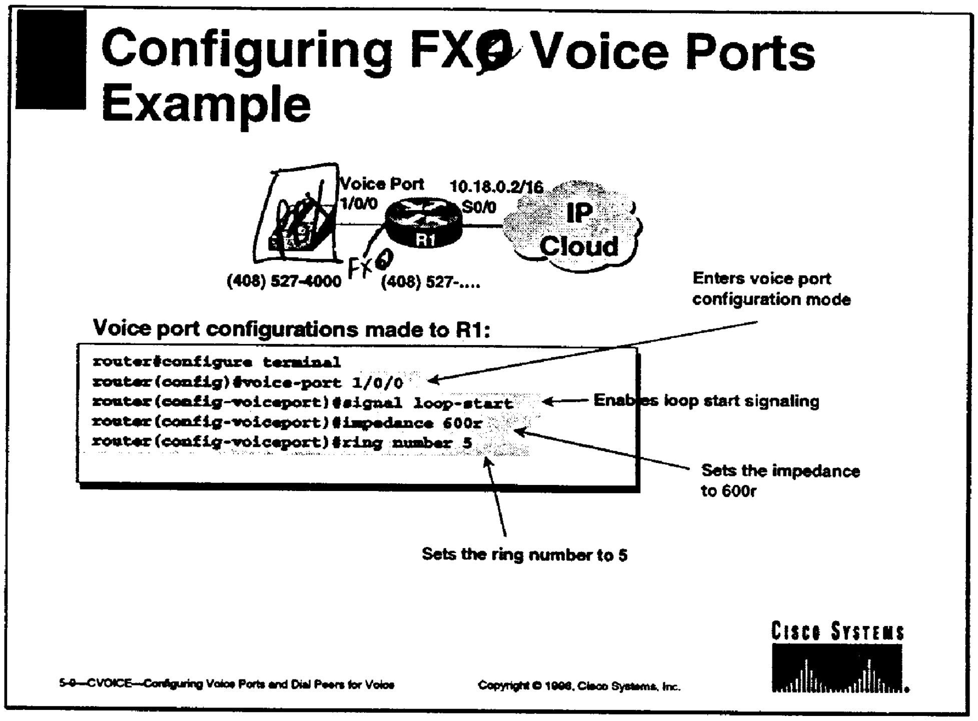
There are also important tuning commands specific to FXO ports.

The **impedance** command allows you to specify the terminating impedance of an FXO voice port interface. If the impedance is set incorrectly and there is an impedance mismatch, a significant amount of echo will be generated that may be masked if you have echo cancellation enabled. The **impedance** command syntax is described in the following table.

impedance Command

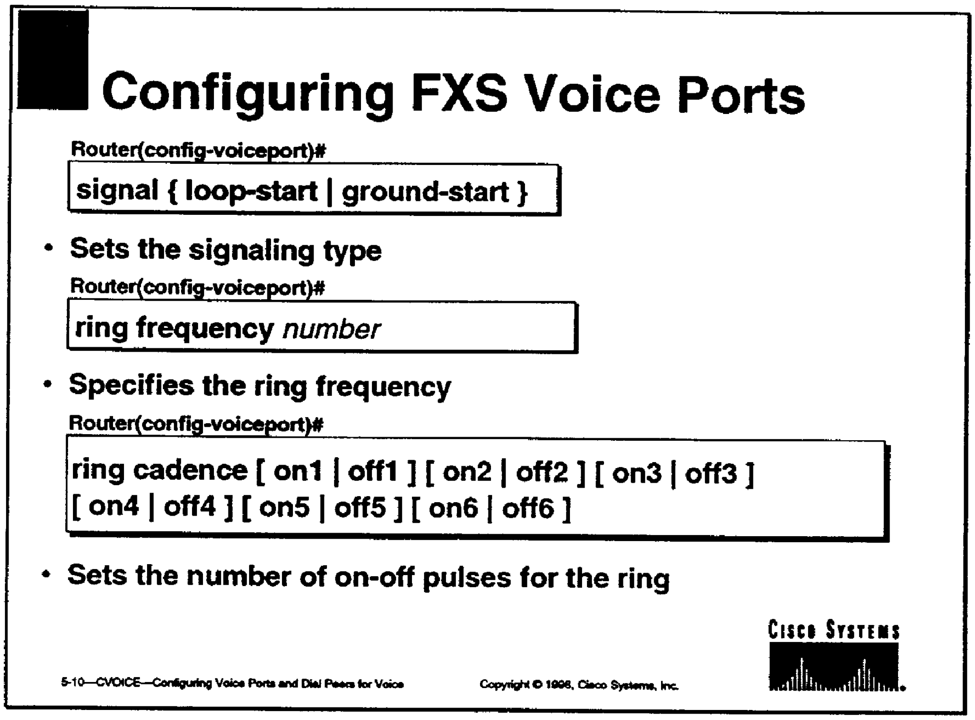
600c	Specifies 600 ohm complex.
600r	Specifies 600 ohm real.
900c	Specifies 900 ohm complex.
900r	Specifies 900 ohm real. (Available only on the MC3810 .)
complex1	Specifies complex1. (Available only on the 2600 and 3600 routers.)
complex2	Specifies complex2. (Available only on the 2600 and 3600 routers.)

The **ring number** *number* command allows you to specify the number of rings before closing the connection. Valid entries are from 1 to 10.



Configuring FXO Voice Ports Example

The configurations in the figure above enable loop-start signaling on a 2600 or 3600's voice port 1/0/0. The impedance is set to 600r and the ring number is set to 5.



Configuring FXS Voice Ports

When you have an analog voice port you may choose to use an FXS interface. The default **voice-port** commands work most of the time for FXS interfaces.

Note This section is not applicable on the AS5300 access server.

However, you may wish to confirm the type of signaling you should use by consulting your telephone vendor. Once you know the type of signaling to use, you can specify the proper signaling with the **signal** command. **signal** command options for FXS interfaces are the same as those on FXO interfaces.

Reference: See the "Configuring FXO Voice Ports" section in this chapter for information on configuring the signal command.

There are also important tuning commands specific to FXS ports.

The **ring frequency** *number* command allows you to specify a ring frequency for your FXS voice port. The ring frequency you select must match the connected telephony equipment. If the frequency is set incorrectly, the attached telephony device may not ring or it may buzz. Valid entries are 20 to 50 Hz and vary by product.

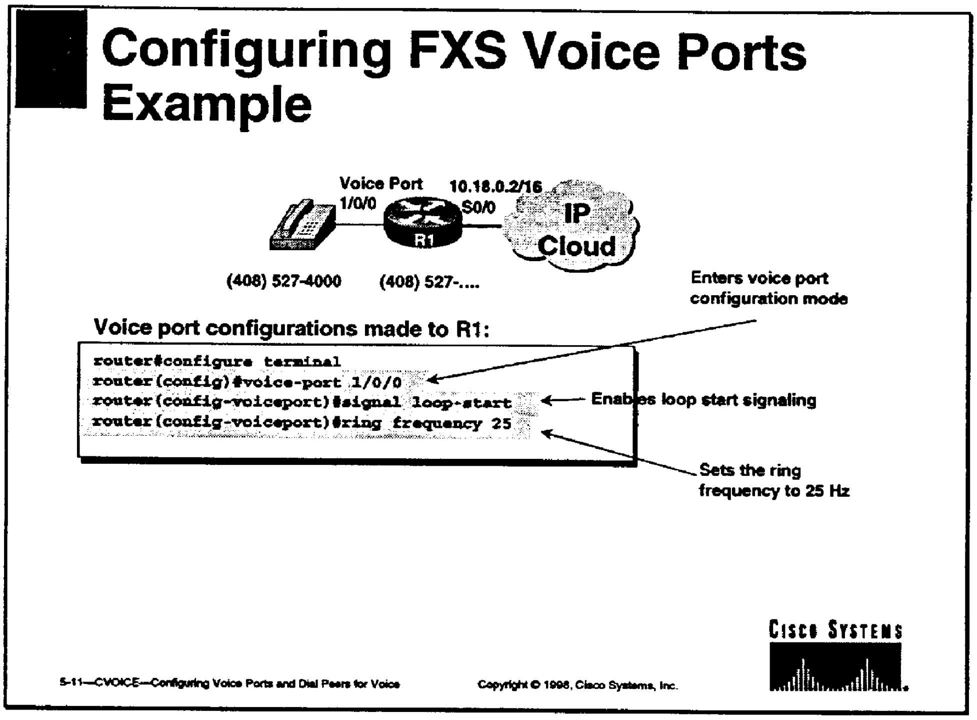
The **ring cadence [on1 | off1] [on2 | off2] [on3 | off3] [on4 | off4] [on5 | off5] [on6 | off6]** command allows you to specify the ring cadence on your FXS voice port. For example, the ring cadence in North America is **ring cadence on2 off4**. The North America cadence is the default.

Note The **ring cadence** command is only applicable on the MC3810.

The following table describes the syntax for the **ring cadence** command:

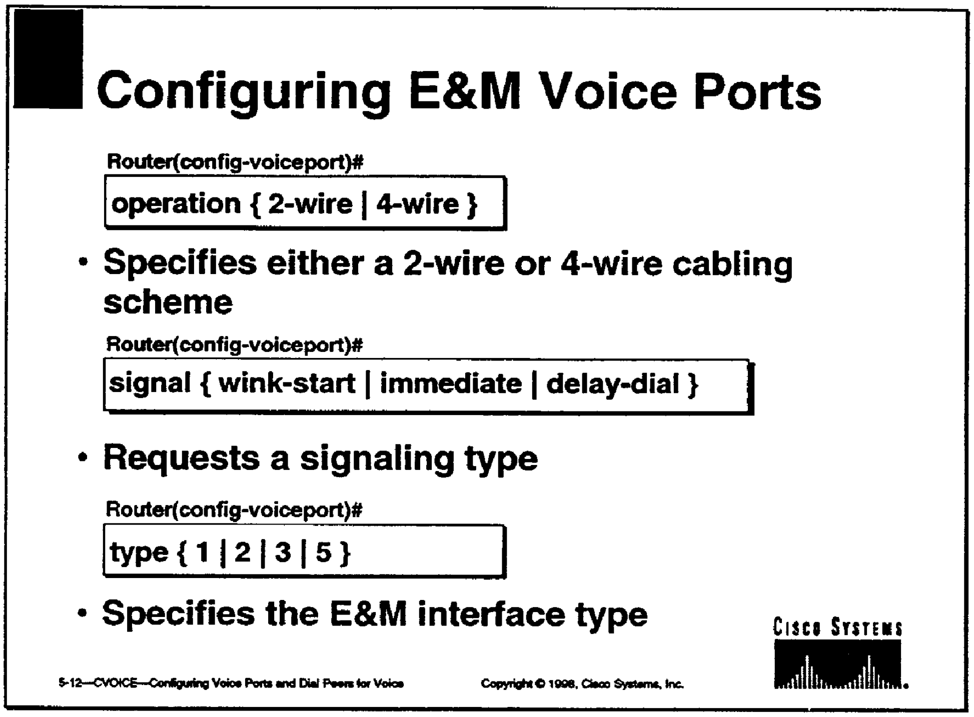
ring cadence Command

on1	Pulse on for 100 milliseconds.
off1	Pulse off for 100 milliseconds.
on2	Pulse on for 200 milliseconds.
off2	Pulse off for 200 milliseconds.
on3	Pulse on for 300 milliseconds.
off3	Pulse off for 300 milliseconds.
on4	Pulse on for 400 milliseconds.
off4	Pulse off for 400 milliseconds.
on5	Pulse on for 500 milliseconds.
off5	Pulse off for 500 milliseconds.
on6	Pulse on for 600 milliseconds.
off6	Pulse off for 600 milliseconds.



Configuring FXS Voice Ports Example

The configurations in the figure above enable loop-start signaling on your 2600 or 3600 FXS voice port 1/0/0. The ring frequency is set to 25 Hz.



Configuring E&M Voice Ports

When configuring an E&M voice port, you should be prepared to spend some time determining the wiring scheme for the PBX you plan to connect. Understanding the timing parameters of the PBX is also imperative. You can get this information from your PBX vendor or reference the manuals that accompanied your PBX.

Note This section is not applicable on the AS5300 access server.

Use the following configuration commands to configure the wiring scheme and the timing.

After you understand the PBX's wiring scheme, you can select a specific cabling scheme for E&M ports with the **operation** voice-port configuration command.

operation Command

2-wire	Specifies a 2-wire E&M cabling scheme.
4-wire	Specifies a 4-wire E&M cabling scheme.

The **operation** command will only affect voice traffic. If the wrong cable scheme is specified, the user might get voice traffic in only one direction.

Note When configuring a 2600 or 3600 router, the **operation** command changes the operation of both voice ports on a VIC.

Note You may need to shut down and reopen your voice port with the **no shut** command for the new value to take effect.

Note This command is not applicable to FXS or FXO interfaces because those are, by definition, 2-wire interfaces. The default for the **operation** command is **2-wire.**

Signaling is independent of 2-wire versus 4-wire settings. To specify the type of signaling for an E&M voice port, use the **signal** voice-port configuration command. Use the **no** form of this command to restore the default value for this command.

Note Notice that the **signal** command options differ from those for an FXS or FXO interface.

signal Command

wink-start	Indicates that the calling side seizes the line by going off-hook on its E-lead, then waits for a short off-hook "wink" indication on its M-lead from the called side before sending address information as DTMF digits or dialing pulses. Used for E&M tie trunk interfaces. This is the default setting for E&M voice ports.
immediate	Indicates that the calling side seizes the line by going off-hook on its E-lead and sends address information as DTMF digits (or dialed pulses on the 2600 and 3600 routers). Used for E&M tie trunk interfaces.
delay-dial	Indicates that the calling side may wait indefinitely for the called side to go on-hook following a configurable initial delay.

Note When configuring a 2600 or 3600 router, the **signal** command changes the operation of both voice ports on a VIC.

Note You may need to shut down and reopen your voice port with the **no shut** command for the new value to take effect.

Some PBXs will miss initial digits if the E&M voice port is configured for immediate signaling. If digits are missed, use delay-dial signaling instead. Some non-Cisco devices have a limited number of DTMF receivers. This type of equipment must delay the calling side until a DTMF receiver is available.

You may need to also specify the E&M interface type for a particular voice port with the **type** command.

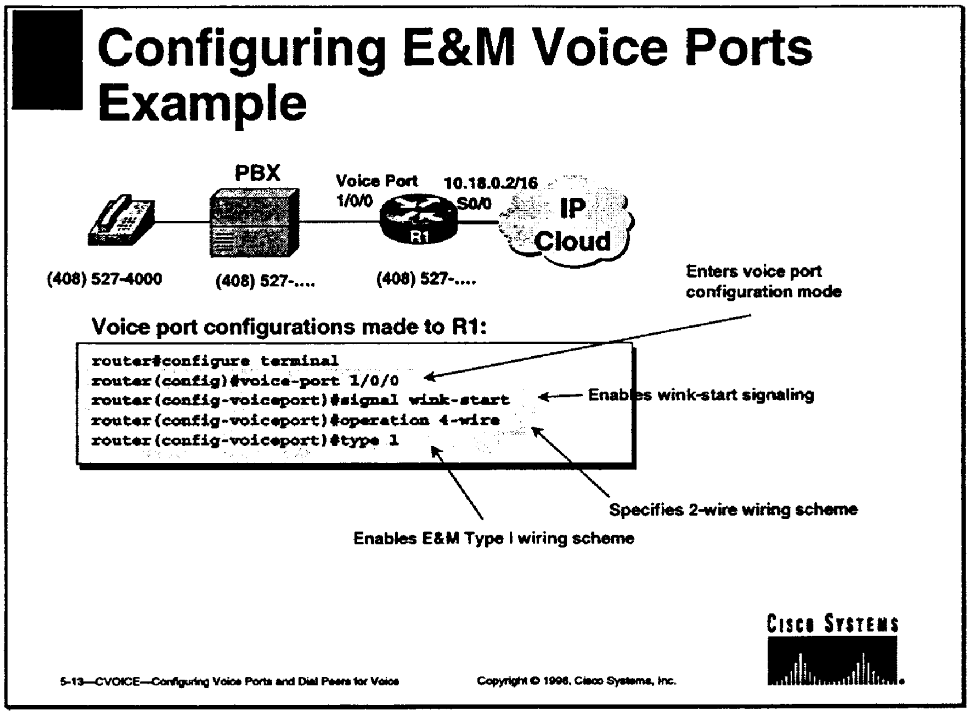
1	Indicates the following default lead configuration: E—output, relay to ground. M—input, referenced to ground.
2	Indicates the following lead configuration: E—output, relay to signal ground (SG). M—input, referenced to ground. Signal battery (SB)—feed for M, connected to -48V. Signal ground (SG)—return for E, galvanically isolated from ground.
3	Indicates the following lead configuration: E—output, relay to ground. M—input, referenced to ground. SB—connected to -48V. SG—connected to ground.
5	Indicates the following lead configuration: E—output, relay to ground. M—input, referenced to -48V.

With **1**, the tie-line equipment generates the E-signal to the PBX type grounding the E-lead. The tie line equipment detects the M-signal by detecting current flow to ground. If you select 1, a common ground must exist between the line equipment and the PBX.

With **2**, the interface requires no common ground between the equipment, thereby avoiding ground loop noise problems. The E-signal is generated toward the PBX by connecting it to SG. M-signal is indicated by the PBX connecting it to SB. Although **2** interfaces do not require a common ground, they do have the tendency to inject noise into the audio paths because they are asymmetrical with respect to the current flow between devices.

With **3**, the interface operates the same as Type **1** interfaces with respect to the E-signal. The M-signal, however, is indicated by the PBX connecting it to SB on assertion and alternately connecting it to SG during inactivity. If you select **3**, a common ground must be shared between equipment.

With **5**, the Type **5** line equipment indicates E-signal to the PBX by grounding the E-lead. The PBX indicates M-signal by grounding the M-lead. A Type **5** interface is quasi-symmetrical in that while the line is up, current flow is more or less equal between the PBX and the line equipment but noise injection is a problem.



Configuring E&M Voice Ports Example

The figure above illustrates wink-start signaling on voice port 1/0/0 on a 2600 or 3600 router. The cabling is set to 4-wire and E&M Type I is enabled.

General Voice Port Tuning Options

Router(config-voiceport)#

input gain *value*

- **Configures a specific input gain for the receiver**

Router(config-voiceport)#

output attenuation *value*

- **Sets the output loss for that transmit side**

Cisco Systems

5-14—CVOICE—Configuring Voice Ports and Dial Peers for Voice

General Voice Port Tuning Options

This section highlights additional voice port configuration commands for general tuning.

The **input gain** *value* command allows you to set the input gain value in decibels for the receiver side of the interface. The value represents the amount of gain to be inserted at the receiver side of the interface. Integer values range from -6 to 14, and the default is 0.

The **output attenuation** *value* command allows you to configure the output attenuation value in decibels for the transmit side of the interface. The value represents the amount of loss to be inserted at the transmit side of the interface. Integer values range from 0 to 14, and the default is 0.

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

General Voice Port Tuning Options (cont.)

Router(config-voiceport)#

`echo-cancel enable`

- **Enables echo cancel**

Router(config-voiceport)#

`echo-cancel coverage { 8 | 16 | 24 | 32 }`

For Listener Echo

- **Sets the coverage time for echo canceller**

Router(config-voiceport)#

`non-linear`

- **Enables nonlinear processing in the echo canceller**

CISCO SYSTEMS

5-15—CVOICE—Configuring Voice Ports and Dial Peers for Voice

The **echo-cancel enable** command enables cancellation of voice that is sent out the interface and is received on the same interface.

The **echo-cancel coverage {8 | 16 | 24 | 32}** command allows you to adjust the size of the echo canceller. This command enables cancellation of voice that is sent out the interface and received back on the same interface within the configured amount of time. If the local loop is longer, the configured value should be extended. The default is 16; 8 is only available on the MC3810.

To enable nonlinear processing in the echo canceller, use the **non-linear** voice-port configuration command.

The **non-linear** 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 no near-end speech is detected.

Note The **non-linear** command has no arguments or keywords.

General Voice Port Tuning Options (cont.)

Router(config-voiceport)#

timeouts initial *seconds*

- **Configures voice port initial timeout values**

Router(config-voiceport)#

timeouts interdigit *seconds*

- **Configures voice port interdigit timeout values**

Router(config-voiceport)#

timing

- **Enables various timing parameters**

Router(config-voiceport)#

vad

- **Enables voice activity detection**

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The **timeouts initial** *seconds* command specifies the number of seconds the system will wait for the caller to input the first digit of dialed digits. Valid entries are integers from 0 to 120, and the default is 10.

Use the **timeouts interdigit** *seconds* command to specify the number of seconds the system will wait (after the caller has input the initial digit) for the caller to input a subsequent digit of the dialed digits. Valid entries are integers from 0 to 120 and the default value is 10.

The **timing** command allows you to set certain timing parameters on your analog voice port.

Note The **timing** command is not applicable on the AS5300.

Reference: Some of the **timing** command options are not available on all routers and all interfaces (i.e., FXO, FXS, and E&M). The valid entries and defaults may also vary. Reference the *Command Reference Guide* for features that apply to your router and applications.

The following table lists the **timing** syntax:

timing Command

clear-wait milliseconds	Indicates the minimum amount of time, in milliseconds, between the inactive seizure signal and the call being cleared.
delay duration milliseconds	Indicates the delay signal duration for delay dial signaling, in milliseconds.
delay-start milliseconds	Indicates the minimum delay time, in milliseconds, from outgoing seizure to out-dial address.
delay-pulse min-delay milliseconds	Indicates the time, in milliseconds, between the generation of wink-like pulses.
digit milliseconds	Indicates the DTMF digit signal duration, in milliseconds.
inter-digit milliseconds	Indicates the DTMF interdigit duration, in milliseconds.
pulse pulses per seconds	Indicates the pulse dialing rate, in pulses per second.
pulse-inter-digit milliseconds	Indicates the pulse dialing interdigit timing, in milliseconds.
wink-duration milliseconds	Indicates the maximum wink-signal duration, in milliseconds, for a wink-start signal.
wink-wait milliseconds	Indicates the maximum wink-wait duration, in milliseconds, for a wink-start signal.

170-240ms

Get From PBX MAN

You may also want to consider configuring voice activity detection (VAD).

In a telephone conversation, you are listening to the conversation approximately half the time and speaking the half. During the silence, voice packets of the silence are continually being generated that can consume much bandwidth in your network. To eliminate the silence and conserve bandwidth, use silence suppression like VAD.

Enable for Low Speed Links

most of Time

Enable VAD with the **vad** command if bandwidth requirements are an issue. Disable VAD with **no vad** if you are operating in a high bandwidth network and voice quality is of the highest importance.

Note Only the MC3810 allows you to configure the **vad** command on the voice port. The VoIP capable routers require you to configure VAD on the dial peer. See Chapter 8 for proper configuration.

To enable VAD for calls using a voice port, use the **vad** voice-port configuration command. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth.

Note Enable VAD with the **vad** command. Disable it with the **no vad** command. The **vad** command has no other arguments, options, or keywords associated with it.

Verifying Voice Ports

```
voipll#show voice port

Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure

<Omitted Information>

 Analog Info Follows:
 Region Tone is set for northamerica
 Currently processing Voice
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm

 Voice card specific Info Follows:
 Signal Type is loopStart
 Ring Frequency is 25 Hz
 Hook Status is On Hook
<Omitted Information>
```

FXS voice port 1/0/0

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Verifying and Troubleshooting Voice Ports

You can verify your voice-port configuration by performing the following tasks:

Step 1 Pick up the handset of an attached telephony device and check for dial tone.

Step 2 If you have dial tone, check for DTMF, voice-band tones like touch tone detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.

Step 3 Use the **show voice port** command to verify that the data configured is correct. The output in the figure above is from a 2600 or 3600 router. You should see output similar to that.

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

Step 1 **ping** the associated IP address to confirm connectivity. If you cannot successfully **ping** your destination, confirm your network configurations with the **show running-config** command.

Step 2 Use the **show voice port** command to make sure that the port is enabled. If the port is off line, use the **no shutdown** command.

Step 3 If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup are correct. Specifically check for 2-wire or 4-wire, wink-start, immediate or delay-dial signal types, and the E&M interface type.

Step 4 Check that the VNM has been correctly installed.

Configuring and Verifying Voice Ports Procedure

To configure and verify voice ports, perform the following steps from global configuration mode:

Step 1 Enter voice-port configuration mode with the **voice-port** command and specify the voice port you wish to configure.

Step 2 Configure the voice port with respect to the type of voice port you are attaching a telephony device to.

Note When configuring an analog E&M interface, pay particular attention to the wiring and timing parameters of the PBX you are connecting. Analog FXS and FXO defaults work most of the time.

Step 3 Exit the voice-port configuration mode and go back to the global configuration mode with the **exit** command.

Step 4 Verify that the voice port was configured properly with the **show voice port** command. If necessary, go back into voice-port configuration mode and make the proper changes.

Step 5 Repeat steps 1 through 4 for all voice ports.

Configuring Voice Ports for ISDN PRI on the 5300

Router(config)#

isdn switch-type [primary-4ess | primay-5ess | primary-dms100 | primary-net5 | primary-ntt | primary-ts014]

- **Specifies your telephone company's switch type**

Router(config)#

controller [t1 | e1] [0 | 1 | 2 | 3]

- **Enter controller configuration mode**

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Configuring Voice Ports for ISDN PRI

If you are using the AS5300, before you configure your voice port, you can configure ISDN PRI. This section will instruct you on the necessary configurations to configure your voice ports as an ISDN PRI group. It will also provide a tutorial on how to configure the D channel.

Reference: This section provides only the necessary information and commands to configure your AS5300 router's voice port as an ISDN PRI group. For an additional tutorial regarding ISDN, refer to the "Configuring, Monitoring, and Troubleshooting Dialup Services" course.

In global configuration mode, you must specify your telephone company's switch that you are attaching to with the **isdn switch-type [primary-4ess | primary-5ess | primary-dms100 | primary-net5 | primary-ntt | primary-ts014]** command.

You must then enter controller configuration mode and specify the controller port you wish to configure. Use the **controller [t1 | e1] [0 | 1 | 2 | 3]** configuration command. The controller ports are labeled 0 to 3 on the quad T1/PRI and quad E1/PRI cards.

(OBR) Cont T1 1 Clock Source internal

(MF1) Cont T1 0 Clock Source Line

Configuring Voice Ports for ISDN PRI on the 5300 (cont.)

Router(config-controller)#

framing [esf | sf | crc4 | nocrc4]

- **Specifies your telephone company's framing type**

Router(config-controller)#

linecode [ami | b8zs | hdb3]

- **Specifies your telephone company's line code**

Router(config-controller)#

clock source [line primary | line secondary | internal]

- **Selects clock source for time-division multiplexing**

Router(config-controller)#

pri-group timeslots [1-24 | 1-31]

- **Specifies all channels for ISDN**

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Find out your telephone company's framing type and line-code type and then use the following commands to configure the framing and line-code types, respectively:

- Use the **framing** command to specify your telephone company's framing type. **sf** and **esf** are used on T1 lines and **crc4** and **no-crc4** are used on E1 lines.

framing Command

sf	Specifies superframe as the T1 frame type.
esf	Specifies Extended Super Frame as the T1 frame type.
crc4	Specifies CRC4 as the E1 frame type.
no-crc4	Specifies no CRC4 as the E1 frame type.

- Use the **linecode** command to select the line-code type for your T1 or E1 line.

linecode Command

ami	Specifies alternate mark inversion (AMI) as the line-code type.
b8zs	Specifies B8ZS as the line-code type. Valid for T1 controllers only.
hdb3	Specifies High Density Bipolar 3 as the line-code type. Valid for E1 controllers only.

You must also enter the clock source for the line. One line should be **clock source line primary**. The others should be configured as **clock source line secondary** or **clock source internal**.

Note Only one PRI can be clock source primary and only one PRI can be clock source secondary. Remaining PRIs must be configured as clock source internal.

You must also configure all channels for ISDN. Enter **pri-group timeslots 1-24** for T1. If E1, enter **pri-group timeslots 1-31**.

Procedure

To configure your voice port for ISDN PRI, perform the following steps:

Step 1 From the global configuration mode, enter your telephone company's switch type with the **isdn switch-type** command.

Step 2 Enter controller configuration mode and prepare to configure the controller with the **controller** command.

Step 3 Enter your telephone company's framing type with the **framing** command.

Step 4 Enter your telephone company's line-code type with the **linecode** command.

Step 5 Enter the **clock source** for the line.

Step 6 Configure all channels for ISDN with the **pri-group timeslots** command.

Step 7 Repeat steps 2 through 6 for the remaining three controllers.

Configuring the D Channels

Router(config)#

interface serial [*controller*:15 | *controller*:23]

- **Enters interface configuration mode**

Router(config-if)#

isdn incoming-voice modem

- **Configures all incoming voice calls to go to the modem**

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Configuring the D Channels

You are now ready to configure the ISDN D channels, which carry the control and signaling information for ISDN calls, for each ISDN PRI line.

Enter serial interface configuration mode by entering the **interface** command. After you have configured the controller, a corresponding D channel serial interface is created instantly. For example, serial interface 0:23 is the D channel for controller 0. You must configure each serial interface to receive incoming and send outgoing modem signaling.

You must also configure all incoming voice calls to go to the modems by using the **isdn incoming-voice modem** command.

Procedure

To configure your D channels, perform the following tasks:

Step 1 From the global configuration mode, enter interface configuration mode with the **interface** command.

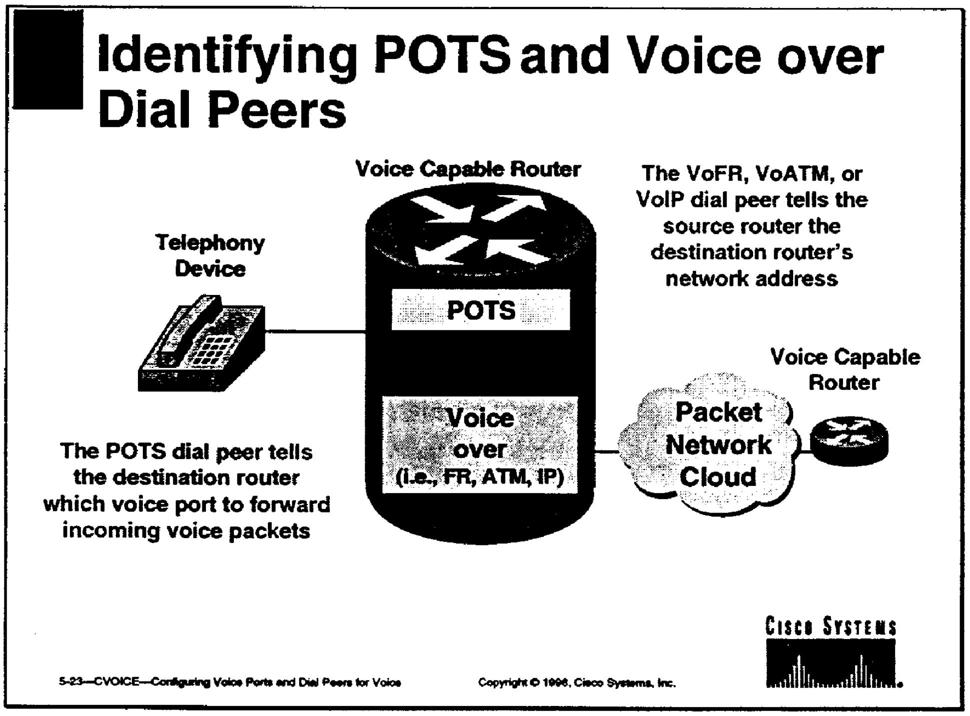
Step 2 Configure incoming voice calls to go to the modem with the **isdn incoming-voice modem** command.

Configuring and Verifying Dial Peers

Dial peers are an addressable call endpoint. It is the dial peers that establish logical connections called call legs to complete an end-to-end call. There are two kinds of dial peers, POTS and voice over (i.e., VoFR, VoATM, VoIP) dial peers.

This section identifies what dial peers are and how to configure them to complete an end-to-end call. The section's topics are as follows:

- Identifying POTS and Voice over Dial Peers
- Configuring POTS Dial Peers
- Configuring VoFR, VoATM, or VoIP Dial Peers
- Verifying Dial Peer Configuration
- Configuring Dial Peers Procedure



Identifying POTS and Voice over Dial Peers

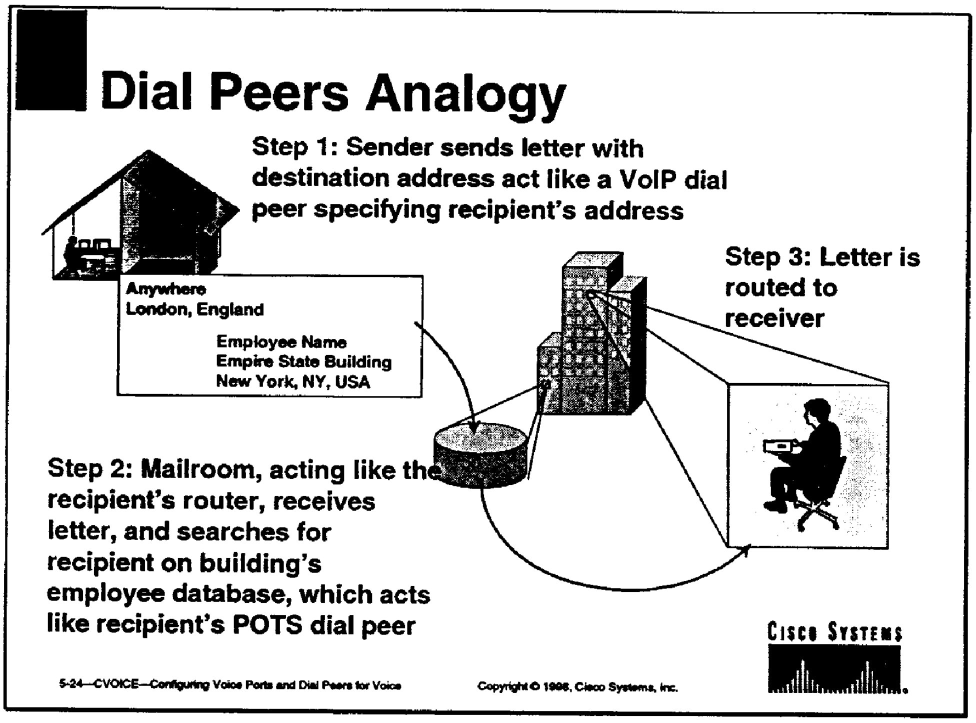
POTS dial peers are those connected to a traditional telephony network. POTS peers point to a particular voice port on a voice network device. POTS dial peers tell the router to which interface port each telephony device connects. The router will then know where to forward incoming calls because the POTS dial peer specified the voice port.

Voice over dial peers like VoFR, VoATM, or VoIP are those connected via a packet network backbone; in the case of VoATM, for example, this is an ATM network. VoFR, VoATM, or VoIP peers point to specific network devices, like the router. VoFR, VoATM, or VoIP dial peers are reserved for outgoing calls. VoFR, VoATM, or VoIP dial peers at the source router associate the destination address with the destination router.

To place a VoFR, VoATM, or VoIP call, the source router must be configured with a VoFR, VoATM, or VoIP dial peer specifying the recipient's destination address. The recipient's router must be configured with a POTS dial peer specifying which voice port and telephony device to forward the voice call.

The next two sections further illustrate POTS and voice over dial peers. They are:

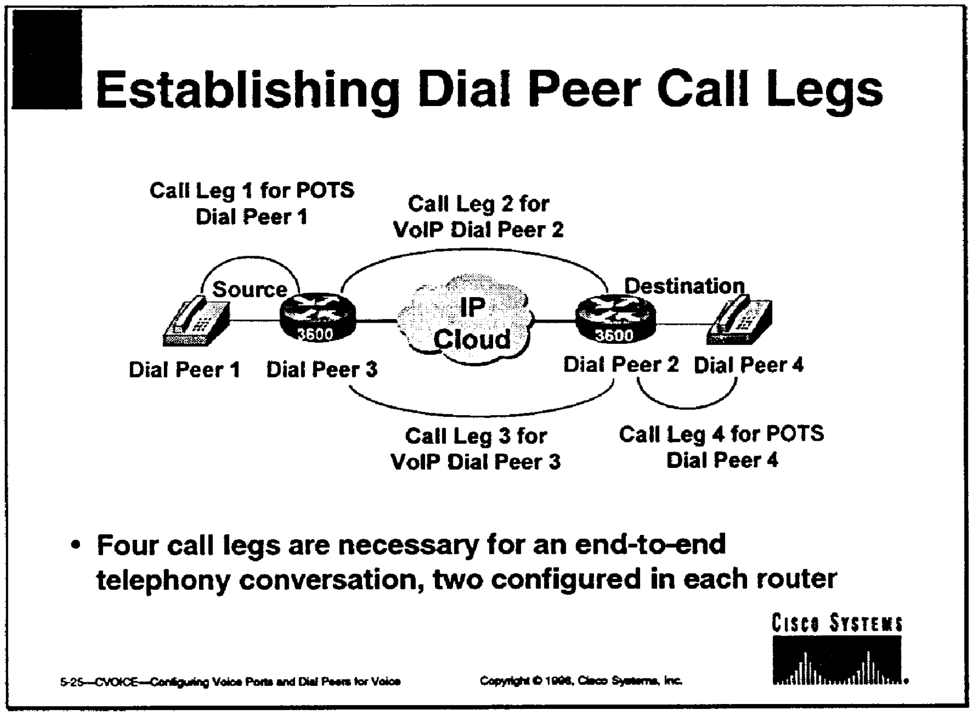
- Dial Peers Analogy
- Establishing Dial Peer Call Legs



Dial Peers Analogy

Imagine working at the Empire State Building in New York City. Your friend sent you a letter with the following address: Your Name, Empire State Building, New York, NY. The odds are good that the letter arrived at the Empire State Building's central mail stop because the building is famous worldwide. However, tens of thousands of people work at the Empire State Building. So, building management maintains an employee locator database. The mailroom simply searched for your name and identified your work location. Only then was it able to forward your mail to you.

The address "Empire State Building" acts like a network destination address specified with the VoFR, VoATM, or VoIP dial peer at the source router. It tells the network the general location of the recipient. The POTS dial peer acts like the employee locator database. The POTS dial peer tells the receiving router which interface port and telephony device to forward calls to.



Establishing Dial Peer Call Legs

A call leg is a logical connection between the router and either a telephony endpoint or another endpoint using a session protocol like Frame Relay, ATM, or IP. Call legs are router centric, which means that when an inbound call arrives on the router, it is processed as a separate call until the destination is determined. A second outbound call leg is then established. The two call legs are then conferenced into an end-to-end call through the router.

The connections are made when dial peers are configured on each interface. An end-to-end call comprises four call legs, two from the perspective of the source router, as shown in the dial peers graphic, and two from the perspective of the destination router. To complete an end-to-end call and send voice packets back and forth, all four dial peers must be configured.

The figure above illustrates an end-to-end call with the four call legs. The source router must have dial peers 1 and 2 configured to specify call legs 1 and 2. To send voice packets back to the source router, the destination router must have dial peers 3 and 4 configured to specify call legs 3 and 4.

Configuring POTS Dial Peers

2600 & 3600

Router(config)# (1 - 2.1 Bill) (1 - 100000) 3810

```
dial-peer voice tag-number pots
```

Do Not Repeat Tag #

- **Enters POTS dial peer configuration mode**

Router(config-dial-peer)#

```
destination-pattern [ + ] string
```

- **Specifies the prefix or full E.164 telephone number to be used for a dial peer**

Router(config-dial-peer)#

```
port slot-number/subunit-number/port
```

- **Specifies voice port the telephony device attaches to on a POTS dial peer on a 2600 or 3600**

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Configuring POTS Dial Peers

Use the **dial-peer voice** *tag-number* **pots** global configuration command to enter the dial peer configuration mode and specify the voice telephony device you wish to configure.

dial-peer voice Command

tag-number	Digit(s), unique to the local router, defining a particular dial peer. Valid entries are from 1 to 2147483647 (up to 100000 on the MC3810).
pots	Indicates that this is a POTS peer using basic telephone service.

Note The *tag-number* is an arbitrary identifier you assign to uniquely identify the dial peer.

Note To modify the tag configuration after you configure the dial peer voice tag, enter the dial-peer voice and the tag-number and press **Return**.

Once in dial peer configuration mode, use the **destination-pattern** command to specify the extension or the full E.164 telephone number (depending on your dial plan) of the destination dial peer.

Note Remember, the destination for a POTS dial peer is a telephony device attached to the router's voice port.

---

string	Digits 0 through 9, letters A through D, or private dialing plan telephone number. Valid entries are: • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit.

To associate the voice telephony dial peer with a specific dial interface, specify the voice port connected to the POTS dial peer with the **port** *port number* command.

The port number will be different for each router. Use the proper port addressing scheme for the router you are attached to. The schemes follow:

- 2600 or 3600—**port** slot-number/subunit-number/port
- MC3810—**port** slot/port
- AS5300—**port** controller number:**D**

Note When configuring dial peers, you need to understand the relationship between the destination pattern and the port. The destination pattern is the telephone number of the voice device attached to the voice port. The port represents the route from the router to telephony device.

Note All three configuration commands are necessary to establish POTS dial peer.

Configuring POTS Dial Peers (cont.)

Router(config-dial-peer)#

prefix *string*

- **Specifies the prefix for the dialed digits on a dial peer**

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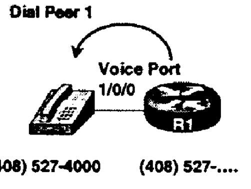
You can use the **prefix** *string* command to optionally specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** *string* value is sent to the telephony interface first, before the telephone number associated with the dial peer.

The *string* can be integers representing the prefix of the telephone number associated with the specified dial peer. Valid numbers are 0 through 9, and a comma (,). Use the comma to include a pause in the prefix.

If you want to configure different prefixes for dialed numbers on the same interface, you need to configure different dial peers.

Configuring POTS Dial Peers Example



Configuration for dial peer 1 on R1:

```
dial-peer voice 1 pots
destination-pattern 14085274000
port 1/0/0
```

5-28—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Configuring POTS Dial Peers Example

The graphic illustrates proper POTS dial peer configuration on a 2600 or 3600 router. By entering **dial-peer voice 1 pots** you are telling router 1 that dial peer 1 is a POTS dial peer and you are calling it 1. With the **destination-pattern 14085274000** command, you are telling the router the telephony device's phone number. The **port 1/0/0** command tells the router that the telephony device is plugged into module slot 1, VIC subslot 0, voice port 0.

Configuring VoFR, VoATM, or VoIP Dial Peers

Router(config)#

dial-peer voice *tag-number* { **vofr** | **voatm** | **voip** }

- Enters VoFR, VoATM, or VoIP dial peer configuration mode

Router(config-dial-peer)#

destination-pattern [+] *string*

- Specifies the prefix or full E.164 telephone number to be used for a dial peer

Router(config-dial-peer)#

session-target { **serial** *interface dlci* | **serial 2** *number* | **ipv4:***IP address* }

- Specifies the dial peer's network address

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Configuring VoFR, VoATM, or VoIP Dial Peers

Use the **dial-peer voice** *tag-number* {**vofr** | **voatm** | **voip**} global configuration command to enter the dial peer configuration mode and specify the method of network related encapsulation.

dial-peer voice Command

tag-number	Digit(s), unique to the local router, defining a particular dial peer. Valid entries are from 1 to 2147483647 (up to 100000 on the MC3810).
vofr	Indicates that this is a VoFR peer using encapsulation on the Frame Relay backbone network. (Applicable on MC3810 only.)
voatm	Indicates that this is a VoATM peer using the real-time AAL5 voice encapsulation on the ATM backbone network. (Applicable on MC3810 only.)
voip	Indicates that this is a VoIP dial peer using voice encapsulation on the IP backbone. (Applicable on 2600, 5300, or AS5300 only.)

Note The *tag-number* is an arbitrary identifier you assign to uniquely identify the dial peer.

Note To modify the tag configuration after you configure the dial peer voice tag, enter the **dial-peer voice** and the *tag-number* and press **Return**.

Once in dial peer configuration mode, use the **destination-pattern** command to specify the full E.164 telephone number (depending on your dial plan) of the destination dial peer.

Note Remember, the destination for a VoFR, VoATM, or VoIP dial peer is another network device (i.e., voice capable router) you are sending a voice call.

destination-pattern Command

string	Digits 0 through 9, letters A through D, or private dialing plan telephone number. Valid entries are: • Plus sign (+), which is optionally used as the first digit to indicate an E.164 standard number. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit.

If you are configuring a VoFR, VoATM, or VoIP dial peer, you must specify a network address for a specified dial peer. For example, in VoIP this address will be the IP endpoint address (the address of the dial peer you wish to contact). Use the **session target** command to specify the network address of the router you are trying to contact.

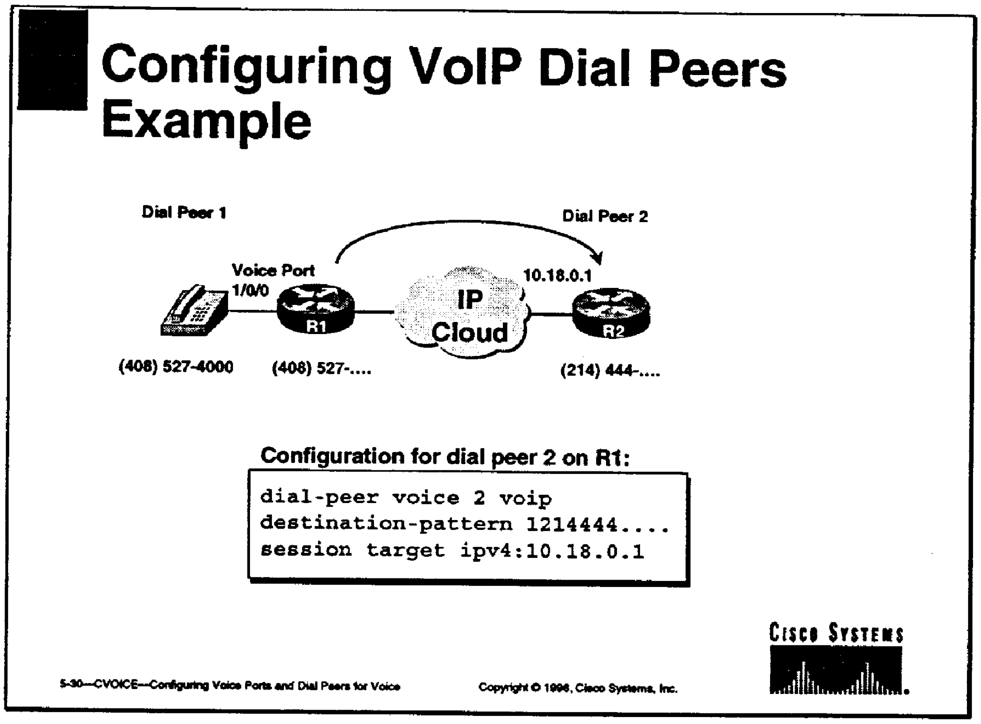
session target Command

serial interface dlci	Configure the Frame Relay session target for the dial peer when running VoFR.
serial 2 number	Configure the ATM session target for the dial peer. The number value is a number from 1 to 1023. Used with VoATM.
ipv4:IP address	IP address of the dial peer. Used with VoIP.

Note The Cisco MC3810 supports ATM traffic over serial port 2 only. Reference the VoATM chapter for more information.

Note When configuring dial peers, you need to understand the relationship between the destination pattern and the session target. The destination pattern is the telephone number of the voice device attached to a voice port at the other end of the network connection. The session target represents the route to a port on the other end of the network connection.

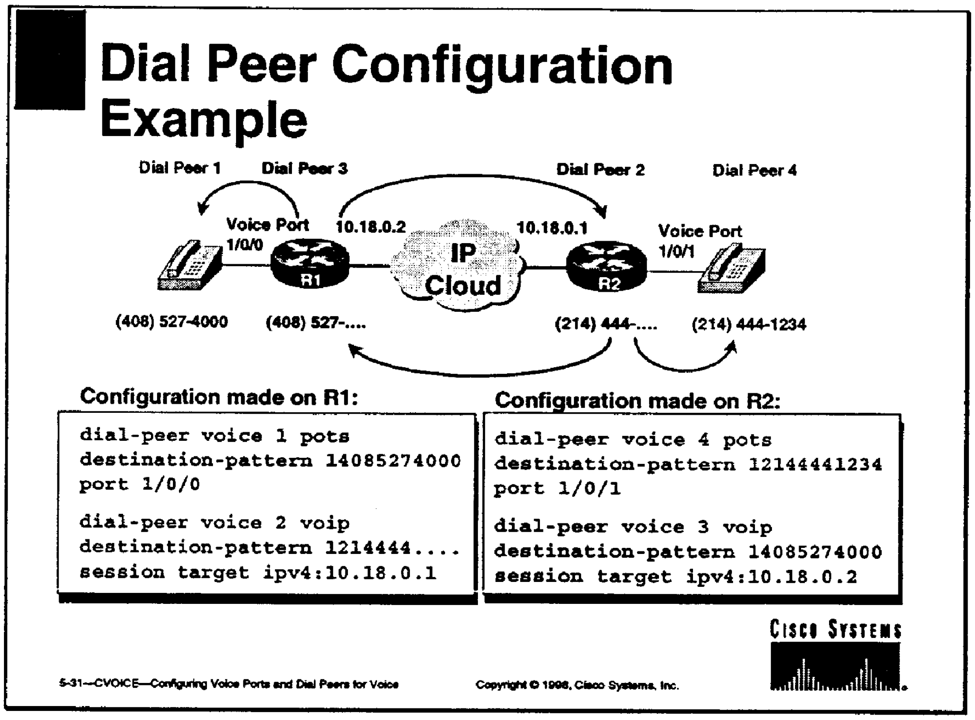
Note All three configuration commands are necessary to establish a VoFR, VoATM, or VoIP dial peer.



Configuring VoIP Dial Peers Example

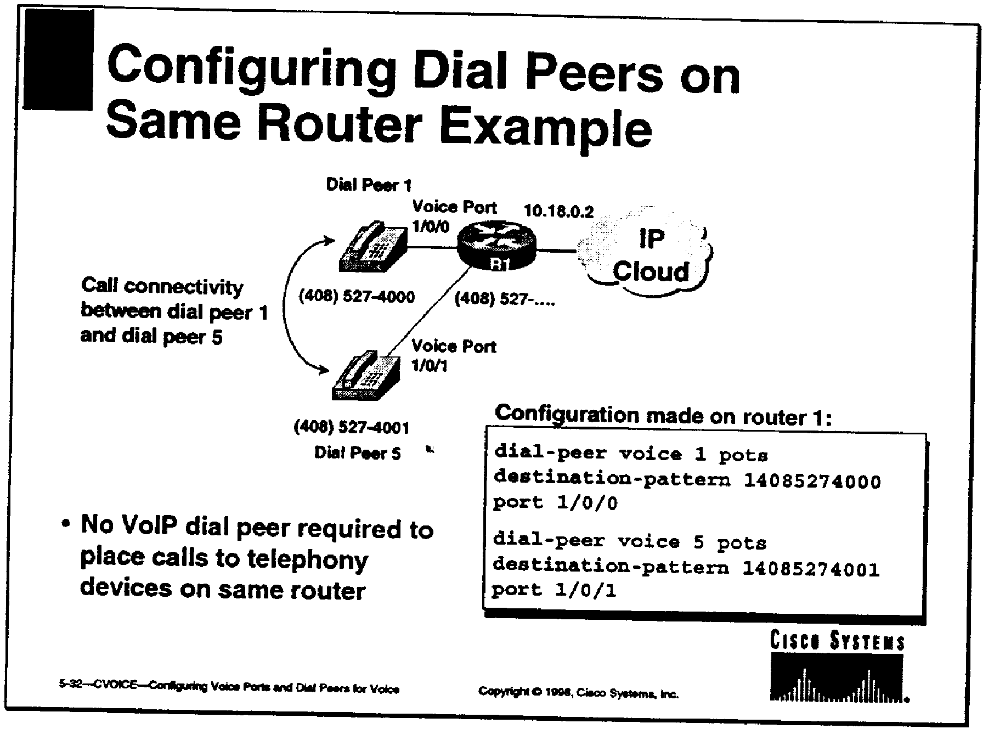
The voice over dial peers example is an example using VoIP. The graphic illustrates proper VoIP dial peer configuration on a 2600 or 3600 router. By entering **dial-peer voice 2 voip** you are identifying (on router 1) router 2 as dial peer 2 and that it is a VoIP dial peer. With the **destination-pattern +1214444....** you are telling router 1 the phone number of the telephony device you are calling. The **session target ipv:10.18.0.1** command is the IP address of router 2.

Note As indicated on the **destination pattern** command table, the "...." replacing the last four telephone number digits are "wildcard" placeholders that match any digit 0 through 9. The placeholder "." is commonly used in the VoIP dial peer configuration mode and means that from router 10.18.0.2, for example, calling any number string that begins with the digits "+1214444" plus the remaining phone number digits will result in a connection to router 10.18.0.1 and implies that router 10.18.0.1 services all numbers beginning with those digits.



Dial Peer Configuration Example

Dial peers 3 and 4 must now be configured on router 2 so there can be an end-to-end call. The example is a VoIP example using a 2600 or 3600 router. See the graphic for correct configuration.



Configuring Dial Peers on Same Router Example

There is one exception to the need for VoFR, VoATM, or VoIP dial peers: when both POTS dial peers are connected to the same router. The graphic shows POTS dial peers configured for each telephony device.

Verifying Dial Peer Configuration—*show dial-peer voice* Output

```
router#show dial-peer voice
VoiceEncapPeer11
    tag = 11, destination-pattern = `1111111111',
    answer-address = `',
    group = 11, Admin state is up, Operation state is up
    type = pots, prefix = `',
    session-target = `', voice-port = 1/0/0
    Connect Time = 46214, Charged Units = 0
    Successful Calls = 50, Failed Calls = 0
    Accepted Calls = 65, Refused Calls = 0
    Last Disconnect Cause is "10 "
    Last Disconnect Text is "normal call clearing."
    Last Setup Time = 34485570

<continued on next slide>
```

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Verifying Dial Peer Configuration

Verify that the voice connection is working by doing the following:

- Pick up the handset on a telephone connected to the configuration and verify that you can get a dial tone.
- Make a call from the local telephone to a configured dial peer and verify that the call attempt is successful.

You can check your dial peer configuration by performing the following task:

- Use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers. See the graphic for configured output.

Output for a POTS dial peer should look like the output in the figure above.

Verifying Dial Peer Configuration—*show dial-peer voice* Output (cont.)

```
VoiceOverIpPeer110
    tag = 110, destination-pattern = `1101101....',
    answer-address = `',
    group = 110, Admin state is up, Operation state is up
    type = voip, session-target = `ipv4:10.1.1.1',
    ip precedence: 0    UDP checksum = disabled
    session-protocol = cisco, req-qos = best-effort,
    acc-qos = best-effort,
    fax-rate = voice, codec = g729r8,
    Expect factor = 10, Icpif = 30,
    VAD = enabled, Poor QOV Trap = disabled
    Connect Time = 44739, Charged Units = 0
    Successful Calls = 35, Failed Calls = 0
    Accepted Calls = 35, Refused Calls = 0
    Last Disconnect Cause is "10 "
    Last Disconnect Text is "normal call clearing."
    Last Setup Time = 34461803
```

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5-34—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Output for a VoFR, VoATM, or VoIP dial peer should similar to the output in the figure above.

Configuring Dial Peers Procedure

To configure dial peers on your router, perform the following steps:

Step 1 Enter the dial peer configuration mode and specify the POTS dial peer you wish to configure.

Step 2 Set the destination pattern of the POTS peer.

Step 3 Specify the voice port the POTS peer is connected to.

Step 4 Exit the dial peer configuration mode for the specific POTS peer.

Step 5 Repeat for all POTS dial peers.

Step 6 Enter the dial peer configuration mode and specify the VoFR, VoATM, or VoIP dial peer you wish to configure.

Step 7 Set the destination pattern of the VoFR, VoATM, or VoIP peer.

Step 8 Specify the session target of the peer you wish to attach to.

Step 9 Exit VoFR, VoATM, or VoIP dial peer configuration mode.

Step 10 Repeat for all VoFR, VoATM, or VoIP dial peers.

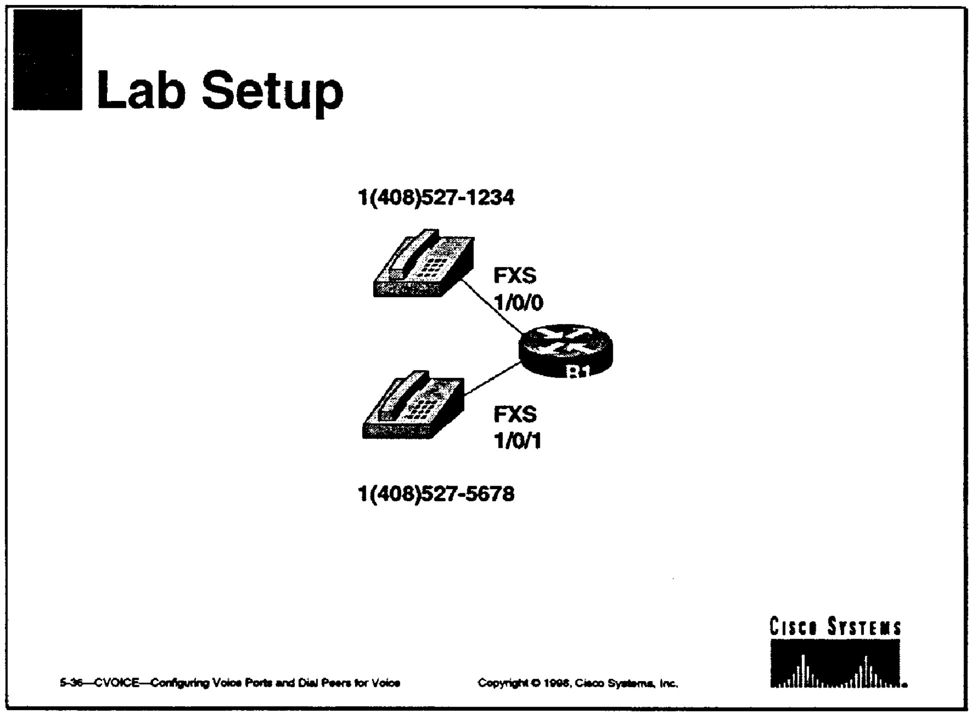
Step 11 Verify proper configuration with the **show dial-peer voice** command.

Lab Exercise: Configuring Voice Ports and POTS Dial Peers

Use the following laboratory exercises to practice what you have learned in this lesson.

Lab Setup

Use the following information to prepare for this exercise. The graphic illustrates the lab configuration for router R1. Use information supplied in the table to make router specific connections.



Router	POTS Dial Peers and Phone Number	POTS Dial Peers and Port
R1	1–14085271234 2–14085275678	1–1/0/0 2–1/0/1

To prepare for the exercise, from the global configuration mode, perform the following steps:

Step 1 Name the router with the **hostname R1** command.

Step 2 Exit configuration mode with the **end** command.

Step 3 Copy the configurations to memory with the **copy running-config startup-config** command.

Scenario

Connect all local telephony devices to your router, so users can communicate without using traditional telephony connections.

Directions

Perform the following tasks for each router.

Exercise: Configuring Voice Ports

To configure the FXS voice ports, perform the following tasks from the global configuration mode:

Step 1 Because you are configuring an FXS voice port, you should have dial tone. Lift up each telephone's receiver to confirm you have dial tone.

Step 2 Enter the voice-port configuration mode and prepare to configure your analog voice port in the router by entering the **voice-port 1/0/0** command.

Step 3 Set the country call progress tone to North America with the **cptone northamerica** command.

Step 4 Set your type of signaling to loop-start signaling. Use the **signal loopstart** command.

Step 5 Set the ring frequency to 25 Hz with the **ring frequency 25** command.

Step 6 Enable echo cancellation with the **echo-cancel enable** command.

Step 7 Set the number of seconds the system will wait for your initial dialed digit to 20 seconds with the **timeouts initial 20** command.

Step 8 Activate your voice port with the **no shutdown** command.

Step 9 Exit all configuration modes and go back to the privileged EXEC mode with the **end** command.

Step 10 Save your configuration with the **copy run start** command.

Step 11 Verify voice port configuration by picking up the telephone's handset and checking for dial tone.

Step 12 If you have dial tone, check for DTMF detection by dialing digits. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.

Step 13 Use the **show voice port 1/0/0** command to verify that the data configured is correct.

Correct configuration should look like the following:

```
R1#show voice port 1/0/0
Foreign Exchange Station 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
Type of VoicePort is FXS
Operation State is DORMANT
Administrative State is UP
The Last Interface Down Failure Cause is
Alias is NULL
Noise Regeneration is enabled
```

```
Non Linear Processing is enabled
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancel Coverage is set to 16 ms
Connection Mode is normal
Connection Number is
Initial Time Out is set to 20 s
Interdigit Time Out is set to 10 s

Analog Info Follows:
Region Tone is set for northamerica
Currently processing Voice
Maintenance Mode Set to None (not in mtc mode)
Number of signaling protocol errors are 0
Impedance is set to 600r Ohm

Voice card specific Info Follows:
Signal Type is loopStart
Ring Frequency is 25 Hz
Hook Status is On Hook
Ring Active Status is inactive
Ring Ground Status is inactive
Tip Ground Status is inactive
Digit Duration Timing is set to 100 ms
InterDigit Duration Timing is set to 100 ms
```

Step 14 Enter global configuration mode again with the **config t** command.

Step 15 Enter the voice-port configuration mode and prepare to configure your second analog voice port in the router by entering the **voice-port 1/0/1** command.

Step 16 Repeat steps 3 through 15 to configure voice port 1/0/1.

Exercise: Configuring POTS Dial Peers

To configure the POTS dial peers, perform the following tasks from the global configuration mode:

Step 1 Enter the local dial peer configuration mode and prepare to configure the local dial peer using the **dial-peer voice 1 pots** command.

Step 2 Specify the phone number using the **destination-pattern 14085271234** command.

Step 3 Associate the local dial peer with voice port 1/0/0 using the **port 1/0/0** command.

Step 4 Exit the local dial peer configuration mode using the **end** command.

Step 5 You can verify proper configuration with the **show dial-peer voice** command.

Proper configuration should look like the following:

```
R1#show dial-peer voice
VoiceEncapPeer1
   tag = 1, destination-pattern = `14085271234',
   answer-address = `',
   group = 1, Admin state is up, Operation state is up
   type = pots, prefix = `',
   session-target = `', voice-port = 1/0/0
   Connect Time = 0, Charged Units = 0
   Successful Calls = 0, Failed Calls = 0
```

```
Accepted Calls = 0, Refused Calls = 0
Last Disconnect Cause is ""
Last Disconnect Text is ""
Last Setup Time =
```

Step 6 Enter global configuration mode with the **config t** command.

Step 7 Enter the local dial peer configuration mode and prepare to configure the local dial peer using the **dial-peer voice 2 pots** command.

Step 8 Specify the phone number using the **destination-pattern 14085275678** command.

Step 9 Associate the local dial peer with voice port 1/0/1 using the **port 1/0/1** command.

Step 10 Exit all configuration modes using the **end** command.

Step 11 Verify proper configuration with the **show dial-peer voice** command.

Step 12 Save your configurations with the **copy run start** command.

Step 13 Verify that both telephony devices still have dial tone. Once you confirm that you have dial tone, place them back on-hook. If you do not have dial tone, check voice port and dial peer configurations again.

Step 14 You should now be able to place a local call. Place a call from telephony device 101 to 102. Make sure that you can hear and speak in both directions.

Step 15 Place a call from telephony device 102 to 101. Make sure that you can hear and speak in both directions.

Q1) Do you need a voice over dial peer to place calls to telephony devices attached to the router? Why or why not?

Completion Criteria

You have completed the lab when you have successfully placed an end-to-end call from each telephone attached to your router.

Answers

Q1) Do you need a voice over dial peer to place calls to telephony devices attached to the router? Why or why not?

No, because both telephony devices are attached to the same router, you do not need to place a call into the network.

Summary

Summary

In this chapter you learned how to complete the following tasks:

- **Configure and verify the voice ports so you properly interface to the attached telephony devices**
- **Configure and verify POTS dial peers so you can place calls to others connected to your router**
- **Configure and verify VoIP, VoFR, or VoATM dial peers so you can place calls to others in your voice network**

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5-39—CVOICE—Configuring Voice Ports and Dial Peers for Voice

Configuring the MC3810 for Voice over Frame Relay

Objectives

Objectives

- Describe VoFR software and WAN trunk interfaces of the MC3810
- List the Cisco IOS™ commands specific to configuring the MC3810 for VoFR
- Configure an MC3810 for VoFR using Cisco IOS software so students can connect and optimize their own VoFR enterprise branch offices

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6-2—CVOICE—Configuring an MC3810 for Voice over Frame Relay

Objectives (cont.)

- **Configure the MC3810 WAN services, Frame Relay voice port interfaces, and dial peers using Cisco IOS software so a phone call can be successfully completed between two MC3810s**

6-3—CVOICE—Configuring an MC3810 for Voice over Frame Relay

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Introduction

This chapter introduces the student to the MC3810, gives a brief overview of how Frame Relay operates, and provides a description of relevant MC3810 Voice over Frame Relay hardware, cable interface connectivity, and software features. The chapter also provides a summary of configuration concepts and commands, and an example of a Voice over Frame Relay configuration for the MC3810.

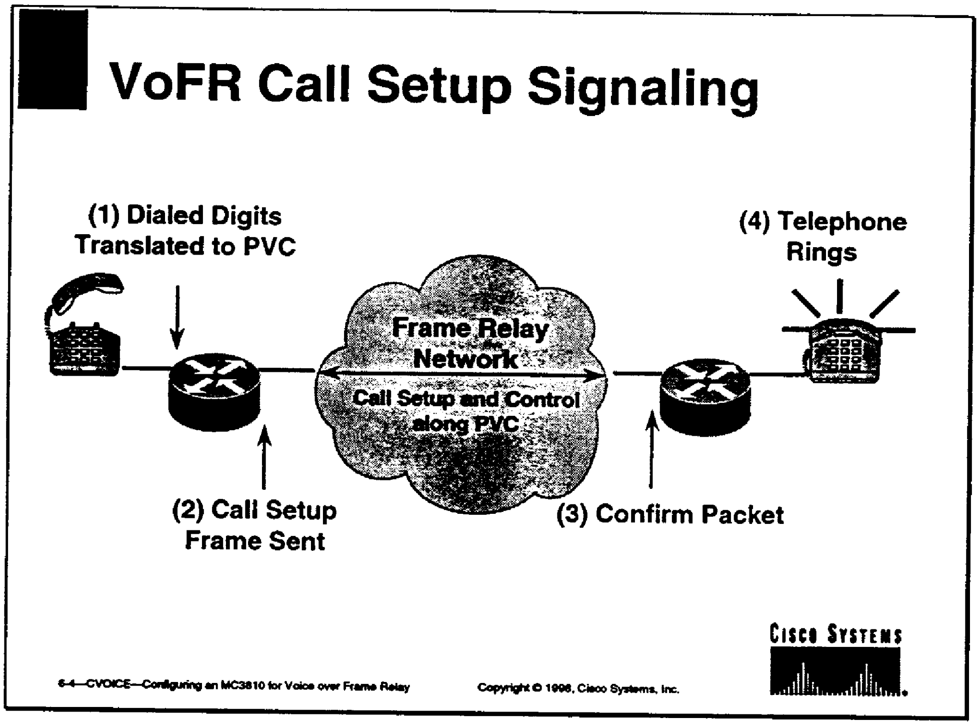
Frame Relay Background Information

Frame Relay is a networking technology that allows multiple logical paths to be connected via a single access line to form virtual meshed networks. This technology provides a new way of sharing network bandwidth that provides flexibility and higher effective throughputs through oversubscription and instantaneous dynamic bandwidth allocation.

Frame Relay supersedes X.25 and older public networking systems and is extremely well-suited for both public and private networking implementations due to its configuration flexibility, high-speed interfaces, and ability to integrate traditional SNA and packet data, bursty LAN data, Internet connectivity, and both voice and video communications.

Whereas TDM-based solutions reserve a fixed time slot for a device whether the device has anything to send or not, Frame Relay statistically multiplexes traffic, thus allocating any unused bandwidth "on demand" or on an as-needed basis.

Frame Relay provides a means for statistically multiplexing many logical virtual circuits over a single physical transmission link. To configure Frame Relay for voice, you must specify the serial interface you wish to use for transporting voice through a network service cloud in the form of Frame Relay data transmission packets.



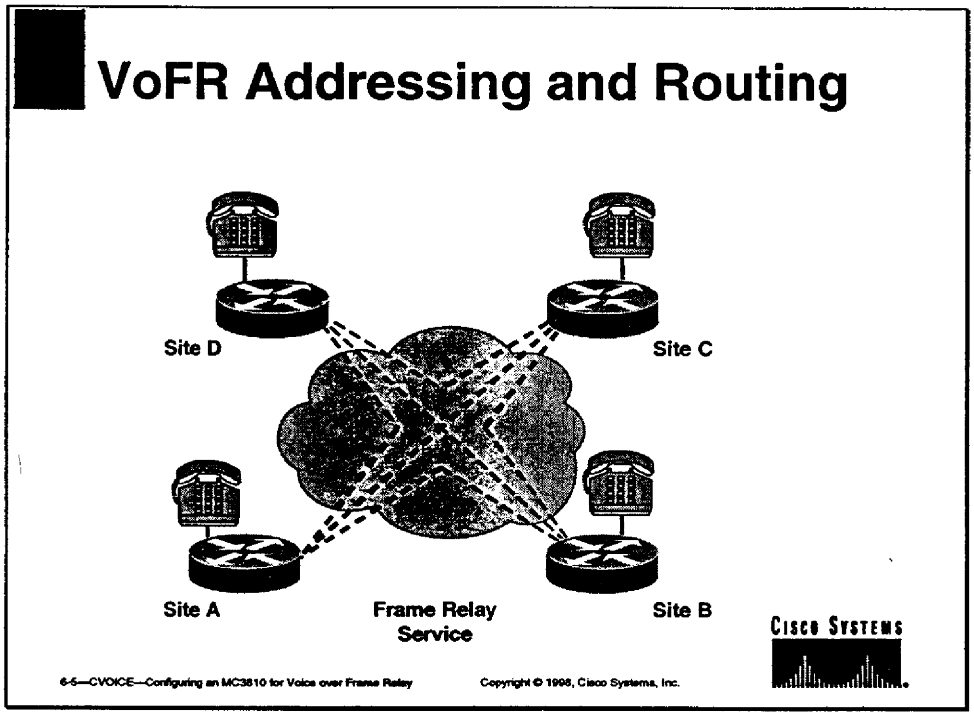
VoFR Call Setup Signaling

When you pick up a phone and dial a number, the following actions are defined:

- Going off-hook (a signal is sent from the telephone hardware interface to the CPU; the CPU allocates resources)
- Dialing digits (the DSP interprets digits, dial mapper interprets a number)
- Finding the route (the end-to-end call manager (EECM) finds the route, requests and completes the connection)
- DSP takes over (call is "released" to DSP)

Frame Relay then performs the four steps listed in the figure above:

1. Dialed digits translated to the PVC
2. Call setup frame sent
3. Confirm packet (i.e., call accepted, call rejected, or congestion on the link)
4. Telephone rings (assuming call accepted packet is returned)



VoFR Addressing and Routing

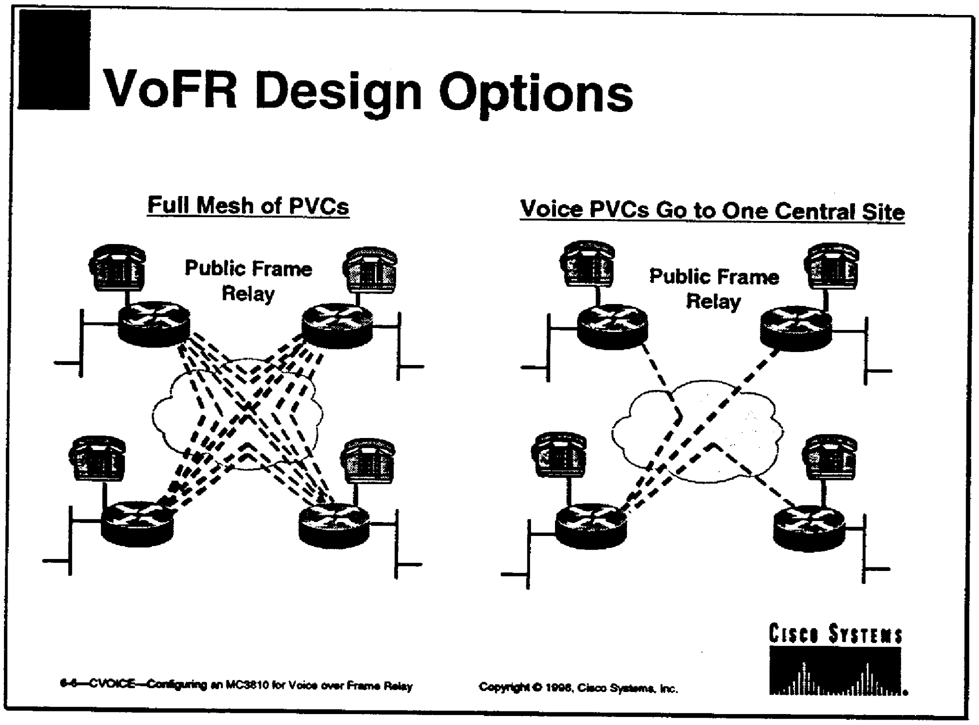
Address mapping of the E.164 numbers will be handled through static tables. Dialed digits will be mapped to specific PVCs.

Voice routing is dependent upon which routing protocol is chosen to establish PVCs and the hardware used in the Frame Relay network.

Routing can be based on bandwidth limits, hops, delay, or some combination, but most routing implementations are based on maximizing bandwidth utilization.

PVC endpoints are determined by the customer.

Note Multiple calls (plus data) can exist over one PVC.



VoFR Design Options

In the graphic on the left, we see a full mesh of voice and data PVCs. This design option minimizes the number of network transit hops and maximizes the ability to establish different qualities of service. A network designed in this fashion minimizes delay and improves voice quality, but this design option represents the highest public network cost.

To reduce costs, both data and voice segments can be configured to use the same PVC (graphic on the right), which reduces the number of PVCs required. (Remember: Most Frame Relay providers charge on the number of PVCs used.)

In the design example on the right, the central site switch reroutes voice calls. This network design has the potential problem of creating a transit hop when voice needs to go from one remote office to another remote office. However, it avoids the compression-decompression common when using a tandem PBX.

Note Separate voice and data PVCs maximize quality of service. Combining voice and data on one PVC–minimizes recurring costs. Or, use some combination.

VoFR Quality of Service

- **Frame fragmentation**

  Use of small frame sizes reduces delay and delay variation

  Shorter frames emulate fixed-size ATM cells

- **Prioritization**

  Voice frames sent before data frames

- **Committed information rate (CIR)**

  Ensures no discard of voice frames

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6-7—CVOICE—Configuring an MC3810 for Voice over Frame Relay

VoFR Quality of Service

Frame fragmentation reduces the maximum outbound frame size and controls delay. By segmenting longer frames, voice packets do not experience excessive delay that degrades voice quality. Voice frames may be compressed down to 8 kbps of bandwidth (though frame fragmentation bytes and frame headers do add some overhead). Frame fragmentation reduces voice latency by about 60 percent across the WAN.

- Basic compression induces about 40 milliseconds delay.
- There is little or no degradation in quality.

Note When designing PVCs across the WAN, PVC pair throughput, port rates, and segment sizes should be set to match the speed of the slowest link in the network.

Prioritization sends time sensitive voice frames before data frames.

Each virtual circuit is assigned a minimum service threshold, called the committed information rate (CIR). A user can transmit traffic at a rate exceeding the CIR, but excess traffic might be discarded in the event of congestion.

Backward explicit congestion notification (BECN) informs the source device that the network is experiencing congestion.

Forward explicit congestion notification (FECN) informs the next device that the network is experiencing congestion.

Burst size in cells is the number of jittery/clumped cells that can arrive and not violate the provider's cell delay variation tolerance (CDVT), which is usually measured in microseconds.

VoFR Minimizing Delay and Delay Variation

- Small frame sizes
- FRF.12—data frame segmentation
- Voice frames prioritized over data frames
- Proper use of CIR

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6-8—CVOICE—Configuring an MC3810 for Voice over Frame Relay

VoFR Minimizing Delay and Delay Variation

Frame Relay has a number of mechanisms to minimize delay and delay variation on a network.

The presence of long data frames on a low-speed Frame Relay link can cause unacceptable delays for time-sensitive voice frames. To minimize the problem of long data frames, some vendors implement smaller frame sizes to help reduce delay and delay variation.

FRF.12 is an industry standard for products from different vendors to be able to interoperate.

Methods for prioritizing voice frames over data frames also help reduce delay and delay variation. Prioritization and the use of smaller frame sizes are vendor specific implementations.

To ensure voice quality, the CIR on each PVC should be set to ensure that voice frames are not discarded.

VoFR Summary

- **Frame Relay is common in many areas**
- **The service can be very cost-effective**
- **Frame Relay is an interface specification but there is room for innovation**
- **Future trends**

  Switched virtual circuits (SVCs)

  Quality of service

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6-8—CVOICE—Configuring an MC3810 for Voice over Frame Relay

VoFR Summary

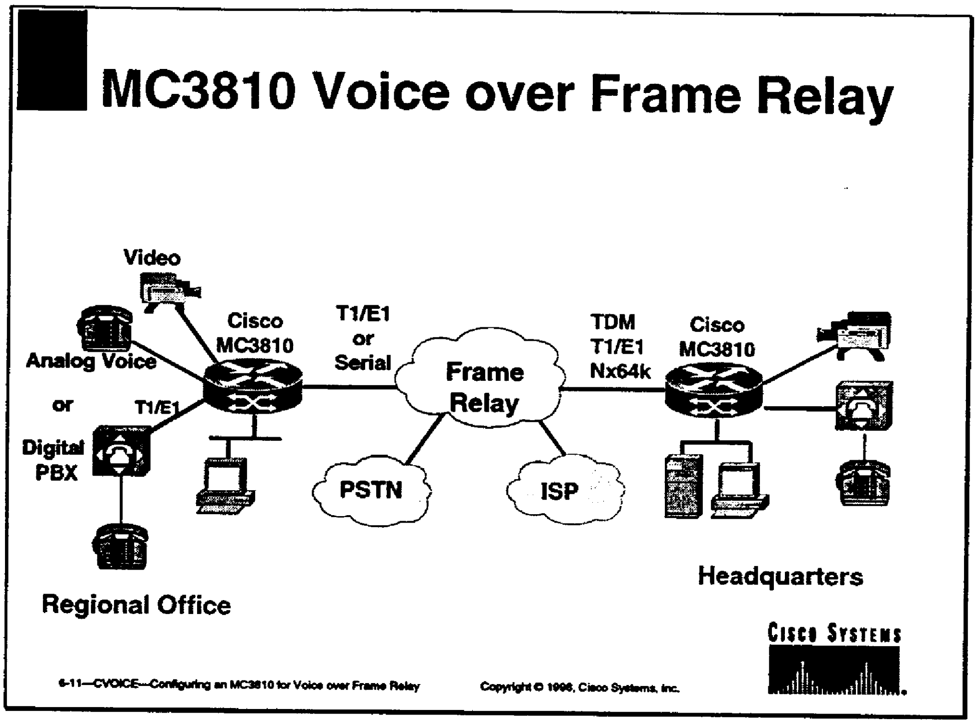
Frame Relay should be considered as a design option for the following reasons:

- Frame Relay is quite common and inexpensive in most parts of the world.
- Frame Relay CPE supports voice now.
- Many branch-to-headquarters access networks have been implemented:
  - Delivering significant cost savings
  - Using proprietary fragmentation, signaling, compression
- VoFR standards provide for better voice support:
  - FRF.12 frame fragmentation
  - FRF.11 voice signaling carried in Frame Relay frames
- CSA-CELP (G.729) is becoming the de facto standard for voice compression in Frame Relay.

Frame Relay is an interface specification, whereas ATM and TCP/IP are more architectural specifications. As an interface specification, Frame Relay will likely be used solely as a transport mechanism. However, many vendors have implemented proprietary Voice over Frame Relay solutions and validate the Voice over Frame Relay market independently.

Future Frame Relay networks will provide SVC signaling for call setup, and may also allow Frame Relay DTEs to request a quality of service (QoS) for a call. QoS will enhance the quality of Voice over Frame Relay in the future. With regard to ongoing efforts to improve Frame Relay functionality, Cisco in particular has enhanced the local management interface (LMI), and the congestion notification methods between Cisco Frame Relay switches and routers.

Packetization
Do Not wait for a 1500 byte Frame to Fill before sending



MC3810 Voice over Frame Relay Introduction

The Cisco MC3810 integrates LAN, synchronous data, voice, 384k video, and fax traffic for transport over a public or private Frame Relay network. The MC3810 optimizes network bandwidth by multiplexing voice and data on the same circuit or physical interface.

The MC3810 has a single Ethernet port and two serial ports that support speeds up to 2 Mbps. The MC3810 supports both digital and analog voice connections:

- The analog voice configuration can provide up to six compressed voice channels.
- The digital voice option houses a single digital voice access port (T1/E1) that offers up to 24 compressed voice channels.
- If voice is passed via TDM channels, up to 30 channels are available. Combinations of compressed and PCM voice are also possible.

The MC3810 implements ITU G.729 and G.729a CSA-CELP to deliver up to 24 channels of toll-quality compressed voice at 8 kbps utilizing the latest in DSP technology. Compressed voice is transmitted through packets and transported over a trunk device.

The MC3810 can connect to the following types of telephone systems:

- Analog telephone set via 2-wire connections
- Analog PBX via 2- or 4-wire interfaces
- Key system via 2- or 4-wire connections
- Digital PBX via T1/E1 (up to 24 compressed voice channels)

MC3810 Frame Segmentation

Frame segmentation reduces the maximum outbound frame size and controls delay. By segmenting longer frames, voice packets do not experience excessive delay that degrades voice quality.

A voice channel on the MC3810 may be compressed down to 8 kbps of bandwidth, though frame fragmentation bytes and frame headers add some overhead. Because the MC3810 primary market position is as a voice/data router, voice latency is of critical importance.

The 8-kbps G.729 and G.729a data algorithms reduce voice latency sensitivity by about 60 percent across the WAN. The basic compression causes about a 40-millisecond delay with little or no degradation in quality.

Segmentation is related to bandwidth, and throughput components are based on voice frame composition. A voice frame is made up of the following elements:

- 2 bytes DLCI
- 2 bytes voice/fax header (FRF.12)
- 1 byte proprietary header
- 30 bytes CSA-CELP payload
- 2 bytes CRC

The resulting frame size is therefore 37 bytes total.

Voice over Frame Relay requires bandwidth totaling 11 kbps based on the following breakdowns:

- 30 bytes payload + 7 bytes overhead = 37
- * 8 kbps = 10.13 kbps
- Rounding upward, the total recommended size becomes 11 kbps.

Note Cisco Engineering uses 11 kbps as the rounded number for the segmentation algorithm.

If any voice is being carried across the WAN using Frame Relay, all frames and cells should be segmented to ensure that voice traffic will not be adversely delayed by large volumes of (nonvoice) data frames/cells.

Note When designing PVCs across the WAN, set the PVC pair throughput, port rates, and segment sizes to match the speed of the slowest link in the network.

Cisco MC3810 Standards Support and Frame Relay Support

Frame Relay Forum FRF.1	User-Network Interface
Frame Relay Forum FRF 3.1	Multiprotocol Encapsulation
Frame Relay Forum FRF.5	Frame Relay/ATM Network Interworking
Frame Relay Forum FRF.9	Data Compression over Frame Relay
ANSI T1.606 and ITU-T I.233.1	Frame Relay Bearer Service Description
ANSI T1.618 and ITU-T Q.922,	Data Transfer Format
ANSI T1.617 Annex D and ITU-T Q.933 Annex A	DSS1 Signaling Specification
IETF RFC-1293	Inverse ARP
IETF RFC 1315	Frame Relay DTE MIB
IETF RFC-1406	T1/E1 MIB
IETF RFC-1490	Multiprotocol Encapsulation over Frame Relay

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6-12—CVOICE—Configuring an MC3810 for Voice over Frame Relay

MC3810 Standards Support and Frame Relay Support

The following Frame Relay Forum specifications are supported:

- FRF.1 User-Network Interface
- FRF.3.1 Multiprotocol Encapsulation (RFC 1490)
- FRF.5 Frame Relay/ATM Network Interworking
- FRF.9 Data Compression over Frame Relay
- FRF.12 Frame Fragmentation

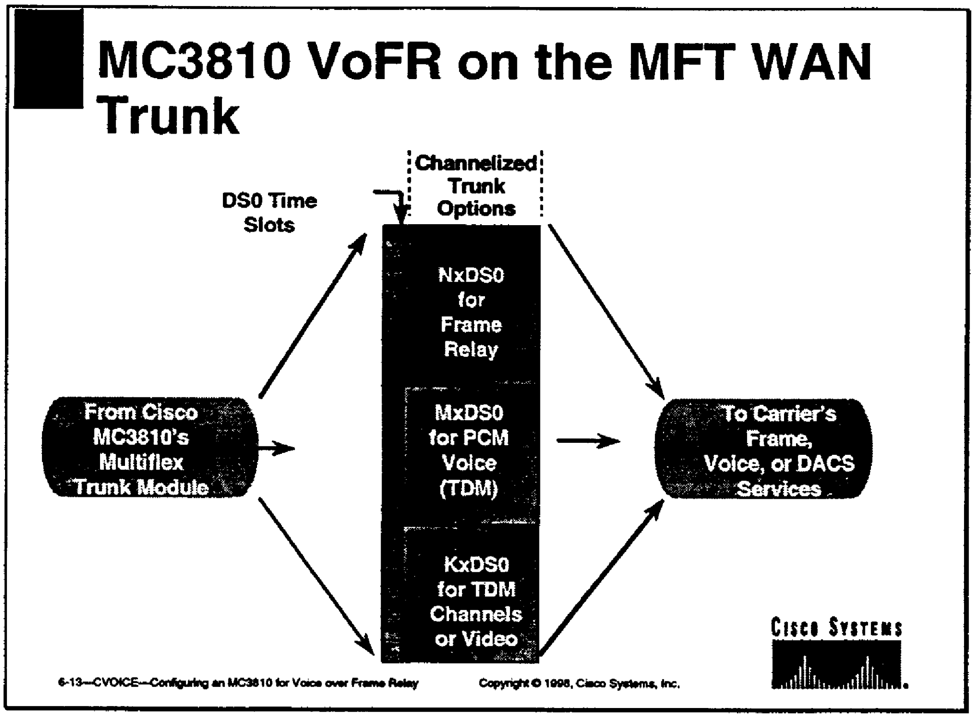
What about FRF.11? (FRF Standard for Voice over Frame Relay)

Cisco currently does NOT support FRF.11. Our frames are one byte different than the standard. The extra byte enables Cisco to do tandem switching. In addition, our bits are different from the standard voice G.729: Samples are 21 bytes, and frames are 26 bytes long, including header and CRC.

What about FRF.12? (FRF Standard for Fragmentation for VoFR)

Cisco is compliant with the March 1997 version of FRF.12. FRF.12 is still in the ratification process. Cisco will be FRF.12 compliant when it becomes final.

MC3810 Interfaces for Frame Relay



MC3810 VoFR on the MFT WAN Trunk

The Cisco MC3810 data interfaces can be configured to support several standard connections. Analog telephone interfaces can also be configured to support several standard connections to either a PSTN and PBX or to a standard telephone handset.

The Universal Input/Output (UIO) serial port supports connectivity to a digital carrier service at a wide variety of clock rates. There are two UIO serial ports, serial/0 and serial/1. Only serial/0 can receive timing (clock) and distribute it to serial/1. Consequently, if a serial port is used as the network trunk port, serial/0 should be used.

A standard Cisco 5-in-1 cable with a 60-pin connector is required to adapt the interface to the electrical interfaces. The user is required to supply the appropriate CSU/DSU.

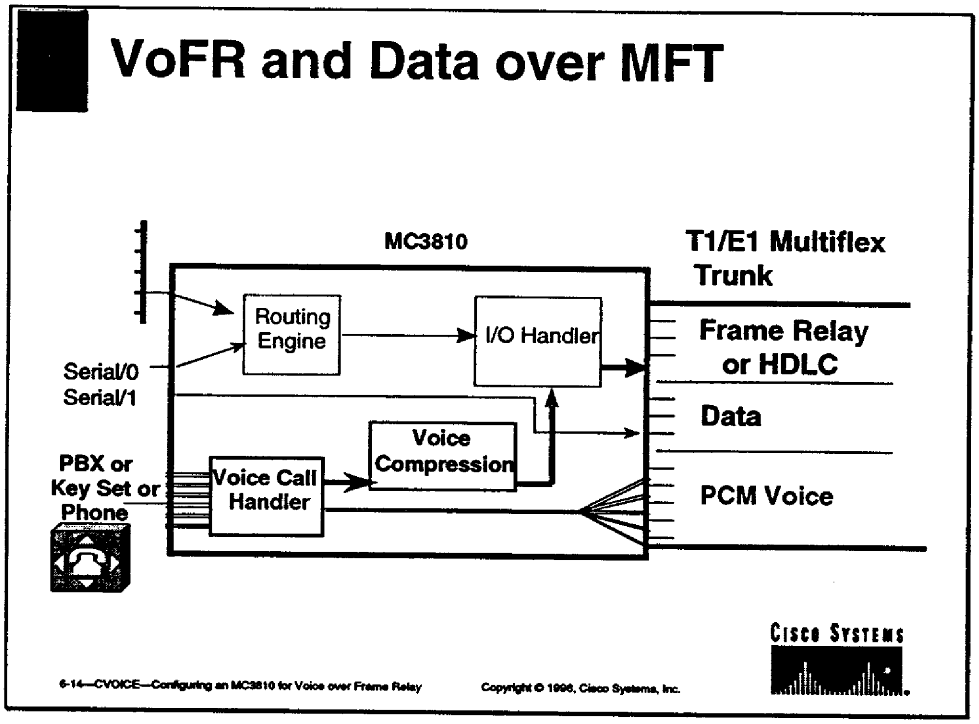
The Multiflex Trunk (MFT) provides support for either ANSI T1.403 (T1) or ITU G.703 (E1). Serial port 2 is used as the trunk interface for many applications on the Cisco MC3810. To serve as the trunk interface, serial port 2 is associated with the MFT and also provides a built-in CSU/DSU.

Note Serial ports can be configured as both physical and logical entities. For example: If an IP address is assigned to S0, subinterfaces cannot be assigned. If a function is being assigned (i.e., Voice over Frame Relay encapsulation), then multiple point-to-point subinterfaces (S0.1, S0.2, etc.) can be assigned individual IP addresses and DLCIs.

The channelized trunk option utilizes the multiservice features of the MFT. The first N time slots are reserved for Frame Relay or HDLC trunk services. Packetized, compressed voice and data are carried within this band.

Trunk Examples for Voice over Frame Relay

The voice capabilities of the MC3810 are closely tied to the trunk options. The following pages show some different options as they relate to Voice over Frame Relay.



VoFR and Data over Multiflex Trunk

The figure illustrates the voice handling options of the MC3810 when utilizing the MFT.

The following is a summary of how voice flows through the MFT interface when the channel is configured for a TDM cross-connect and the calls are configured for CSA-CELP:

Step 1 Voice calls are routed first through the voice call handler.

Step 2 Calls are routed directly to the T1/E1 trunk, bypassing the voice compression process.

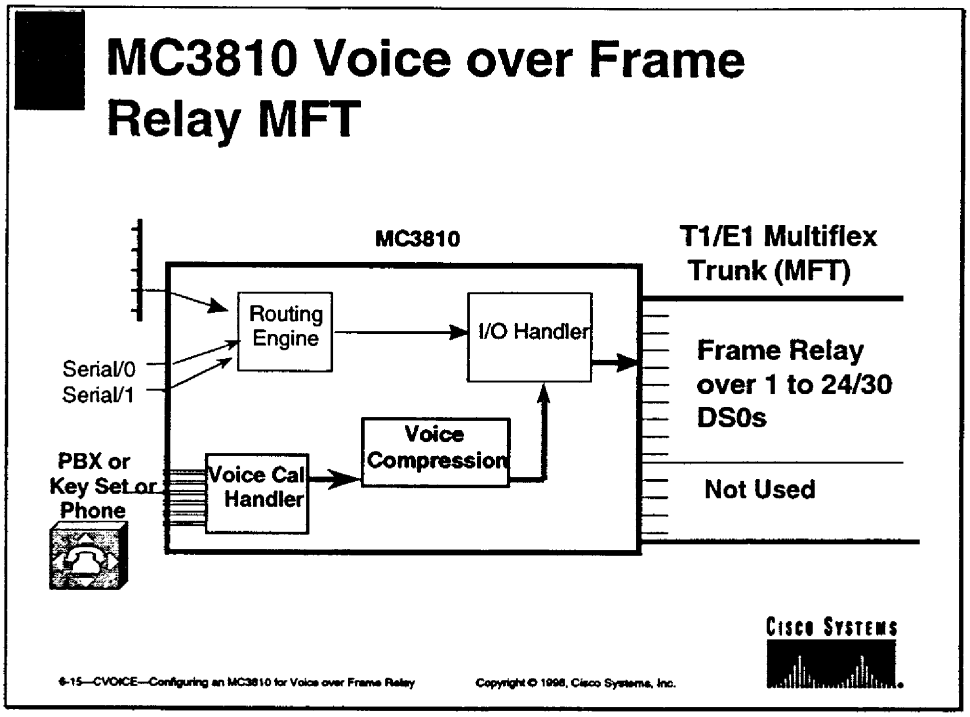
Step 3 Within its DACS network, the carrier peels off the PCM channels and routes them to the PSTN.

Step 4 A bit stream is passed to the voice compression subsystem.

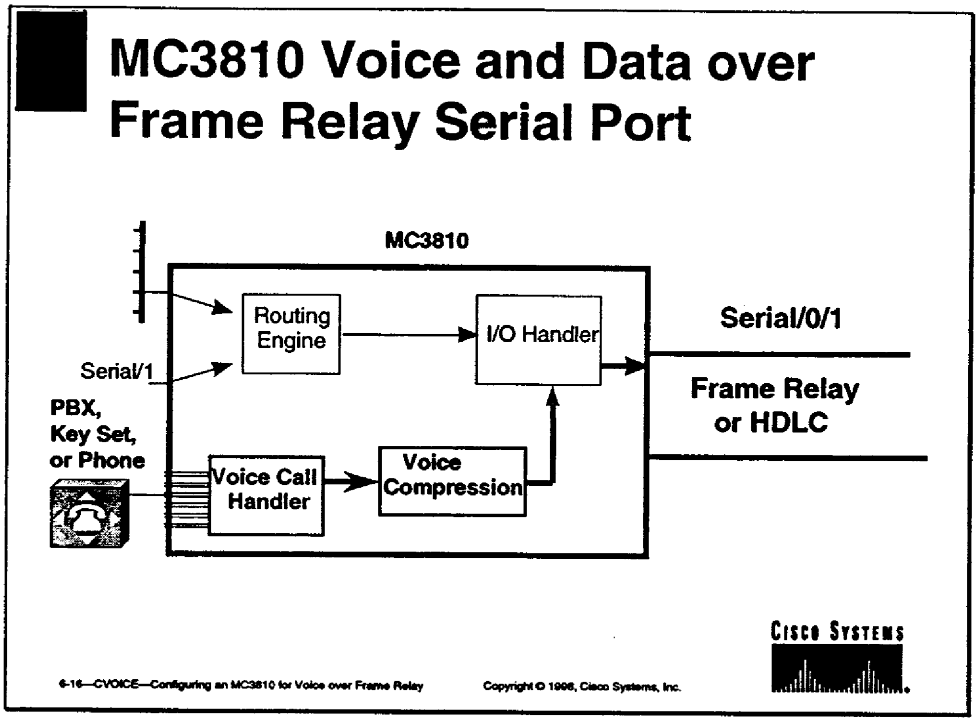
Step 5 Bit stream is then passed to the I/O handler.

Step 6 The I/O ensures that voice is encapsulated in Frame Relay.

Step 7 Encapsulated voice over Frame Relay is then passed to the trunk.



The multiflex configuration in the figure above shows voice running over Frame Relay through a partially or entirely dedicated T1/E1 trunk.



MC3810 Voice and Data over Frame Relay Serial Port

The figure above illustrates data and voice flows in the MC3810 when using the serial port as the network interface.

All calls are routed to the voice compression engine. PCM voice is not available in this configuration.

Note When using serial port 0 or 1 as the trunk, features such as voice pass-through and TDM pass-through are not available.

MC3810 VoFR Configuration Commands

```
router(config-if)#encapsulation frame-relay
```

- Sets the encapsulation mode

```
router(config-if)#frame-relay traffic-shaping
```

- Turns on traffic shaping on the interface

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MC3810 VoFR Configuration Commands

The **router(config-if)#encapsulation frame-relay** command turns on Frame Relay traffic shaping on the interface. Note that Frame Relay traffic shaping should be enabled on the interface if sending voice and data traffic over a single Frame Relay PVC over a public Frame Relay network. If sending voice and data traffic over a back-to-back Frame Relay configuration, or on a private Frame Relay network, traffic shaping is not required. Also, use Frame Relay traffic shaping only; do not use generic traffic shaping.

The **router(config-if)#frame-relay traffic-shaping** interface configuration command is used to enable both traffic shaping and per-virtual circuit queuing for all PVCs on a Frame Relay interface. To disable traffic shaping and per-virtual circuit queuing, use the **no** form of this command.

For virtual circuits for which no specific traffic shaping or queuing parameters are specified, a set of default values is used. The default queuing is performed on a first-come, first-served basis.

Note The standard Cisco IOS™ form of this command also supports Frame Relay SVCs. The Cisco MC3810 does not support Frame Relay SVCs.

VoFR Commands (cont.)

router(config-if)#frame-relay interface-dlci *dlci* [broadcast] [ietf | cisco] [voice-encap *size*]

- **Configure the Frame Relay DLCI to support FRF.12 data segmentation for voice encapsulation**

router(config-dlci)#class *map-class-name*

- **Associate the map class with the DLCI**

router(config-dlci)#exit

- **Exit the DLCI configuration mode**

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6-19—CVOICE—Configuring an MC3810 for Voice over Frame Relay

Use the **router(config-if)#frame-relay interface-dlci** *dlci* **[broadcast] [ietf | cisco] [voice-encap** *size*] command to configure the Frame Relay DLCI to support FRF.12 data segmentation for voice encapsulation. Use this command for subinterfaces on a router or access server. Use of the command on an interface, rather than on a subinterface, prevents the device from forwarding packets intended for that DLCI.

Subinterfaces are logical interfaces associated with a physical interface. You must specify the interface and subinterface before you can use this command to assign any DLCIs and any encapsulation or broadcast options.

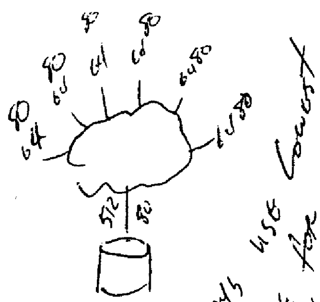
This command is required for all point-to-point subinterfaces; it is also required for multipoint subinterfaces for which dynamic address resolution is enabled. It is not required for multipoint subinterfaces configured with static address mappings.

frame-relay interface-dlci Command

Dlci	DLCI number to be used on the specified subinterface.
Broadcast	Send broadcasts on this DLCI.
ietf	(Optional) Encapsulation type: Internet Engineering Task Force (IETF) Frame Relay encapsulation.
Cisco	(Optional) Encapsulation type: Cisco Frame Relay encapsulation.
voice-encap size	(Optional) Specifies that FRF.12 segmentation will be used to support Voice over Frame Relay. For the **voice-encap** option on the Cisco MC3810, set the data segmentation size based on the port access rate. size denotes the data segmentation size. The valid range is from 8 to 1600. You must configure the voice encapsulation option to support voice traffic. The table below gives recommended data segmentation sizes.

Data Segmentation Sizes

Port Access Rate	Recommended Data Segmentation Size
64 kbps	80 bytes
128 kbps	160 bytes
256 kbps	320 bytes
512 kbps	640 bytes
1536 kbps (full T1)	1600 bytes
2048 kbps (full E1)	1600 bytes



The data segmentation size is based for back-to-back Frame Relay. If sending traffic through an IGX™switch with standard Frame Relay, add an extra 15 bytes to the recommended data segmentation size.

Note When configuring the voice encapsulation data segmentation, use the slower PVC circuit rate of either the local or remote device to calculate which data segmentation size to use. If you configure a data segmentation size too high for either the local or remote device, the PVC circuit rate will become throttled because the slower device cannot handle the larger data segmentation size. For example, if the PVC circuit rate at the local device is 512 kbps and the circuit rate of the remote device is 256 kbps, configure the data segmentation size based on the slower 256 kbps circuit rate.

Use the **router(config-dlci)#class** *map-class-name* command to associate the map class with the DLCI.

Use the **router(config-dlci)#exit** command to exit the DLCI configuration mode.

VoFR Commands (cont.)

router(config-if)#map-class frame-relay *map-class-name*

- **In (config-if) mode, configure the Frame Relay map class**

router(config-map-class)#frame-relay bc { in | out } *bits*

- **Configure the incoming and/or outgoing burst size**

router(config-map-class)#frame-relay cir { in | out } *bps*

- **Configure the incoming and/or outgoing CIR for the PVC**

router(config-map-class)#frame-relay adaptive-shaping becn

- **Configure the adaptive traffic rate adjustment to support BECN**

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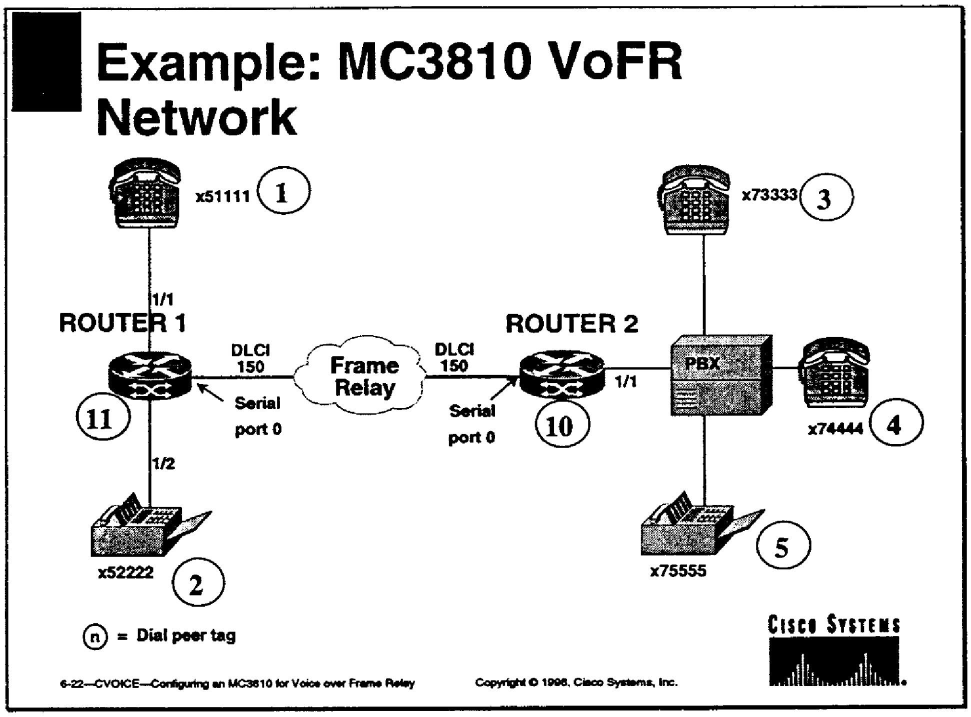
Use the **router(config-if)#map-class frame-relay** *map-class-name* command in interface configuration mode to configure the Frame Relay map class.

Use the **router(config-map-class)#frame-relay bc {in | out}** *bits* command to configure the incoming and/or outgoing burst size. Configure the bits value to a minimum of 1000 for the voice traffic.

Use the **router(config-map-class)#frame-relay cir {in | out}** *bps* command to configure the incoming and/or outgoing CIR for the PVC.

Use the **router(config-map-class)#frame-relay adaptive-shaping becn** command to configure the adaptive traffic rate adjustment to support BECN.

Use the **router(config-map-class)#exit** command to exit map-class configuration mode.



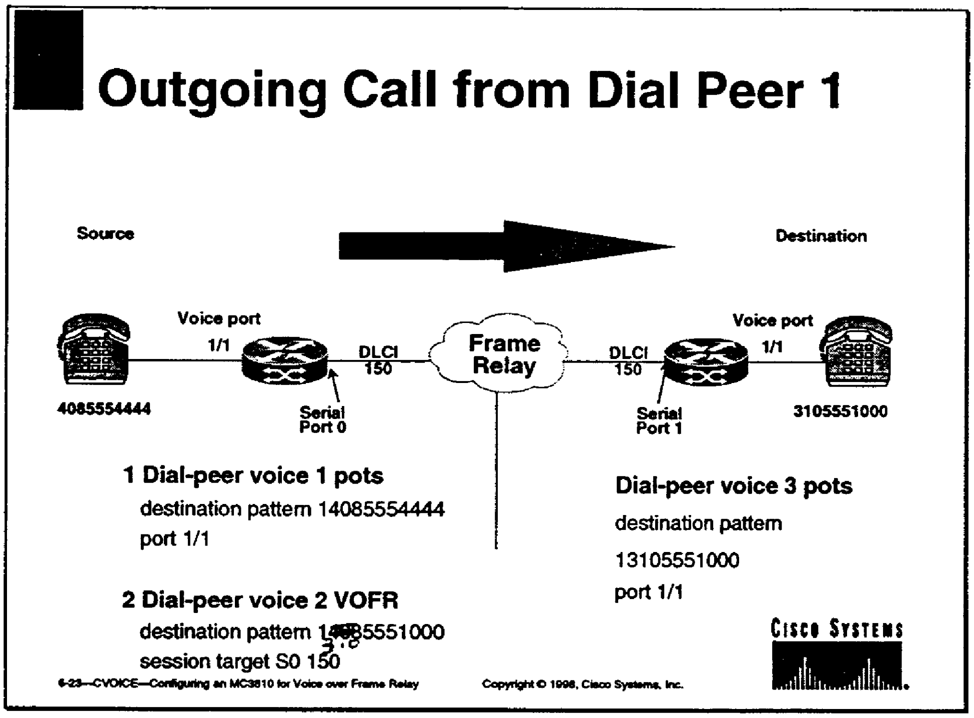
Example: Voice over Frame Relay Network

The figure shows a diagram of a small voice network in which router 1 connects a small sales branch office to the main office through router 2. Only two devices in the sales branch office need to be established as dial peers: a basic telephone and a fax machine. Router 2 is the primary gateway to the main office and is connected to the company's PBX. Three devices need to be established as dial peers in the main office, all of which are basic telephones connected to the PBX.

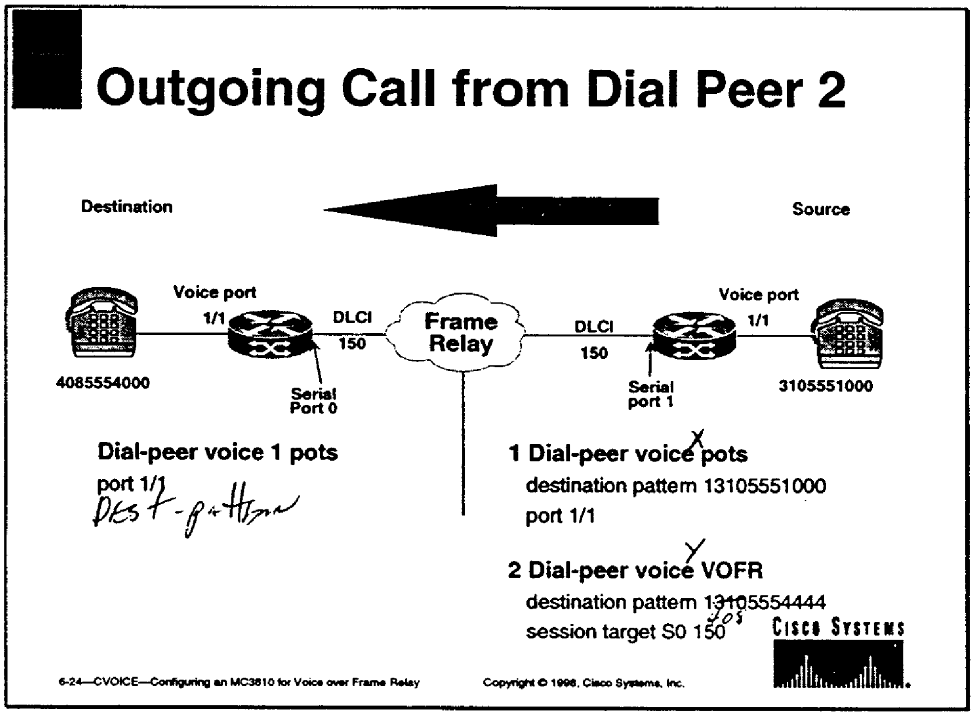
The following peer configuration table is for the example illustrated in the following figure.

Configuration Table for Voice over Frame Relay Network

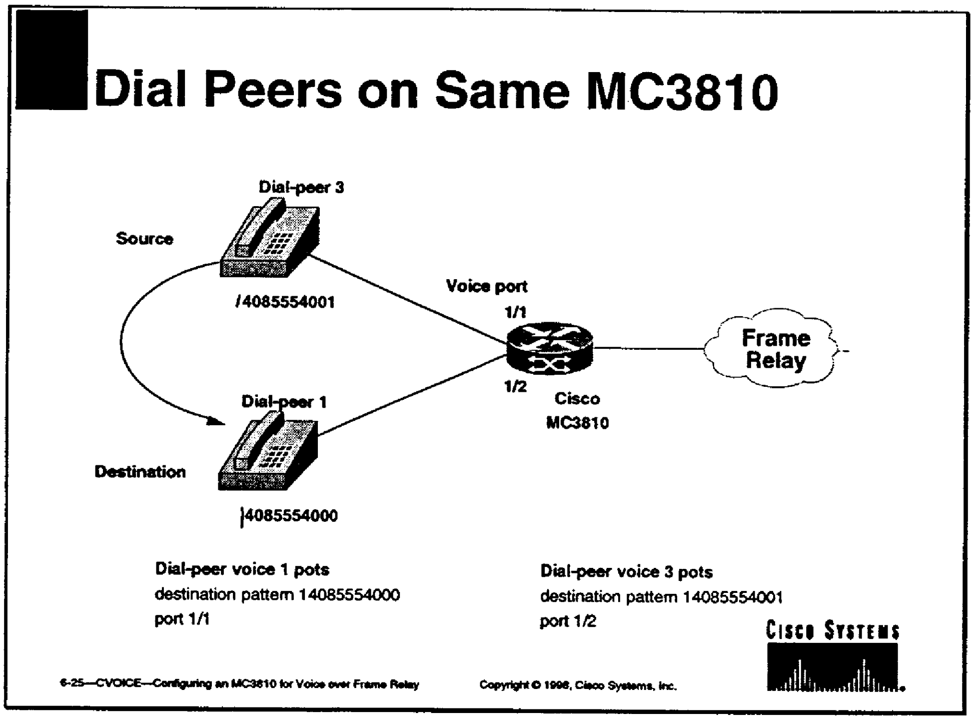
Dial Peer	Ext	Prefix	Dest-Pattern	Type	Voice Port	Session Target
Router 1						
1	51111		13107651111	POTS	1/1	
2	52222		13107652222	POTS	1/2	
10			1310767....	vofr		S0 150
Router 2						
11			1310765....	vofr		S0 150
3	73333	7	1310767....	POTS	1/1	
4	74444	7	1310767....	POTS	1/1	
5	75555	7	1310767....	POTS	1/1	



Outgoing Call from Dial Peer 1

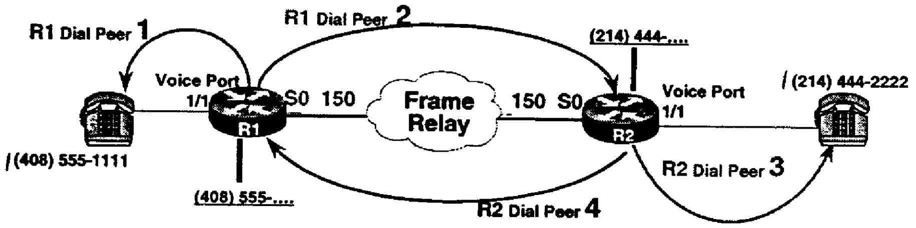


Outgoing Call from Dial Peer 2



Communication between Dial Peers Sharing the Same Router

VoFR Session Target and Destination Pattern



Cisco IOS Configuration on R1

```
dial-peer voice 1 pots
destination-pattern 14085551111
port 1/1

dial-peer voice 2 vofr
destination-pattern 1214444....
session target Serial0 150
```

Cisco IOS Configuration on R2

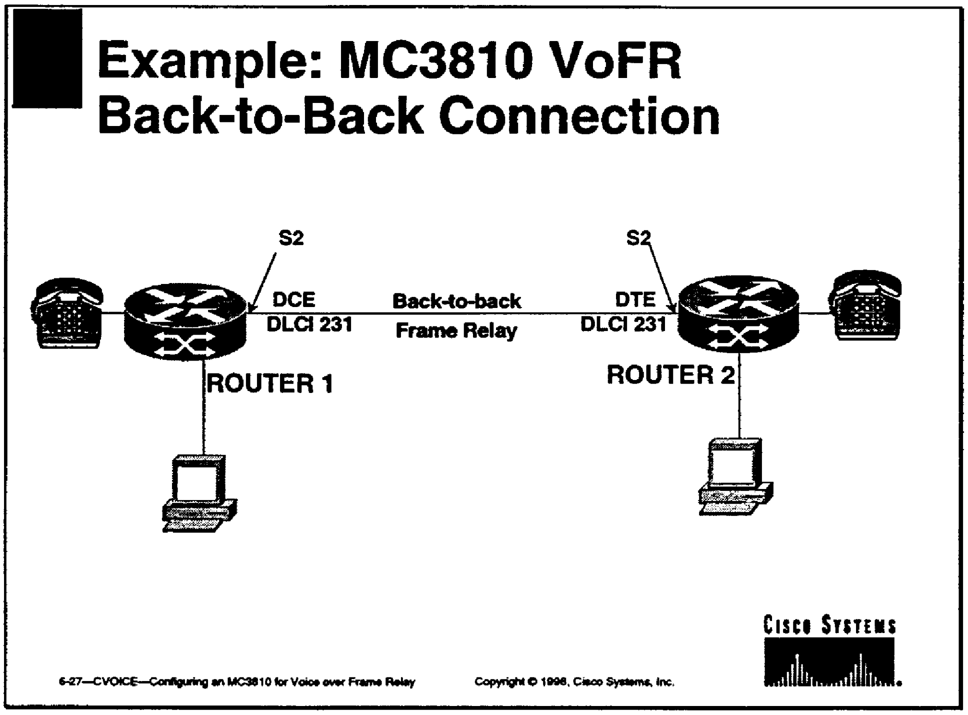
```
dial-peer voice 3 pots
destination-pattern 12144442222
port 1/1

dial-peer voice 4 vofr
destination-pattern 14085551111
session target Serial0 150
```

CISCO SYSTEMS

6-26—CVOICE—Configuring an MC3810 for Voice over Frame Relay

VoFR Session Target and Destination Pattern

The figure above illustrates the necessary dial peer configurations to place an end-to-end VoFR call.

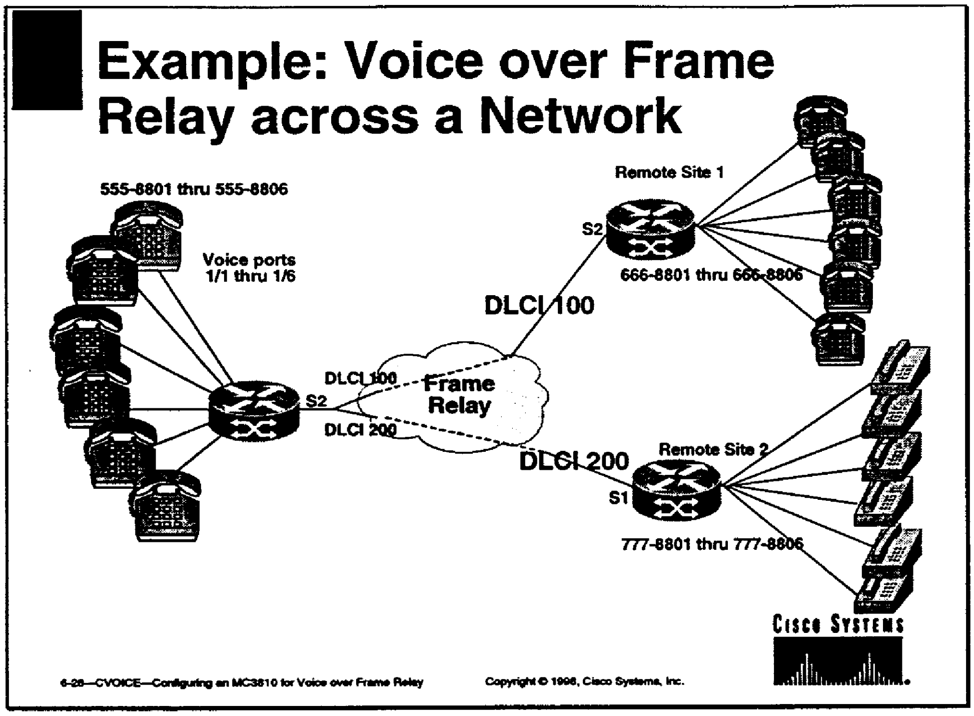


Example: MC3810 VoFR Back-to-Back Connection

The figure is a configuration example for two Cisco MC3810 concentrators configured back-to-back, with Voice over Frame Relay configured for both concentrators. The following table lists the commands required to configure the Cisco MC3810 concentrators in this example.

Frame Relay Traffic Shaping

Configuration for Cisco MC3810 ROUTER 1	Configuration for Cisco MC3810 ROUTER 2
no service pad	interface Ethernet0
no service udp-small-servers	ip address 10.1.20.1 255.0.0.0
no service tcp-small-servers	router rip
hostname Frame-top	redistribute connected
frame-relay switching	network 12.0.0.0
interface Ethernet0	interface Serial0
ip address 10.1.10.1 255.0.0.0	ip address 12.0.0.1 255.0.0.0
interface Serial0	encapsulation frame-relay
ip address 12.0.0.3 255.0.0.0	bandwidth 2000000
encapsulation frame-relay	frame-relay class fr1
no fair-queue	frame-relay map ip 12.0.0.3 231 broadcast
clock rate 2000000	frame-relay interface-dlci 231 voice-encap 1600
frame-relay class frs1	map-class frame-relay fr1
frame-relay map ip 12.0.0.1 231 broadcast	frame-relay adaptive-shaping becn
frame-relay interface-dlci 231 voice-encap 1600	frame-relay cir 64000
frame-relay intf-type dce	frame-relay bc 1000
map-class frame-relay fr1	map-class frame-relay fr2
frame-relay adaptive-shaping becn	frame-relay adaptive-shaping becn
frame-relay cir 64000	frame-relay cir 128000
for Voice frame-relay bc 1000 ALWAYS At least 1000	frame-relay bc 1000
map-class frame-relay fr2	voice-port 1/1
frame-relay adaptive-shaping becn	dial-peer voice 1 pots
frame-relay cir 128000	destination-pattern 20.
frame-relay bc 1000	port 1/1
	dial-peer voice 101 vofr
router rip	destination-pattern 1.
redistribute connected	session target Serial0 231
network 12.0.0.0	end
voice-port 1/1	
dial-peer voice 1 pots	
destination-pattern 10	
port 1/1	
dial-peer voice 101 vofr	
destination-pattern 2	
session target Serial0 231	
end	



Example: Voice over Frame Relay across a Network

The figure above represents a VoFR network. The following table lists the configurations required to configure the Cisco MC3810s in this example.

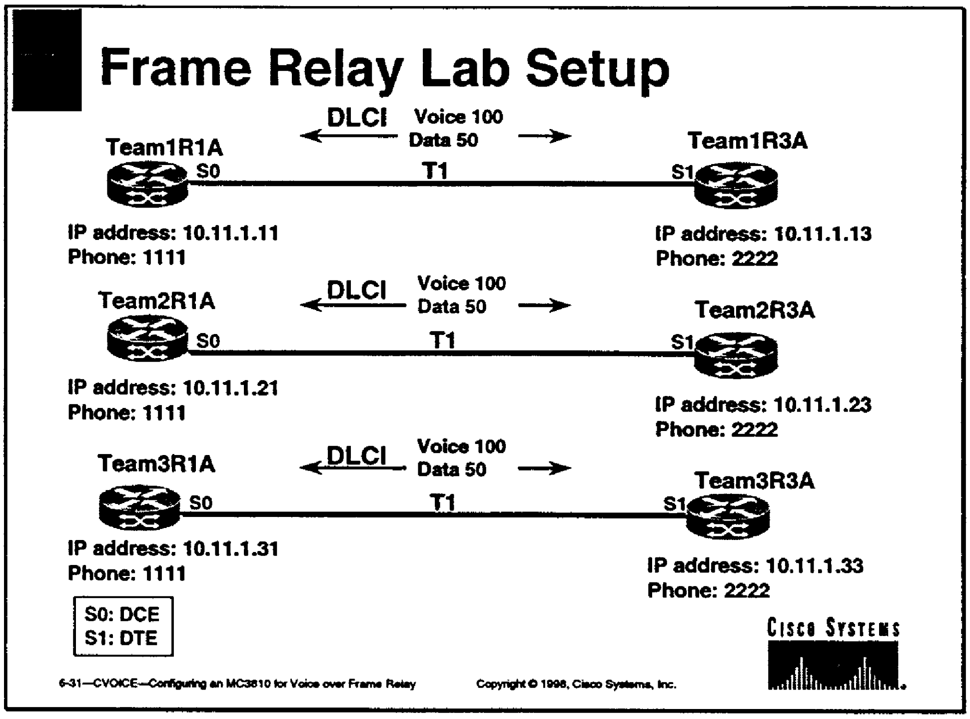
Central Site Router	Remote Site 1 Configuration	Remote Site 2 Configuration
hostname central controller T1 0 framing esf linecode b8zs channel-group 1 timeslots 1-24 speed 64 interface Ethernet0 ip address 172.22.124.66 255.255.255.0 interface Serial2 ip address 223.223.224.229 255.255.255.0 encapsulation frame-relay no fair-queue frame-relay traffic-shaping frame-relay interface-dlci 100 voice-encap 80 class fr1 frame-relay interface-dlci 200 voice-encap 160 class fr2 router igrp 1 network 172.22.0.0 network 223.223.224.0 no ip classless map-class frame-relay fr1 frame-relay adaptive-shaping becn frame-relay cir 64000 frame-relay bc 1000 map-class frame-relay fr2 frame-relay adaptive-shaping becn frame-relay cir 128000 frame-relay bc 1000 line con 0 exec-timeout 0 0 line aux 0 line vty 0 voice-port 1/1 voice-port 1/2 voice-port 1/3 voice-port 1/4 voice-port 1/5 voice-port 1/6 dial-peer voice 1 pots destination-pattern 5558801 port 1/1 dial-peer voice 2 pots destination-pattern 5558802 port 1/2 dial-peer voice 3 pots destination-pattern 5558803 port 1/3 dial-peer voice 4 pots destination-pattern 5558804 port 1/4 dial-peer voice 5 pots destination-pattern 5558805 port 1/5 dial-peer voice 6 pots destination-pattern 5558806 port 1/6 dial-peer voice 2000 vofr destination-pattern 666.... session target Serial2 100 dial-peer voice 2001 vofr destination-pattern 777.... session target Serial2 200 end	hostname remote1 controller T1 0 framing esf linecode b8zs channel-group 1 timeslots 1 speed 64 interface Ethernet0 ip address 172.22.125.66 255.255.255.0 interface Serial2 ip address 223.223.224.227 255.255.255.0 encapsulation frame-relay no fair-queue frame-relay traffic-shaping frame-relay interface-dlci 100 voice-encap 80 class fr1 router igrp 1 network 172.22.0.0 network 223.223.224.0 no ip classless map-class frame-relay fr1 frame-relay cir 64000 frame-relay bc 1000 frame-relay adaptive-shaping becn line con 0 exec-timeout 0 0 line aux 0 line vty 0 voice-port 1/1 voice-port 1/2 voice-port 1/3 voice-port 1/4 voice-port 1/5 voice-port 1/6 dial-peer voice 1 pots destination-pattern 6668801 port 1/1 dial-peer voice 2 pots destination-pattern 6668802 port 1/2 dial-peer voice 3 pots destination-pattern 6668803 port 1/3 dial-peer voice 4 pots destination-pattern 6668804 port 1/4 dial-peer voice 5 pots destination-pattern 6668805 port 1/5 dial-peer voice 6 pots destination-pattern 6668806 port 1/6 dial-peer voice 7 vofr destination-pattern 777.... session-target Serial2 100 dial-peer voice 2000 vofr destination-pattern 555.... session target Serial2 100 end	hostname remote2 interface Ethernet0 ip address 172.22.126.66 255.255.255.0 interface Serial1 ip address 223.223.224.226 255.255.255.0 encapsulation frame-relay no fair-queue frame-relay traffic-shaping frame-relay interface-dlci 200 voice-encap 160 class fr1 clock rate 128000 router igrp 1 network 172.22.0.0 network 223.223.224.0 no ip classless map-class frame-relay fr1 frame-relay cir 128000 frame-relay bc 1000 frame-relay adaptive-shaping becn line con 0 exec-timeout 0 0 line aux 0 line vty 0 voice-port 1/1 voice-port 1/2 voice-port 1/3 voice-port 1/4 voice-port 1/5 voice-port 1/6 dial-peer voice 1 pots destination-pattern 7778801 port 1/1 dial-peer voice 2 pots destination-pattern 7778802 port 1/2 dial-peer voice 3 pots destination-pattern 7778803 port 1/3 dial-peer voice 4 pots destination-pattern 7778804 port 1/4 dial-peer voice 5 pots destination-pattern 7778805 port 1/5 dial-peer voice 6 pots destination-pattern 7778806 port 1/6 dial-peer voice 2000 vofr destination-pattern 555.... session target Serial2 200 dial-peer voice 2001 vofr destination-pattern 666.... session target Serial2 200 end

Lab Exercise: Establishing a VoFR Call between Two MC3810 Routers

Use the following laboratory exercise to practice what you have learned in this lesson.

Lab Setup

Use the following information to prepare for this exercise. The figure below illustrates the general back-to-back Voice over Frame Relay lab configuration.



Scenario

Configure the Cisco IOS software to make an FXS Voice over Frame Relay phone call between two MC3810 stations wired back-to-back. Refer to the Voice over Frame Relay lab diagram in Appendix A for complete details as to the device to which you are to make the call.

Exercise: Configuring the Frame Relay WAN Interface for VoFR

The configuration commands that follow are commands specific to Team1R1A, but can be used as a template of the steps you must follow to configure your own router. Use the Voice over Frame Relay lab diagram in Appendix A and perform the following steps to configure your router.

Step 1 To begin the configuration, establish a console link to the MC3810, using an ASCII terminal or a laptop set to terminal mode (i.e., Hyper Term or Procomm).

Step 2 Begin by entering the privileged EXEC mode. Enter the **enable** Cisco IOS command.

Step 3 Enter global configuration mode with the **config t**(erminal) command.

Step 4 Set the host name of your router. Use the **hostname Team_R_A** command. (Refer to the voice over Frame Relay lab network design diagram in Appendix A for the proper information and name your host. The host name is a code to describe your **Team** number, **R**(outer) number, and whether your router has an **A**nalog or **D**igital chassis).

Step 5 Set the router for RIP with the **router rip** command.

Step 6 Set the router for IP routing capability and set the network address with the **network 10.0.0.0** command.

Step 7 Exit router configuration mode with the **exit** command.

Step 8 To configure the network clock base rate for universal I/O serial ports 0 and 1 to 64k, enter the **network-clock base-rate 64k** command.

Step 9 Enter interface configuration mode and prepare to configure interface serial 1 with the **interface serial 1** command. (Refer to the lab setup instructions for your serial interface number.)

Step 10 Set an IP address on your interface with the **ip address 10.10.1.13 255.255.255.0** command. (Refer to the lab setup instructions for your IP address.)

Step 11 Set the serial interface for Frame Relay encapsulation with the **encapsulation frame-relay** command.

Step 12 Set the bandwidth for your interface to 64 kbps with the **bandwidth 64** command.

Step 13 Because the end-to-end lab connection is back-to-back, you do not need keepalive. Disable keepalive with the **no keepalive** command.

Step 14 If your router is the DCE, you must set the clock rate. To set the clock rate to 64000 bps, enter **clock rate 64000.** If you have a DTE router, do not set the clock rate, instead, go to step 16. (Refer to the Voice over Frame Relay lab diagram in Appendix A to learn if your router is a DCE or DTE router.)

Step 15 When running the line at high speeds and long distances, use the **dce-terminal-timing enable** interface configuration command to prevent shifting of the data with respect to the clock. Enter the **dce-terminal-timing-enable** command.

Step 16 To ensure the serial line is active, perform the following sequence of commands:

- At the interface config prompt: **shut**
- Wait until a message appears, informing you that serial 1 is down.
- Then at the interface config prompt: **no shut**

Step 17 Disable weighted fair queuing on the interface with the **no fair-queue** command.

Step 18 Enter the **frame-relay map ip 10.10.1.11 50** command to associate your data traffic with DLCI 50. (Refer to the Frame Relay Lab diagram in Appendix A for your correct IP address.)

Step 19 Enter the **frame-relay interface-dlci 100 voice-encap 80** command to specify DLCI 100 as your voice circuit. Also, specify the segmentation at 80.

Step 20 Enter the **class labexercise** command to specify a map class name, **labexercise,** associated with the virtual circuit.

Step 21 Activate your serial interface with the **no shut** command.

Step 22 Exit interface configuration mode with the **exit** command.

Step 23 Enter map class configuration mode and specify the lab exercise map class so you can define a quality of service (QoS) on your virtual circuit. Use the **map-class frame-relay labexercise** command.

Step 24 Enable Frame Relay BECN on your lab exercise map class with the **frame-relay adaptive-shaping becn** command.

Step 25 Configure the incoming or outgoing CIR on the virtual circuit to 64k with the **frame-relay cir 64000** map class configuration command.

Step 26 Configure the incoming or outgoing committed burst size (Bc) on the virtual circuit to 64k with the **frame-relay bc 64000** map class configuration command.

Step 27 Exit the map class configuration mode with the **exit** command.

Step 28 Return to the privileged EXEC mode with the **end** command.

Step 29 Save your configurations with the **copy run start** command.

Exercise: Configuring Your Voice Ports

To configure your voice ports, perform the following steps from the privileged EXEC mode:

Step 1 Enter global configuration mode with the **config t** command.

Step 2 Prepare to configure voice port 1/1 by entering **voice-port 1/1** to enter voice port configuration mode.

Step 3 Adjust the size of the echo cancel to 16 ms on your voice port with the **echo-cancel coverage 16** command.

Step 4 Enable silence suppression and VAD with the **vad** command.

Step 5 Set the voice compression to G.729, 8k CSA-CELP Annex A with the **codec g729ar8** command.

Step 6 Enter **end** to exit all configuration mode.

Step 7 Enter **show voice port** to verify proper configuration.

Exercise: Configuring Your Dial Peers

To configure your dial peers, perform the following steps from the privileged EXEC mode:

Step 1 Enter global configuration mode with the **config t** command.

Step 2 Configure a POTS dial peer for your phone. To define a POTS dial peer and enter dial peer configuration mode, use the **dial-peer voice 10 pots** command.

Step 3 Configure the POTS dial peer's destination pattern with the **destination-pattern 1111** command. (Refer to the Frame Relay lab diagram in Appendix A for your POTS telephone number.)

Step 4 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer. Enter **port 1/1**.

Step 5 Exit from dial peer configuration mode with the **exit** command.

Step 6 You must now configure a VoFR dial peer. To configure a VoFR dial peer, use the **dial-peer voice 20 vofr** command to enter VoFR dial peer configuration mode.

Step 7 Configure the VoFR dial peer's destination pattern with the **destination-pattern 2222** command. (Refer to the lab setup instructions for your VoFR telephone number you will call.)

Step 8 Set the session target for the VoFR line for **session target Serial0 100**. Refer to the lab diagram in Appendix A for your VoFR session target (i.e., serial port number and DLCI).

Step 9 Exit from all configuration modes with the **end** command.

Step 10 Take turns placing telephone calls using your VoFR network.

Completion Criteria

The lab is complete when all Cisco IOS parameters have been configured and a call can be successfully completed between the two back-to-back 3810s.

Summary

Summary

- Describe VoFR software and WAN trunk interfaces of the MC3810
- List the Cisco IOS commands specific to configuring the MC3810 for VoFR
- Configure an MC3810 for VoFR using Cisco IOS software so students can connect and optimize their own VoFR enterprise branch offices

6-32—CVOICE—Configuring an MC3810 for Voice over Frame Relay

Summary (cont.)

- Configure the MC3810 WAN services, Frame Relay voice port interfaces, and dial peers using Cisco IOS software so a phone call can be successfully completed between two MC3810s

6-33—CVOICE—Configuring an MC3810 for Voice over Frame Relay

Configuring the MC3810 for Voice over ATM

Objectives

Objectives

- Describe VoATM software and WAN trunk interfaces of the MC3810
- List the Cisco IOS™ commands specific to configuring the MC3810 for VoATM
- Configure an MC3810 for VoATM using Cisco IOS software so students can connect and optimize their own VoATM enterprise branch offices

Cisco Systems

Objectives (cont.)

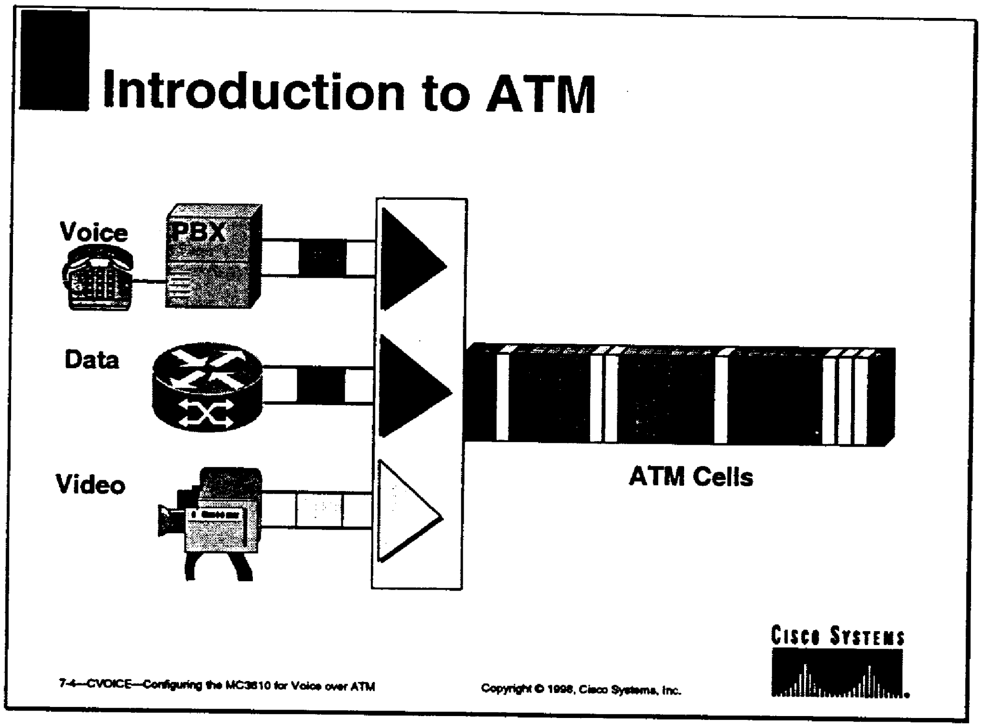
- **Configure the MC3810 WAN services, ATM voice port interfaces, and dial-peers using Cisco IOS software so a phone call can be successfully completed between two MC3810s**
- **Configure the MC3810 as a "channel bank" using the MC3810's advanced functionality**

7-3—CVOICE—Configuring the MC3810 for Voice over ATM

Cisco Systems

Introduction

Chapter 7 covers the topic of Voice over ATM. The equipment used in this chapter is the Cisco MC3810. The goal of this chapter is to teach the student voice over ATM using the Cisco IOS software.



Introduction to ATM

Asynchronous Transfer Mode (ATM) is a switching method for transmitting information in fixed length cells, based on application demand and priority.

The basic characteristics of ATM are:

- Uses small, fixed-sized cells (53 octets)
- Connection-oriented protocol
- Supports multiple service types
- Applicable to LAN and WAN traffic
- ATM virtual circuits emulate PSTN circuits
- ATM minimizes delay and delay variation

ATM Service Classes: CBR, VBR, ABR, UBR

VBR

CBR

Real-Time:

- **Variable bit rate (VBR): Compressed voice with silence suppression—Proprietary now, standards in process**
- **Constant bit rate (CBR): Circuit emulation**

UBR

ABR

Nonreal-Time:

- **Unspecified bit rate (UBR): "Best effort"**
- **Available bit rate (ABR): Best for bursty data transport (guaranteed thruput +)**

7-5—CVOICE—Configuring the MC3810 for Voice over ATM

Cisco Systems

Constant bit rate (CBR) and variable bit rate (VBR) classes have provisions for passing real-time traffic, such as voice or video conferencing, and are suitable for guaranteeing a certain level of service. CBR, in particular, allows the amount of bandwidth, end-to-end delay, and delay variation to be specified during the call setup.

Unspecified bit rate (UBR) and available bit rate (ABR) have been designed with bursty traffic in mind and are more suitable for data applications. UBR in particular makes no guarantees with the delivery of the data traffic.

Each class is given a guaranteed minimum bandwidth, which ensures deterministic behavior under load (e.g., voice will not degrade when data bursts). If additional bandwidth is available, each class can access a fair share, which allows ATM to support multiple, complex classes of service (CoS).

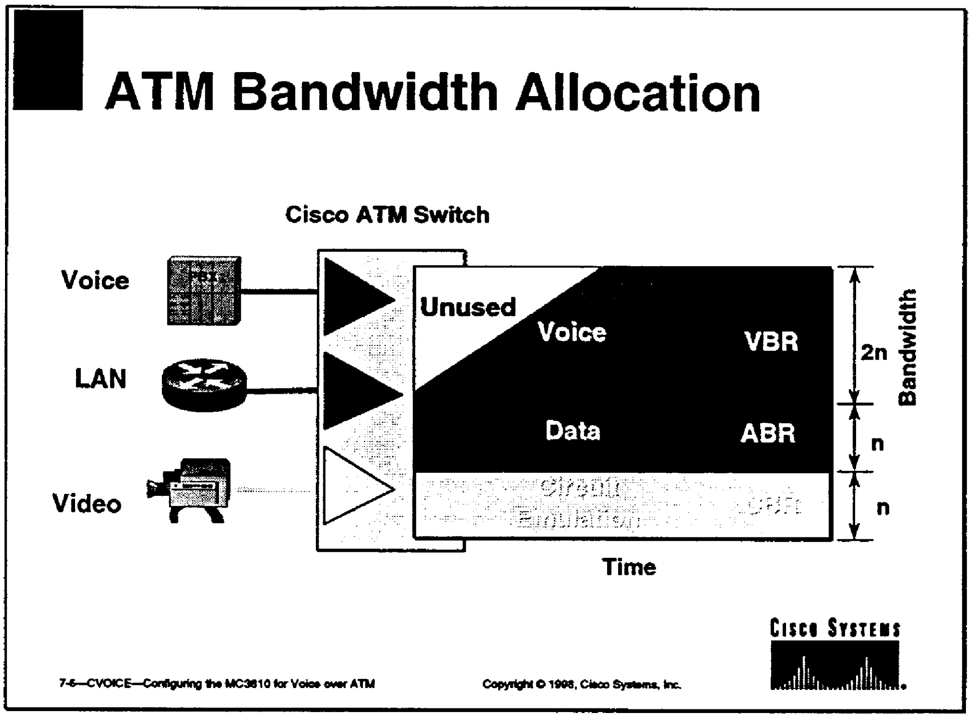
The first architectural key requirement is the necessity to implement internal dedicated network (trunk) queues for every service type that needs to be supported.

A simple example is voice versus data: Voice has stringent requires when it comes to delay behavior. The delay needs to be low and constant (to avoid starvation at the egress point). Voice tolerates a certain degree of cell loss. Data, on the other side, does not tolerate cell loss at all. Thus, upon facing overload, an architecture that handles these traffic types on dedicated queues can, with intelligent queue handling algorithms, decide that, as one cell needs to be dropped, it should be one out of the voice queue.

The more queues, the more granularity the queuing algorithms have in order to "outplay" the different service types against each other and guarantee QoS requirements even when available resource become constricted.

Only a design that allocates dedicated network queues based on QoS susbsets and handles these queues intelligently can deliver deterministic QoS in an efficient way that does not require overengineering of resources, thus delivering the promise of ATM technology.

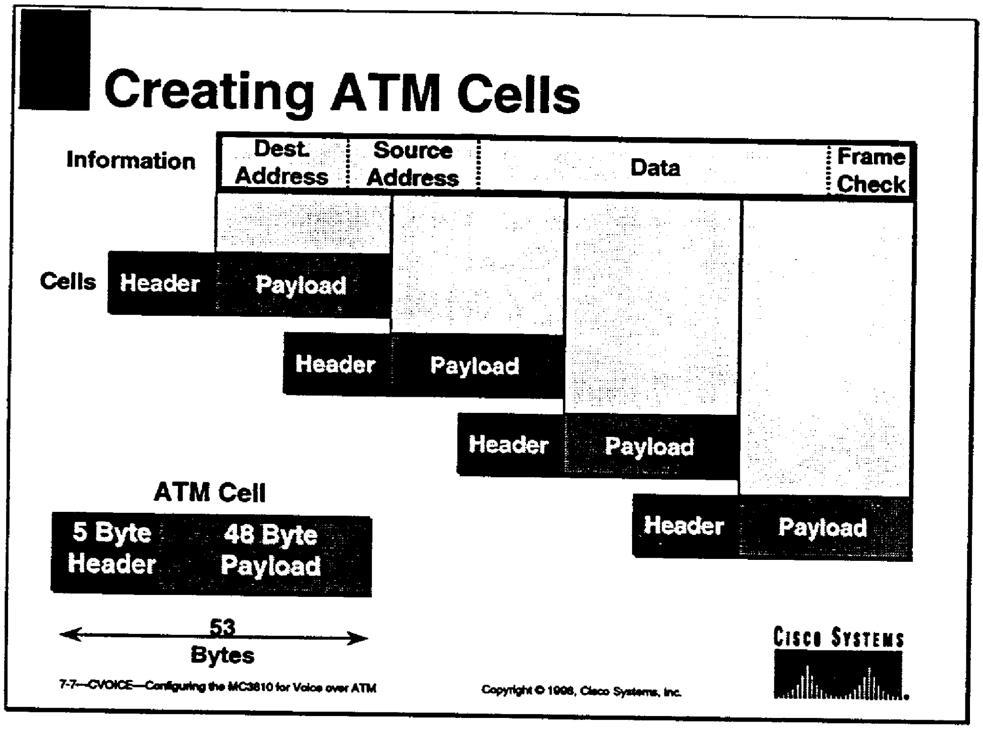
Note To be able to guarantee CoS and QoS, dedicated network queues are required for different service types.



Unlike LAN data, voice has strict requirements in terms of delay and jitter in order to remain intelligible. There is a balance between this requirement and the need to use available bandwidth as efficiently as possible.

The graphic shows we can allocate minimum amounts of bandwidth to each application. However, when one application does not need all the bandwidth, it can be used by other applications.

Cisco VoATM devices use queuing and prioritization techniques that handle bandwidth on a per-virtual circuit basis to ensure that each separate application has the bandwidth and quality of service it requires.



An ATM cell is made up of a 5-octet header and a 48-octet payload. The header identifies cells belonging to the same virtual channel (VC).

The 53 octets is a protocol data unit (PDU) of ATM, similar to a Layer 2 frame or a packet. Information is segmented into the payload for transmission across the ATM network.

VoATM—Minimizing Delay and Delay Variation

- **Quality of service (QoS)**
- **Virtual circuit (VC) queuing**
- **Small fixed-size cells**

7-8—CVOICE—Configuring the MC3810 for Voice over ATM

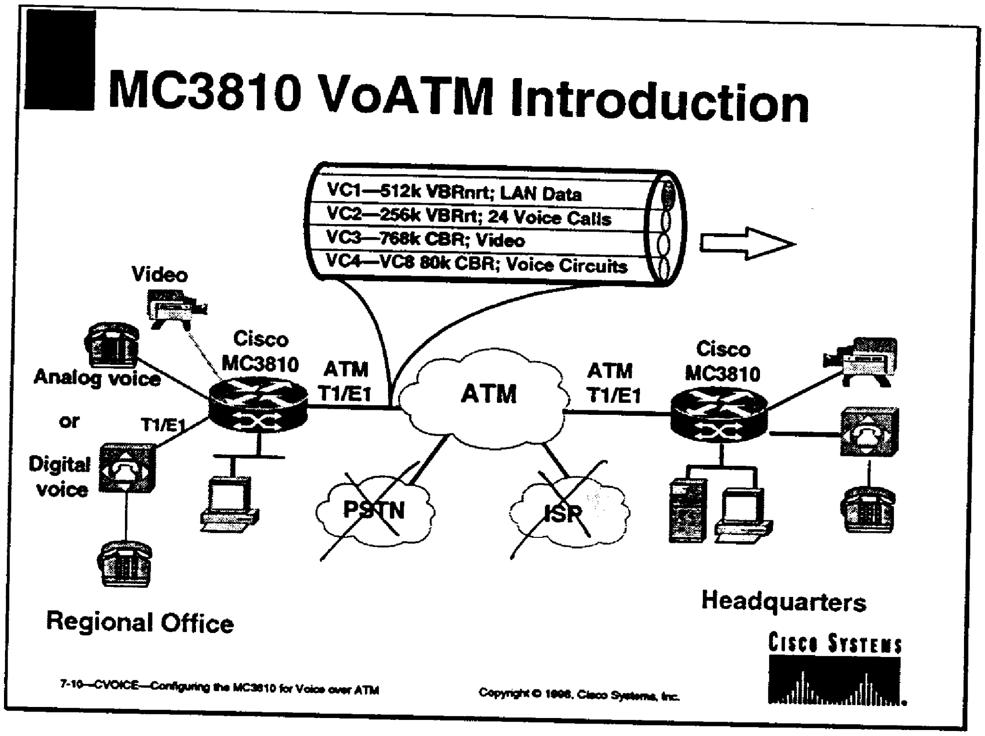
Cisco Systems

ATM has several mechanisms for controlling delay and delay variation.

The quality of service (QoS) capabilities of ATM allow for the specific request of constant bit rate traffic with bandwidth and delay variation guarantees.

Virtual circuit (VC) queues allow each traffic stream to be treated uniquely. In the case of voice traffic, priority can be given for its transmission ahead of other delay-insensitive traffic.

Small fixed-size cells reduce queuing delays and the delay variation associated with variable-sized packets. Small-sized cells are also extremely useful in reducing delay through intermediate switches.



MC3810 VoATM Introduction

A Cisco MC3810 multiservice concentrator combines data, voice/fax, and video signals and connects them to an ATM network. It can multiplex voice, video, and data onto ATM trunks running at either T1 or E1.

This section concentrates on Voice over ATM and covers analog and digital Voice over ATM.

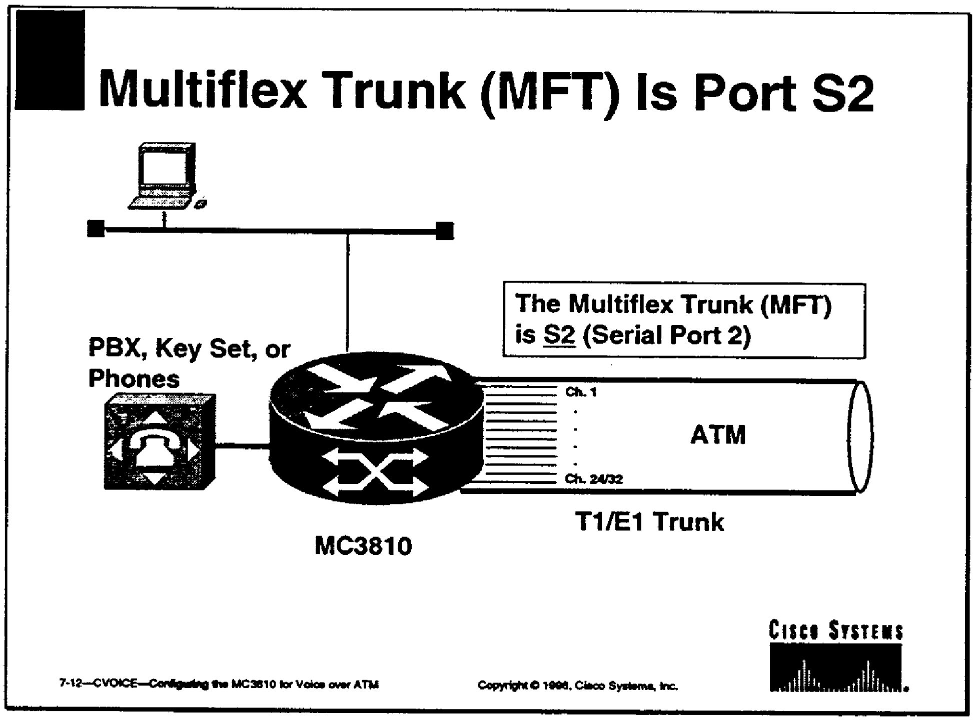
Analog Voice

In its optional analog voice configuration, the Cisco MC3810 has up to six analog voice ports. By installing specific signaling modules known as analog personality modules, you can equip these ports for the following signaling types in various combinations:

- FXS
- FXO
- E&M

Digital Voice

In its optional digital voice configuration, the Cisco MC3810 has a digital voice port that can interface with a digital PBX or channel bank to support up to 24 digital voice channels through an RJ-48 connector.



Multiflex Trunk (MFT) Is Port 2

Serial port 2 is used as the trunk interface for many applications on the Cisco MC3810. To serve as the trunk interface, serial port 2 is associated with the MFT, which provides support for either ANSI T1.403 (T1) or ITU G.703 (E1). The MFT also provides a built-in CSU/DSU. T1 0

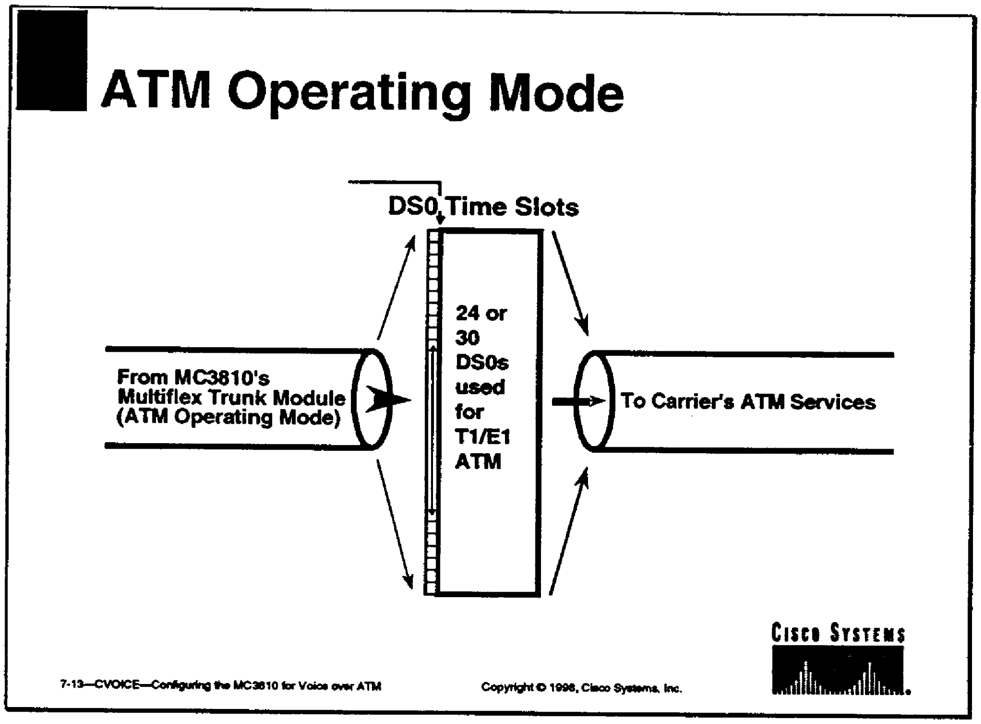
Note Controller 0 is always serial port 2.

T1 1

When configuring the T1/E1 controller settings when using the DVM, select a framing format that best suits the type of signals carried by the T1/E1. Some T1/E1 equipment can be confused by repeating data patterns, such as when all voice channels are idle, and may have difficulty maintaining frame synchronization with some frame formats. In these cases, choose a different framing format or vary the location and CAS type of voice channels, to help the external T1/E1 equipment maintain frame synchronization.

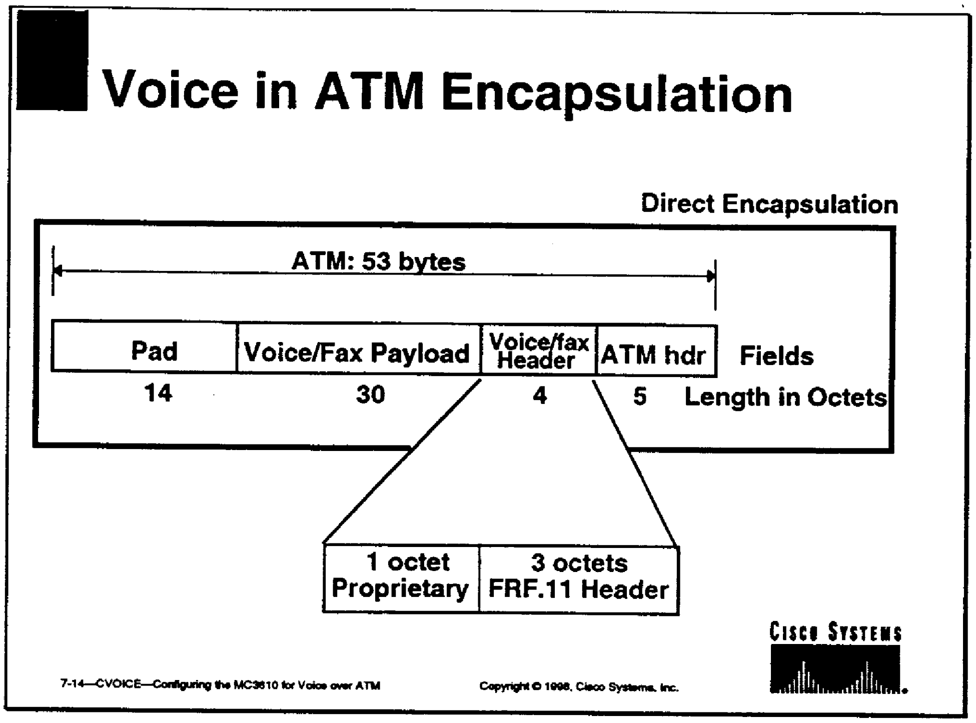
There are two operating modes for the MFT: the multiflex mode and the ATM mode.

Note Only the multiflex ATM operating mode is discussed in the following section.



ATM Operating Mode

The ATM mode utilizes the ATM functions of the Cisco MC3810. In this mode the full T1/E1 is (and must be) devoted to ATM. Voice connections are made via compressed voice using AAL5 VBR services. Similar to the multiflex mode, multiple services, including LAN and video, may be passed over the ATM trunk. PCM coded voice in AAL1 circuit emulation format is not supported in Release 2.0.

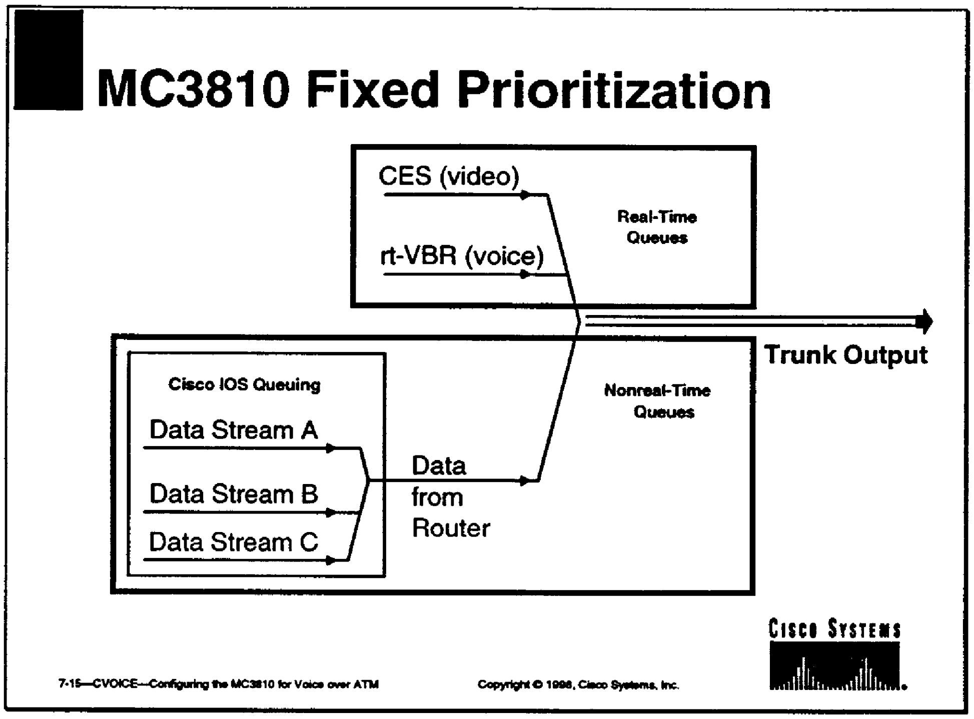


Voice in ATM Encapsulation

The ATM cell has the following components:

5 bytes ATM Header
4 bytes voice/fax header
30 bytes ACELP payload
14 bytes of padding
----
53 byte cell with (30 bytes of relevant payload, 23 bytes of overhead)

The voice/fax header comprises three octets of an FRF.11 header, plus one octet of a Cisco propriatary byte.



MC3810 Fixed Prioritization

The ability to distinguish voice and video data types, and service them before nonreal time data types, is a distinguishing feature of the Cisco MC3810. To do this function, priority queues are used on the trunk, each supporting one of two service classes.

The MC3810 supports two service classes: real time and nonreal time. The real-time class is used for voice and video, and the nonreal-time class is used for data. Real-time virtual channel queues are serviced in a round-robin fashion, and they are completely emptied before any nonreal-time channel queues are serviced. Non real-time virtual channel queues will be serviced only when there are no real-time cells in the queues. Data channel queues are also serviced in a round-robin fashion.

Non real time may be prioritized within its own domain by the core router. Using Cisco IOS features such as priority queuing and weighted fair queuing, different nonreal-time data streams may be prioritized relative to each other before they are presented to the trunk.

Assuming the Cisco IOS data streams are in priority order by alphabet, the priority of all the data streams in the figure above is:

1. CES (Video)
2. rt-VBR (voice)
3. Data A
4. Data B
5. Data C

Voice and video traffic always have top priority. There is no way to prioritize nonreal time data over real-time data. Voice always have top priority to prevent telephone users from being locked out. However, nonreal-time data streams will never be completely locked out by real-time data if the unit is correctly configured.

For example, out of an entire T1/E1 ATM trunk, if 24 compressed voice channels are present, they will use approximately 240 kbps of bandwidth (about 15 percent of a T1 and 11 percent of an E1). Even if a video service takes up a another 384 kbps, plenty of bandwidth would be available for data.

Note Guarding against data lockout: To ensure that critical data gets through, the network design should include bandwidth for data over and above the greatest amount of voice bandwidth possible. The total bandwidth includes locally terminated voice connections as well as tandem connections.

ATM Voice and Fax Bandwidth Requirements

The voice bandwidth required is derived from the inherent bandwidth of the voice compression algorithms plus the overhead required to move the framed data across the network. The table below shows the bandwidth required for voice and fax payloads using an ATM connection.

ATM Voice and Fax Bandwidth Requirements

Payload Type	Nominal Bandwidth	Required Bandwidth Using ATM
Voice	8 kbps	14.13 kbps
Fax	9.6 kbps	16.96 kbps

Two types of ATM frames are used to transport voice or fax: direct and FRF.5 encapsulation. For the purposes of calculating required bandwidth, the two are identical because, relative to the voice or fax payload, overhead is overhead, whether it is padding or FRF.5 information.

The calculation to determine voice bandwidth for a directly encapsulated ATM voice or fax cell is as follows:

5 bytes ATM header
4 bytes voice/fax header
30 bytes CSA-CELP payload
14 bytes of padding

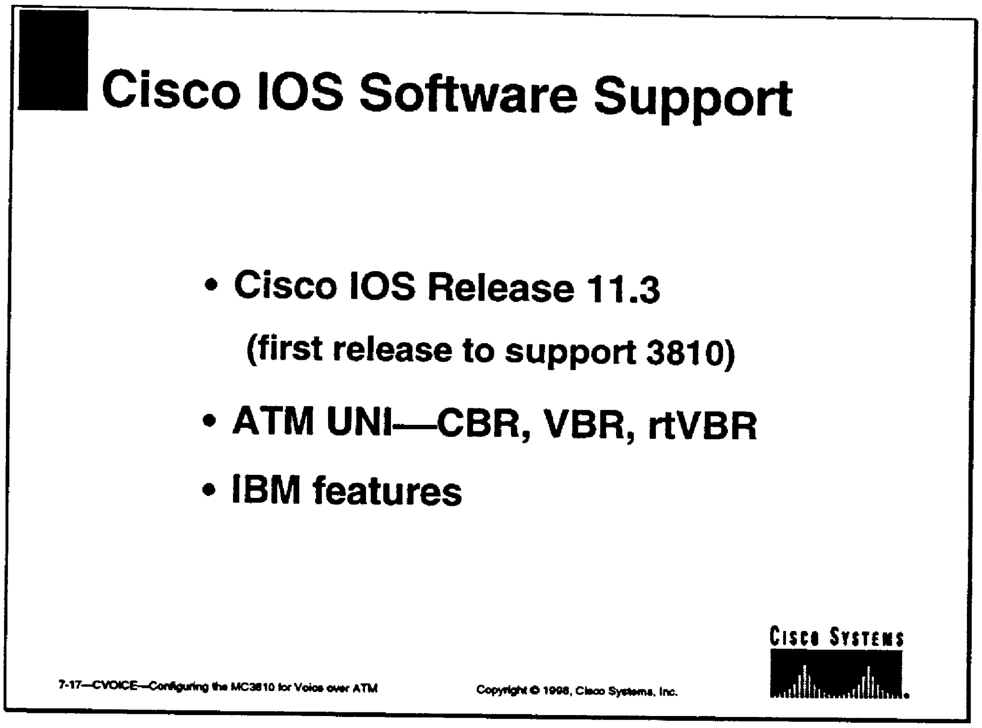
53-byte cell with (30 bytes of relevant payload, 23 bytes of overhead)
53/30 = 77 percent overhead ==> 8.0k * 1.77 = **14.13 kbps** required bandwidth

For a fax connection:

53/30 = 77 percent overhead ==> 9.6k * 1.77 = **16.96 kbps** required bandwidth

---

Note The results for an ATM frame using FRF.5 encapsulation are identical.



Cisco IOS Software Support

There are several Cisco IOS images with different core routing features. The difference between the images is support for voice or ATM. Refer to the Cisco Web site to obtain the latest feature sets and Cisco IOS images available.

Note Cisco IOS feature list details are listed in Appendix A.

The Cisco IOS image supported in a configuration with the MFT is SF381MH-11.3.1M. This Cisco IOS image includes IP/IPX/IBM/ATM with voice.

MC3810 Standards Support ATM Support

ATM Forum af-phy-0016.000	DS1 Physical Layer Specification
ATM Forum af-phy-0064.000	E-1 Physical Layer Specification
ATM Forum af-saa-0032.000	Circuit Emulation
ATM Forum af-uni-0010.002	UNI Signaling 3.1
IETF RFC1483	Multiprotocol over ATM
IETF RFC1577	IP over ATM
IETF RFC1695	AToM MIB
ANSI T1.630, ITU I.363, I.363.1	ATM AAL 1 (Constant Bit Rate)
ANSI T1.635, ITU I.363.5	ATM AAL 5 (Variable Bit Rate)

Cisco Systems

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MC3810 Standards Support ATM Support

Although the MC3810 supports other standards, the standards shown in the figure above are specifically related to its operation in the ATM arena.

Network Management Software

- Cisco command-line interface (CLI)
  All system configuration commands
- CiscoView
- CiscoWorks
- Extensive MIB support
- HTML interface

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Cisco Systems

MC3810 Cisco IOS Commands for VoATM

MFT Controller Commands

router(config)#controller { t1 | e1 } *number*

- **Identifies whether your controller is T1 or E1, and the number**

router(config-controller)#clock source { internal | line | loop-timed }

- **Identifies the controller clock source for a DS1 link**

router(config-controller)#description *line*

- **Enters a description of the controller**

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CISCO SYSTEMS

MFT Controller Commands

The commands shown in the figure above configure the MFT for T1/E1 controller settings. These commands are accessed in controller configuration mode.

At the router> prompt, enter **enable** to enable privileged EXEC command mode. Next, at the router prompt, enter **conf t** to enter global configuration mode.

Use **router(config)#controller {t1 | e1}** *number* to enter controller configuration mode. Specify whether your controller is E1 or T1, and enter the controller number. If the DVM is installed, the controller number can be either 0 or 1. If the MFT is installed, the number must be 0.

Use **router(config-controller)#clock source {internal | line | loop-timed}** to configure the controller clock source for a DS1 link. The default for the DVM is *internal*, while the default for the MFT is *line*.

The *loop-timed* parameter specifies that the T1/E1 controller will take the clock from the Rx (line) and use it for Tx. This setting decouples the controller clock from the system-wide clock set with the *network-clock-select* parameter. The loop-timed clock enables the DVM to connect to a PBX and to connect the MFT to a central office (CO) when both the PBX and the CO function as DCE clock sources. This situation assumes that the PBX also takes the clocking from the CO, thereby synchronizing the clocks on the DVM and the MFT.

Use **router(config-controller)#description** *line* to enter a description of the controller, such as the destination or its application, for the *line* value. The description for the controller interface can be up to 80 characters long.

T1 Controller Settings

router(config-controller)#cablelength short { 133 | 266 | 399 | 533 | 655 }

- **For cable lengths shorter than 655 feet**

router(config-controller)#cablelength long { gain26 | gain36 } { -15db | -22.5db | -7.5db | 0db }

- **Sets parameters if the cable is longer than 655 feet**

router(config-controller)#framing { sf | esf }

- **Sets the DS1 link framing format**

router(config-controller)#linecode { ami | b8zs }

- **Sets the line encoding format for DS1**

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T1 Controller Settings

The following Cisco IOS commands show controller settings for a T1 ATM, These commands are accessed in controller configuration mode.

The **router(config-controller)#cablelength short {133 | 266 | 399 | 533 | 655}** command configures the cable length for the T1 controller if the cable length is 655 feet or shorter.

The **router(config-controller)#cablelength long {gain26 | gain36} {-15db | -22.5db | -7.5db | 0db}** command configures the cable length if the length is longer than 655 feet.

The **router(config-controller)#framing {sf | esf}** command configures the DS1 link framing format.

Note Extended Super Frame (esf) format is *required* for ATM traffic.

The **router(config-controller)#linecode {ami | b8zs}** command configures the line encoding format for the DS1 link.

Note The b8zs setting is *required* for ATM (and Frame Relay) traffic.

After configuring these parameters for the T1 controller, enter the following commands:

Use **router(config-controller)#no shutdown** to activate the T1 controller.

Use **router(config-controller)#exit** to exit controller configuration mode.

Use **router(config)#exit** to exit configuration mode.

Use the **router#show controller T1 0** command if you want to verify your T1 controller configuration settings. (Remember: Controller 0 is always serial port 2.)

E1 Controller Settings

router(config-controller)#framing { crc4 | no-crc4 } [*name*]

- **Sets the E1 frame format**

router(config-controller)#linecode { ami | hdb3 }

- **Sets the line encoding format**

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E1 Controller Settings

To configure E1 controller settings, complete the following tasks in controller configuration mode:

Use **router(config-controller)#framing {crc4 | no-crc4}** [*name*] to configure the E1 framing format. **crc4** sets framing for CRC type 4. *name* is optional, for E1 only. If the trunk will be connected to a device in say, Ireland, enter the "Ireland" option.

Use **router(config-controller)#linecode {ami | hdb3}** to configure the line encoding format. **ami** specifies alternate mark inversion (AMI) encoding. For E1 only, **hdb3** specifies HDB3 encoding.

Note The HDB3 setting is *required* for ATM and Frame Relay traffic.

Use **router(config-controller)#no shutdown** to activate the E1 controller.

After configuring these parameters for the E1 controller, you will need to enter the following commands:

Use **router(config-controller)#exit** to exit controller configuration mode.

Use **router(config)#exit** to exit configuration mode.

Use **router#show controller E1 0** if you want to verify your E1 controller configuration settings. (Remember: Controller 0 is always serial port 2.)

Channel Associated Signaling

NEVER USE this for ATM Voice

router(config-controller)#mode cas

- **Sets CAS support for the T1/E1 link**

router(config-controller-cas)#voice-group *channel-no* **timeslots** *timeslot-list* **type**

- **Configures a list of time slots to form a CAS group for the T1/E1**

Cisco Systems

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Channel Associated Signaling (CAS)

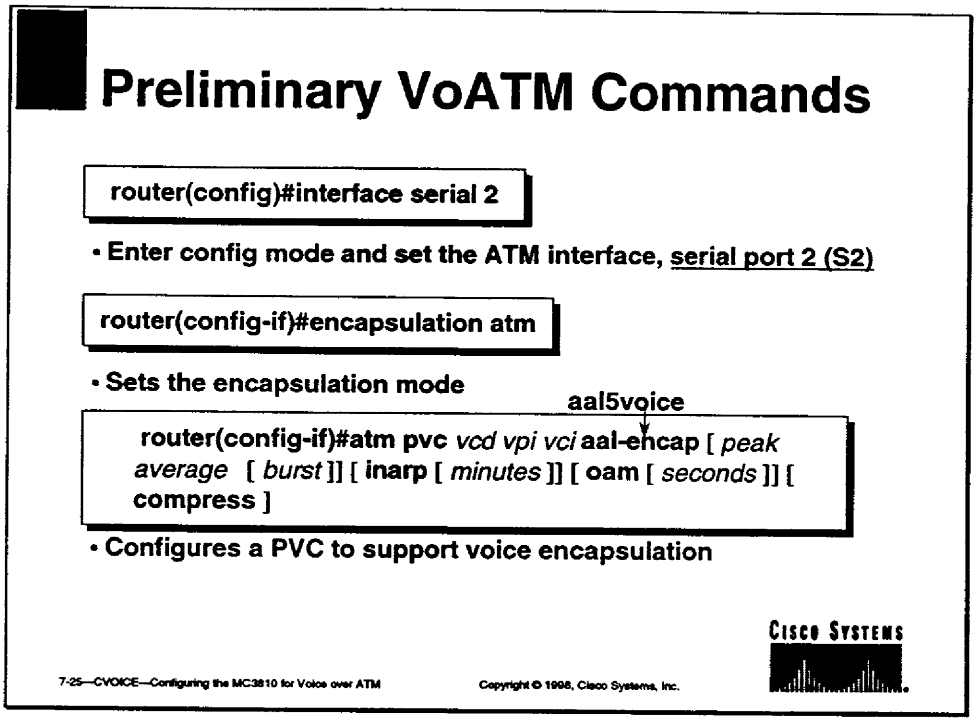
CAS extraction and insertion is used for T1/E1 voice ports or the trunk transmitting and receiving digital voice traffic from a PBX. In this case, DS0 channels can be configured to directly connect to the T1/E1 network trunk in TDM mode, or the channels can be connected to the Cisco MC3810's digital signaling processor for voice compression. During this process, the CAS bits are extracted from the defined T1/E1 channels and passed to the voice signaling handler.

Use **router(config-controller)#mode cas** to configure the T1/E1 line to support CAS.

Use **router(config-controller-cas)#voice-group** *channel-no* **timeslots** *timeslot-list* **type** and the various options for this command, **{e&m-immediate | e&m-delay | e&m-wink | e&m-melcas | fxs-ground-start | fxs-loop-start | fxs-melcas | fxo-ground-start | fxo-loop-start | fxo-melcas}**, to configures a list of time slots to form a CAS group for the T1/E1 line. Refer to the *Cisco MC3810 Multiservice Concentrator Software Command Reference Guide* for additional information regarding the various optional parameters. Repeat the *voice-group* command for each CAS group defined.

Note If you plan to configure digital voice ports, you must also configure the CAS group to assign which time slots are assigned which voice signaling protocol.

Note The "**melcas**" options are supported only on E1 and apply to the Mercury Exchange Limited (MELCAS) interpretation of the CEPT standard, used primarily in the United Kingdom.



Preliminary VoATM Commands

Voice over ATM enables a Cisco MC3810 to carry live voice traffic (for example, telephone calls and faxes) over an ATM network. Before you can configure your Cisco MC3810 to use Voice over ATM, you must:

1. Establish a working ATM network. For more information about configuring ATM, refer to the chapter in the *Cisco IOS Wide Area Networking Configuration Guide.*
2. Complete your company's dial plan.
3. Establish a working telephony network based on your company's dial plan.

The commands and procedures in this section are specific to the Cisco MC3810 for supporting Voice over ATM. To configure ATM to support voice traffic, complete the following tasks in global configuration mode:

Use **router(config)#interface serial 2** to enter interface configuration mode and configure the serial interface. Serial port 2 (i.e., trunk) is the only serial port that supports ATM traffic.

Use **router(config-if)#encapsulation atm** to set the encapsulation type to ATM on serial port 2.

Use **router(config-if)#atm pvc** *vcd* **vpi vci aal5voice** [*peak average burst*]] **[inarp** [*minutes*]] **[oam** [*seconds*]] **[compress]** to configure a PVC to support voice encapsulation. To configure the PVC to support voice traffic over ATM, you must specify **aal5voice** encapsulation.

Using the other **aal-encap** encapsulation settings, you can configure different PVCs for voice and data. Because voice and data traffic are on different PVCs, voice traffic over ATM will not impact data traffic over ATM.

Cisco IOS Parameters for Creating ATM PVCs

- vcd
- vpi
- vci
- aal-encap
- peak
- average
- burst
- inarp [*minutes*]
- oam [*seconds*]
- compress

Cisco Systems

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Cisco IOS Parameters for Creating ATM PVCs

The ***vcd*** (virtual circuit descriptor) is a unique number that identifies to the processor which VPI-VCI pair to use for a particular packet. Valid values range from 1 to the value set with the **atm max vc** command. The AIP or ATM port adapter requires this feature to manage packet transmission. The **vcd** value is not associated with the VPI-VCI pair used for the ATM network cells. The NPM has a hard coded **max vcd** value of 1023. Valid values are from 1 to 1023.

The ***vpi*** (virtual path identifier of this PVC) is an 8-bit field in the header of the ATM cell. The VPI value is unique only on a single link, not throughout the ATM network, because it has local significance only. The VPI value must match that of the switch. Valid values are from 0 to 255.

Note The arguments *vpi* and *vci* cannot both be set to 0; if one is 0, the other cannot be 0.

The ***vci*** (virtual channel identifier of this PVC) is in the range of 0 to 1 less than the maximum value set for this interface by the **atm per-vc per-vp** command. Typically, lower values of 0 to 31 are reserved for specific traffic and should not be used (for example, F4 OAM, SVC signaling, ILMI, and so on). Valid values are from 0 to 1023.

The VCI is a 16-bit field in the header of the ATM cell. The VCI value is unique only on a single link, not throughout the ATM network, because it has local significance only.

ATM Adaptation Layer (AAL) and *encapsulation* Values

The possible values for the ***aal-encap*** ATM adaptation layer (AAL) and encapsulation type are as follows:

- aal5snap—Used for Logical Link Control/Subnetwork Access Protocol (LLC/SNAP). Precedes the protocol datagram. This is the only encapsulation support for Inverse ARP.
- aal1—Used to encapsulate streaming video packets over ATM.
- aal5fratm—Used for the Frame Relay-ATM interworking function.
- aal5voice—Used for voice traffic over ATM.
- aal5mux—Used for multiplex-type virtual circuit.
- aal5nlpid—Used for High-Speed Serial Interfaces (HSSIs) that are using an ATM data service unit (ADSU) and running ATM-Data Exchange Interface (DXI).
- ilmi—Used to define the PVC for ILMI.

Calculating Peak, Average, and Burst Values for Voice

You must also configure the peak, average, and burst options for voice traffic. Configure the burst value if the PVC will be carrying bursty traffic. The peak, average, and burst values are needed so the PVC can effectively handle the bandwidth for the number of voice calls. To calculate the peak, average, and burst values for the number of voice calls, use the following calculations:

Peak value: (2 x the number of calls) x 16

Average value: (1 x the number of calls) x 16

Burst value: (4 x the number of calls)

Caution *Do not configure the* ***inarp*** *or* ***compress*** *values for a voice PVC.*

The **peak** (optional) value is the maximum rate (in kbps) at which this virtual circuit can transmit. The valid range is from 56 to 10000. If configuring Voice over ATM, you must configure the peak, average, and burst values. To calculate the peak rate for the number of voice calls, use the following calculation:

(2 x the number of calls) x 16

The **average** (optional) value is the rate (in kbps) at which this virtual circuit transmits. Valid values are platform dependent. If configuring Voice over ATM, you must configure the peak, average, and burst values. To calculate the average rate for the number of voice calls, use the following calculation:

(1 x the number of calls) x 16

The **burst** (optional) value relates to the maximum number of ATM cells the virtual circuit can transmit to the network at the peak rate of the PVC for bursty traffic. If configuring Voice over ATM, you must configure the peak, average, and burst values. To calculate the burst rate for the number of voice calls, use the following calculation:

4 x the number of calls

Additional ATM PVC Parameters

The **inarp** *minutes* (optional) (set for the aal5snap encapsulation only) parameter specifies how often Inverse ARP datagrams are sent on this virtual circuit. The valid range is from 1 to 60. The default value is 15 minutes.

The **oam** *seconds* (optional) parameter specifies how often to generate an OAM F5 loopback cell from this virtual circuit. The valid range is from 1 to 600. The default value is 10 seconds.

The **compress** (optional) parameter enables the payload compression. If compression hardware is present, then the compression method is hardware. If compression hardware is not present, software compression is used. This option is available only on the Cisco MC3810.

Defaults

If the *oam* keyword is omitted, OAM cells are not generated. If the *oam* keyword is present but the *seconds* value is omitted, the default value of OAM seconds is 10 seconds.

If the *inarp* keyword is omitted, Inverse ARPs are not generated. If the *inarp* keyword is present, but the timeout value is not given, then Inverse ARPs are generated every 15 minutes.

If peak and average rate values are omitted, the PVC defaults to peak and average rates equal to the link rate. The peak and average rates are then equal. By default, the virtual circuit is configured to run as fast as possible.

Note The order of command options is important. The *inarp* keyword can be specified either separately or before the *oam* keyword has been enabled. The *peak, average,* and *burst* arguments, if specified, cannot be specified after either the *inarp* or the *oam* keywords.

The Cisco IOS software dynamically creates rate queues as necessary to satisfy the requests of **atm pvc** commands. The software dynamically creates a rate queue when an **atm pvc** command specifies a peak or average rate that does not match any user-configured rate queue.

The **atm pvc** command creates a PVC and attaches it to the VPI and VCI specified. Both *vpi* and *vci* cannot be specified as 0; if one is 0, the other cannot be 0. The *aal-encap* argument determines the AAL mode and the encapsulation method used. The *peak* and *average* arguments determine the rate queue used.

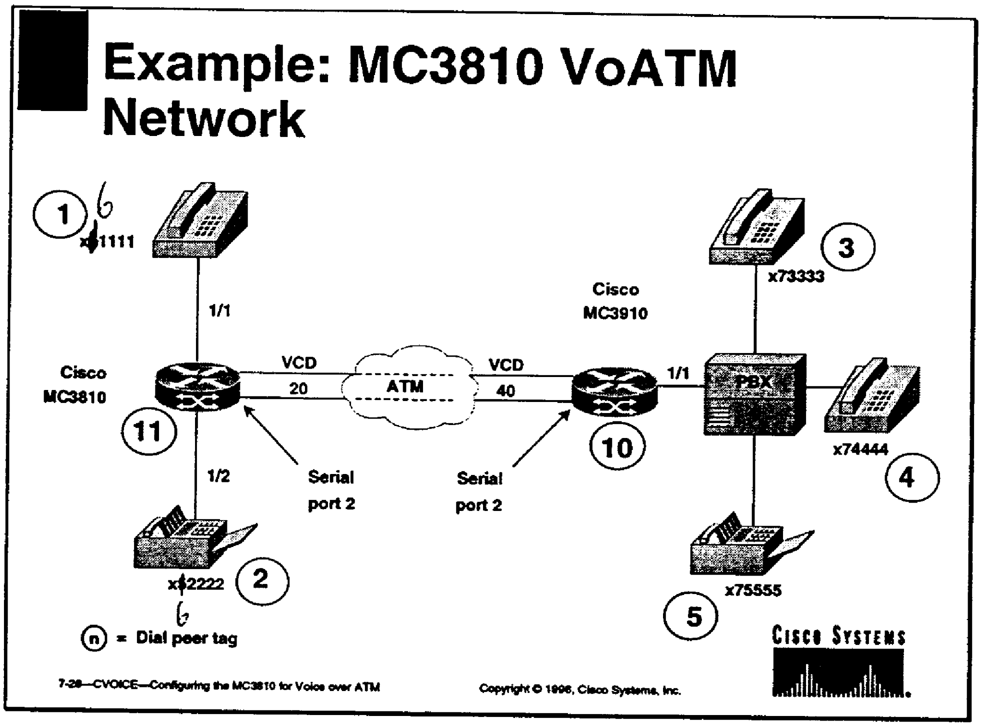
Note If you choose to specify *peak* or *average* values, you must specify both.

The router generates and echoes OAM F5 loopback cells, which verify connectivity. Once OAM cell generation is enabled, a cell is transmitted periodically. The remote end must respond by echoing back the cells.

The router does not generate alarm indication signal (AIS) cells, which are used for alarm surveillance functions. However, if it receives an AIS cell, it responds by sending an OAM far-end remote failure (FERF) cell.

Repeat the **atm pvc** command process for each ATM PVC to be configured over serial port 2.

To verify the ATM PVC configuration, enter the **router#show atm vc** command.



Example 1: Voice over ATM Network

The figure above shows a diagram of a small voice network in which router 1, with ATM virtual circuit 20, connects a small sales branch office to the main office through router 2. There are only two devices in the sales branch office that need to be established as dial peers: a basic telephone and a fax machine. Router 2, with an ATM virtual circuit of 40, is the primary gateway to the main office; as such, it needs to be connected to the company's PBX. There are three devices that need to be established as dial peers in the main office, all of which are basic telephones connected to the PBX.

The table below shows the peer configuration table for the example VoATM network illustrated in the figure above.

Peer Configuration Table for VoATM Network Example

Dial Peer	Ext.	Prefix	Destination Pattern	Type	Voice Port	Session Target
Router 1						
1	61111		13107661111	pots	1/1	
2	62222		13107661111	pots	1/2	
10			1310767....	voatm		S2 20
Router 2						
11			1310766....	voatm		S2 40
3	73333	7	1310767....	pots	1/1	
4	74444	7	1310767....	pots	1/1	
5	75555	7	1310767....	pots	1/1	

Creating a Peer Configuration Table

After you have merged your telephony and WAN networks together, there are tasks you can do to simplify configuring Voice over ATM. One is to collect all of the information directly related to each dial peer by creating a peer configuration table.

There is specific information relative to each dial peer that needs to be identified before you can configure Voice over ATM. One way to do identify this specific information is to create a peer configuration table.

Configuring Dial Peers

Dial peers describe the entities to or from which a call is established. Dial peer configuration tasks define the address or set of addresses serviced by that dial peer and the call parameters required to establish a call to or from that dial peer.

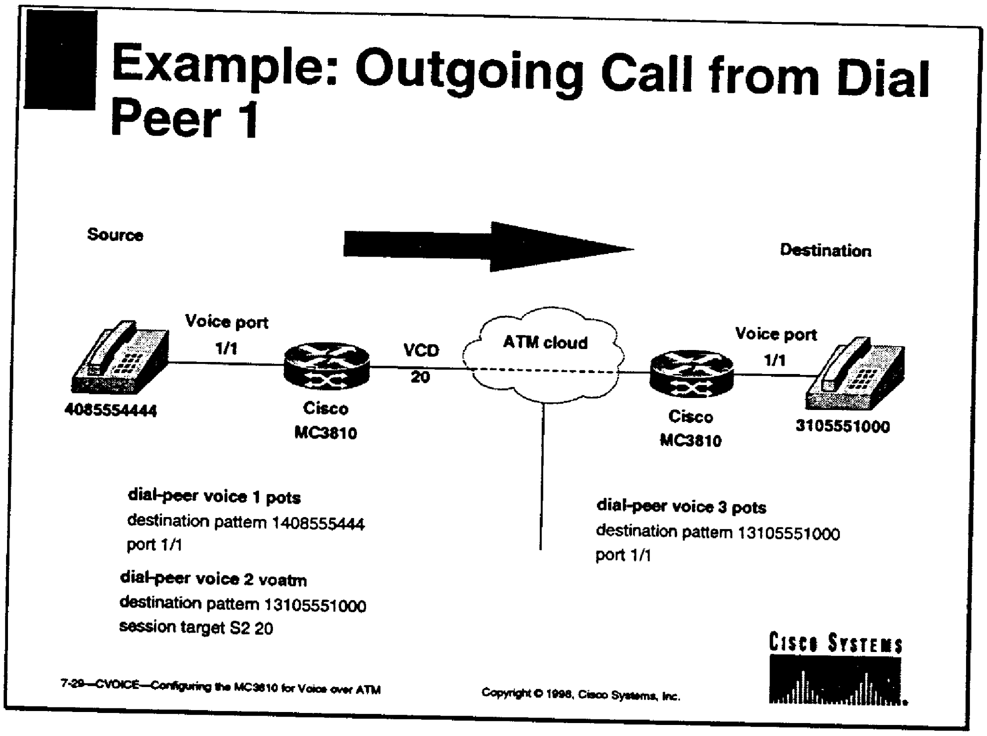
There are two different kinds of dial peers:

- POTS—Dial peer connected via a traditional telephony network. POTS dial peers point to a particular voice port on a voice-network device.
- Voice over ATM—Dial peer connected via an ATM WAN backbone. Voice over ATM dial peers point to specific voice-network devices.

POTS Dial Peers

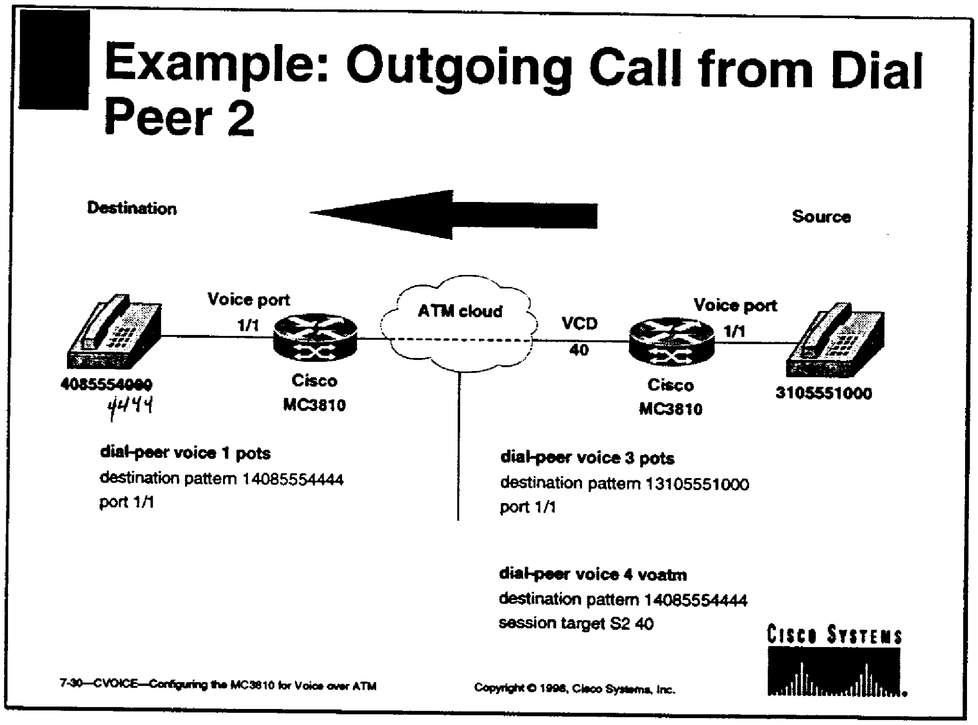
POTS dial peers associate a telephone number with a particular voice port so that incoming calls for that telephone number can be received and outgoing calls can be placed. Voice over ATM dial peers point to specific voice-network devices (by associating destination telephone numbers with a specific ATM VC) so that incoming calls can be received and outgoing calls can be placed. Both POTS and Voice over ATM dial peers are required if you want to both send and receive calls using Voice over ATM.

Establishing two-way communication using Voice over ATM requires establishing a specific voice connection between two defined endpoints. As shown in the following figure, for outgoing calls (from the perspective of the POTS dial peer 1), the POTS dial peer establishes the source (the originating telephone number and voice port) of the call. The Voice over ATM dial peer establishes the destination by associating the destination phone number with a specific ATM virtual circuit.

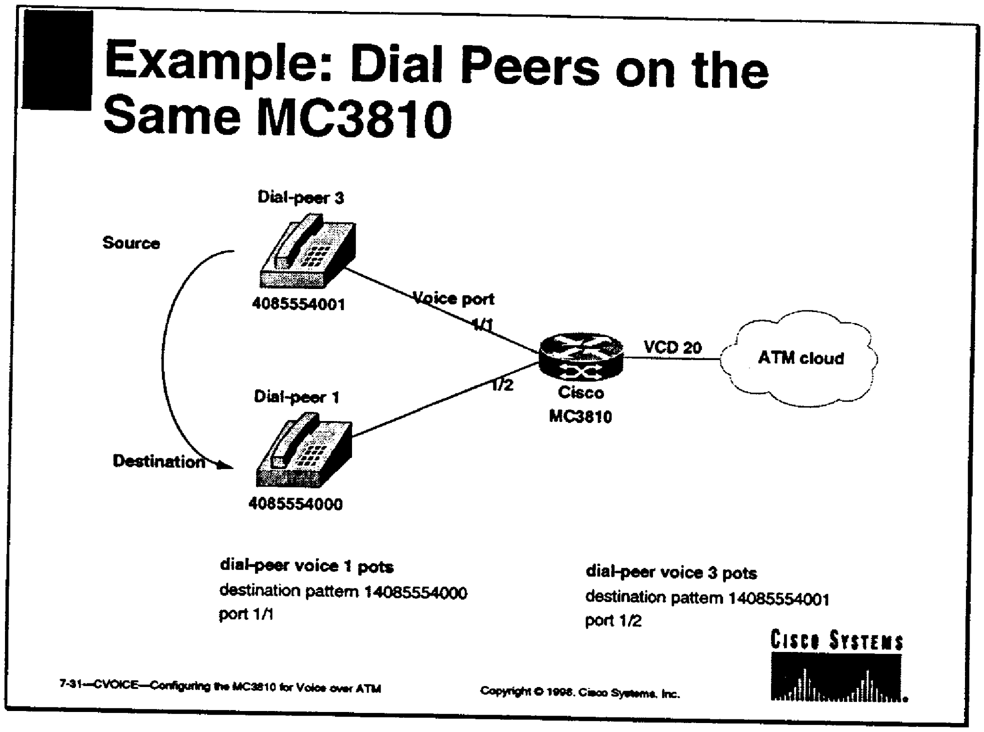


Example: Outgoing Call from Dial Peer 1

To complete the two-way communications loop, you need to configure a Voice over ATM dial peer 2 as shown in the figure above.



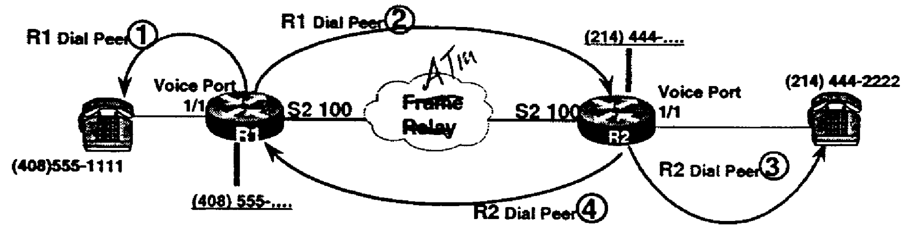
Example: Outgoing Call from Dial Peer 2



Example: Dial Peers on the Same MC3810

The only exception is when dial peers are connected to the same router, as shown in the figure above. In this circumstance, because both dial peers share the same destination IP address, you need not configure a Voice over ATM dial peer.

VoATM Session Target and Destination Pattern



Cisco IOS Configuration on R1

```
dial-peer voice 1 pots
destination-pattern 14085551111
port 1/1

dial-peer voice 2 voatm
destination-pattern 1214444....
session target Serial2 100
```

Cisco IOS Configuration on R2

```
dial-peer voice 3 pots
destination-pattern 12144442222
port 1/1

dial-peer voice 4 voatm
destination-pattern 14085551111
session target Serial2 100
```

CISCO SYSTEMS

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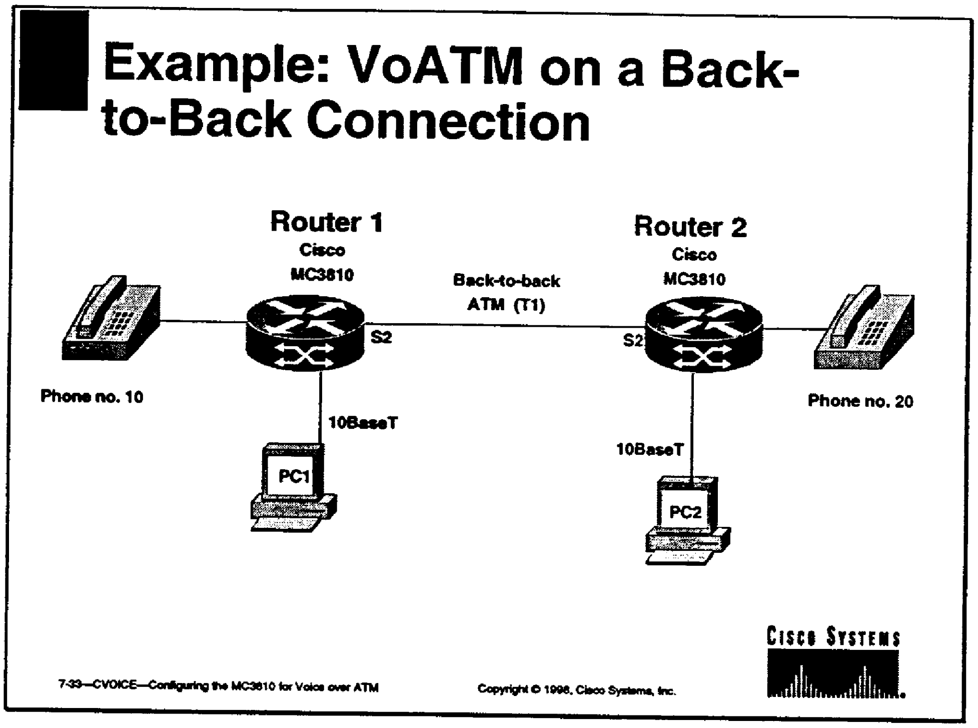
VoATM Session Target and Destination Pattern

When configuring dial peers, you need to understand the relationship between the destination pattern and the session target. The destination pattern represents the pattern for the device at the voice connection endpoint, such as a telephone or a PBX. The session target represents the serial port on the peer Cisco MC3810 at the other end of the ATM connection. The figure above shows the relationship between the destination pattern and the session target, as seen from the perspective of both Cisco MC3810 concentrators in a Voice over ATM configuration.

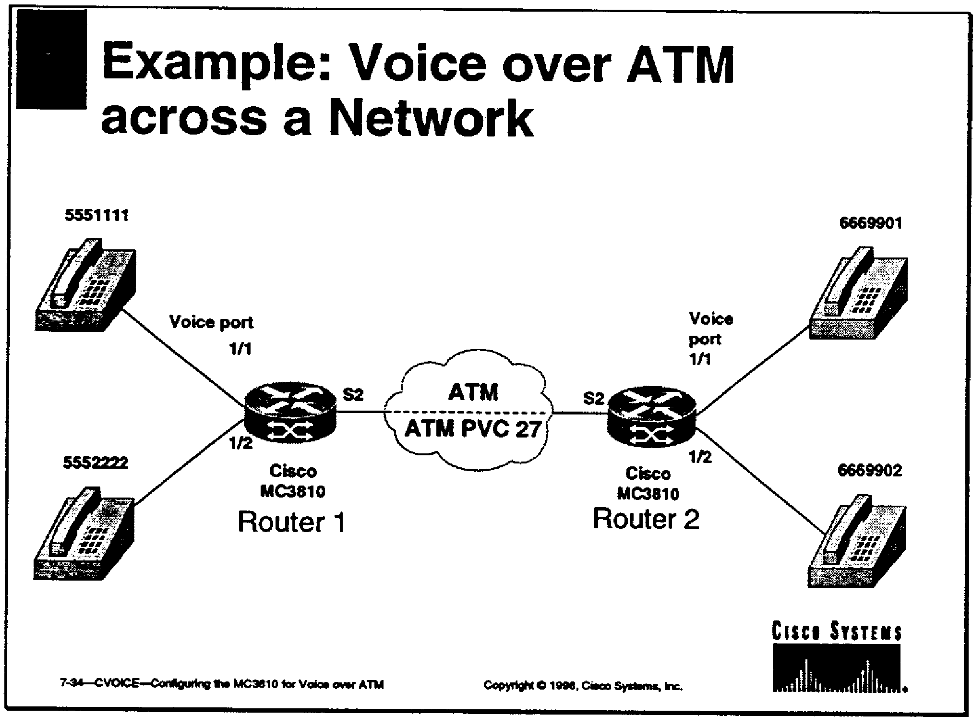


Example: VoATM on a Back-to-Back Connection

Here is a configuration example for two Cisco MC3810 concentrators configured back-to-back, with Voice over ATM configured for both concentrators. The following table lists the commands required to configure the Cisco MC3810 routers in the figure above.

Cisco IOS Commands for VoATM Back-to-Back Connection Example

Configuration for Cisco MC3810 Router 1	Configuration for Cisco MC3810 Router 2
hostname router1	hostname router2
no ip domain-lookup	no ip domain-lookup
controller T1 0	controller T1 0
framing esf	framing esf
clock source ~~internal~~ Line	clock source internal
linecode b8zs	linecode b8zs
interface Ethernet0	interface Ethernet0
ip address 10.1.10.1 255.255.255.0	ip address 10.1.20.1 255.255.255.0
no ip mroute-cache	no ip mroute-cache
no ip route-cache	no ip route-cache
interface Serial2	interface Serial2
ip address 10.1.1.1 255.255.255.0	ip address 10.1.1.2 255.255.255.0
no ip mroute-cache	no ip mroute-cache
encapsulation atm	encapsulation atm
no ip route-cache	no ip route-cache
atm pvc 1 1 100 aal5voice 384 192 48	atm pvc 1 1 100 aal5voice 384 192 48
atm pvc 2 1 200 aal5snap	atm pvc 2 1 200 aal5snap
map-group atm1	map-group atm1
router rip	router rip
redistribute connected	redistribute connected
network 10.0.0.0	network 10.0.0.0
no ip classless	no ip classless
ip route 0.0.0.0 0.0.0.0 10.1.1.1	ip route 0.0.0.0 0.0.0.0 10.1.1.2
map-list atm1	map-list atm1
ip 10.1.1.2 atm-vc 2 broadcast	ip 10.1.1.1 atm-vc 2 broadcast
dial-peer voice 1 pots	dial-peer voice 1 pots
destination-pattern 10	destination-pattern 20
port 1/1	port 1/1
dial-peer voice 202 voatm	dial-peer voice 202 voatm
destination-pattern 2.	destination-pattern 1.
Session target Serial2 1	session target Serial2 1
line con 0	
line aux 0	
line vty 0 4	
password cisco	
login	



Example: Voice over ATM across a Network

The figure above shows an example for both voice and data traffic over ATM between two Cisco MC3810 concentrators, including configuration for voice ports and dial peers. The following table lists the commands required to configure the Cisco MC3810 routers in the figure above.

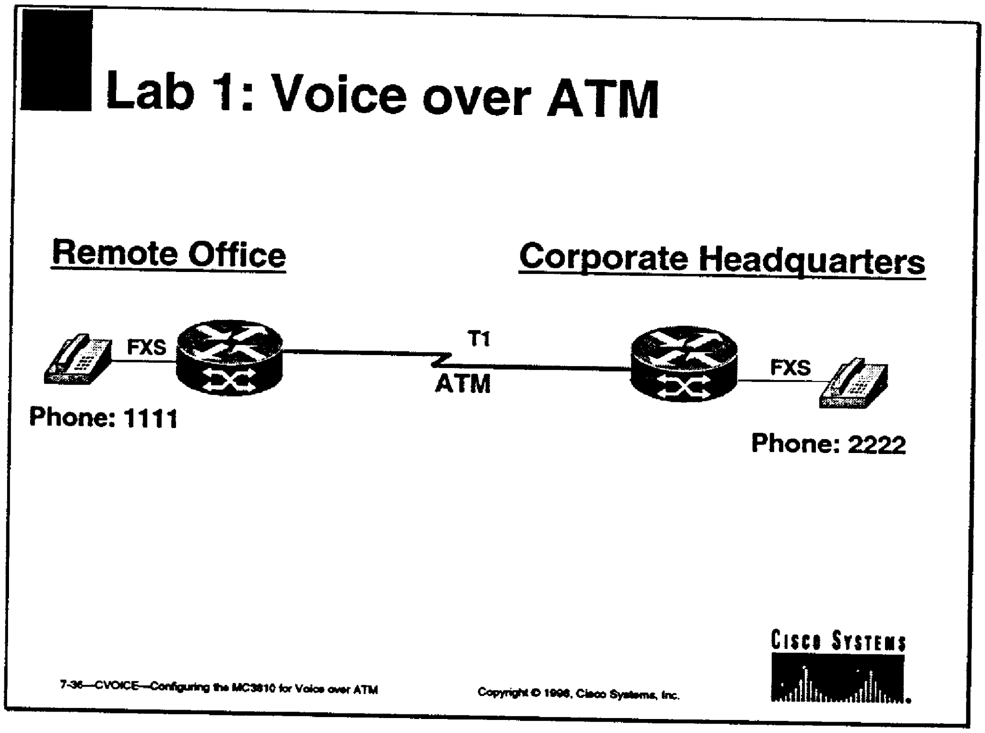
Cisco IOS Commands for VoATM across a Network Connection Example

Configuration of MC3810 1 Router 1	Configuration of MC3810 Router 2
version 11.3	version 11.3
controller T1 0	controller T1 0
framing esf	framing esf
linecode b8zs	linecode b8zs
interface Ethernet0	interface Ethernet0
ip address 172.22.124.239 255.255.0.0	ip address 172.22.124.247 255.255.0.0
interface Serial2	interface Serial2
ip address 223.223.224.229 255.255.255.0	ip address 223.223.224.228 255.255.255.0
no ip mroute-cache	no ip mroute-cache
encapsulation atm	encapsulation atm
no ip route-cache	no ip route-cache
map-group atm1	map-group atm1
atm pvc 26 26 200 aal5snap	atm pvc 26 26 200 aal5snap
atm pvc 27 27 270 aal5voice 384 192 48	atm pvc 27 27 270 aal5voice 384 192 48
no ip classless	no ip classless
map-list atm1	map-list atm1
ip 223.223.224.228 atm-vc 26 broadcast	ip 223.223.224.229 atm-vc 26 broadcast
voice-port 1/1	login
voice-port 1/2	line vty 1 4
voice-port 1/3	login
voice-port 1/4	voice-port 1/1
dial-peer voice 1 pots	voice-port 1/2
destination-pattern 5551111	voice-port 1/3
port 1/1	voice-port 1/4
dial-peer voice 2 pots	dial-peer voice 1 pots
destination-pattern 5552222	destination-pattern 6669901
port 1/2	port 1/1
dial-peer voice 1001 voatm	dial-peer voice 2 pots
destination-pattern 666....	destination-pattern 6669902
session target serial2 27	port 1/2
	dial-peer voice 1001 voatm
	destination-pattern 555....
	session target serial2 27

Lab Exercise 1: Establishing a Voice over ATM Call between Two Remote MC3810s

This lab exercise describes how to configure voice over an ATM connection on the Cisco MC3810 between two remotely connected MC3810s running Cisco IOS 11.3 software.

Lab Setup



Scenario

The ABC Company has several remote offices that are connected to corporate headquarters. It has an existing data network using ATM. The company recently acquired MC3810s to specifically upgrade its network to carry both voice and data traffic between the remote offices and corporate headquarters. Upgrading will result in substantial telecom savings for the company.

As a data/voice specialist for this company, it is your job to install the proper hardware and cables, plus set up and configure the software for Voice over ATM using an MC3810 between the remote offices and corporate headquarters.

Exercise: Configure and Verify the Cisco IOS Software for Host Information, WAN Services, and Voice Services

Configure the Cisco IOS software to make an FXS Voice over ATM call between two MC3810 stations wired back-to-back. Refer to the Lab 1 diagram in Appendix A for complete details such as IP and Ethernet addresses, destination patterns, and other pertinent information.

Directions

Step 1 To begin the configuration, establish a console link to the MC3810, using an ASCII terminal or a laptop set to terminal mode (i.e., Hyper Term or Procomm).

Step 2 Begin by entering privileged EXEC mode. Enter the following Cisco IOS command: **router> en**(able)

Step 3 Enter global configuration mode, **router> config t**(erminal)

Step 4 Set the host name of your router. Refer to the network design layout in Appendix A. **hostname Team_R__** (The hostname is a code to describe your **Team** number, **R**(outer) number, and whether your router has an Analog or **D**igital chassis.)

Step 5 Use the Controller configuration mode to specify whether your controller is E1 or T1, and enter the controller number. If the DVM is installed, the number can be either 0 or 1. If the MFT is installed, the number must be 0. For this lab exercise, we will be using a T1 MFT: **router(config)#controller T1 0**

Step 6 Configure the controller clock source for a DS1 link. The default for the DVM is *internal*, while the default for the MFT is *line*. Because this lab is back-to-back, the "corporate" side will be *internal*, that is, corporate will set clock. The "remote" site will be *line*, receiving clock from the network: **router(config-controller)#clock source {internal | line}**. There is another option, *loop-timed* that will take the clock from the Rx line and use it for Tx. See the *Cisco MC3810 Multiservice Concentrator Command Reference Guide* for more details on how to use this option.

Step 7 Enter a description of the controller, such as the destination or its application, for the line value. The description can be up to 80 characters long. Because this controller is for a T1 trunk, it would be meaningful to give it that description: **router(config-controller)#description T1 trunk**

Step 8 Configure the DS1 link framing format. Extended Super Frame format (**esf**) is required for ATM traffic: **router(config-controller)#framing esf**

Step 9 Configure the line encoding format for the DS1 link. The B8ZS setting is required for Frame Relay and ATM traffic: **router(config-controller)#linecode b8zs**

Step 10 **Exit** from the controller. The prompt should now look like this: **router (config)>**

Step 11 Set up and address the Ethernet interface. At the **router#** prompt, enter the following command to enter the interface configuration mode: **interface Ethernet0**

Step 12 At the **router (config-if)>** prompt, enter the Ethernet IP address and subnet mask: **router (config-if)> ip address 172.16.100.x 255.255.255.0** (The last octet of the Ethernet address will reflect your team and router number. Refer to the Voice over ATM Lab 1 sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 13 Enter the following command for the Ethernet interface: **no ip mroute-cache**

Step 14 **Exit** from the Ethernet interface. The prompt should now look like this:

router (config)>

Step 15 Configure the MFT trunk interface. The MFT trunk is designated as Serial2. At the prompt, enter the following command to configure the T1 interface to be an MFT trunk interface: **router (config)> interface Serial2**

Step 16 Configure the IP address of the Serial2 interface: **router(config-if)> ip-addr 10.11.1.xx** (The last octet of the S2 interface address will reflect your team and router number. Refer to Appendix A for Ethernet IP address assignments for your specific router).

Step 17 Set the MFT trunk interface for ATM encapsulation: **router (config-if)> encapsulation atm**

Step 18 To associate an ATM map list to an interface or subinterface, for either a PVC or SVC, use the following *map-group* command: **router (config-if)> map-group atm1,** where "atm1" is the arbitrary name given to the map list. See steps 22 and 23.

Step 19 To create a PVC that will transport voice traffic on an ATM interface, and to identify the *peak, average,* and *burst* data rates for the voice traffic, use the following command: **router (config-if)> atm pvc 1 1 100 aal5voice 384 192 48.** The values you use will depend on your calculations for peak, average, and burst data rates. To calculate the formula, refer to earlier material for more information.

Step 20 To create a PVC for transporting data over the ATM network, use the following command: **router (config-if)> atm pvc 2 1 200 aal5snap x x x**. The values you use will depend on your calculations for *peak, average,* and *burst* data rates. To calculate the formula, refer to earlier material for more information.

Step 21 To define an ATM map statement for either a PVC or SVC, use the **map-list** global configuration command: **router (config-if)> map-list atm1**

Step 22 To define an ATM map statement for a PVC, use the **atm-vc** command followed by the **map-list** command. The command requires you to identify a protocol and a protocol address along with the PVC's virtual circuit identifier: **router (config-if)> ip 10.11.1.xx atm-vc 2 broadcast** Because we are using virtual circuit descriptor 1 for the voice PVC, the number for the data virtual circuit descriptor is "2." Broadcast allows this map entry protocol to send broadcast packets to the interface. (The last octet of the S2 interface address will reflect your team and router number. Refer to the VoATM Lab 1 summary sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 23 **Exit** from interface configuration mode. The prompt should now look like this: **router (config)>**

Step 24 Set the router for RIP: **router (config)> router rip**

Step 25 Set the router for IP routing capability: **router (config)> ip routing**

Note Refer to the ATM Lab 1 diagram in Appendix A for complete details for the correct numbering requirements for the following commands. If you are seated at a "Corporate" site, go directly to step 35. If you are seated at a "Remote" station, proceed to the next step.

Step 26 Use the "Remote" site phone configuration commands to configure a POTS dial peer for the "Remote" site's phone. To define a POTS dial peer and enter dial peer configuration mode, use the following command: **router (config)> dial-peer voice 1 pots**

Step 27 Configure the phone's dial peer destination pattern: **router (config-dialpeer)> destination-pattern 1111.**

Step 28 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 1/1**

Step 29 To configure a Voice over ATM dial peer, use the following command: **router (config)> dial-peer voice 2 voatm**

Step 30 Configure the VoATM dial peer's destination pattern: **router (config-dialpeer)> destination-pattern 2222.**

Step 31 Configure the session target for the VoATM line:

router (config-dialpeer)> Session target Serial2 2.

Step 32 **Exit** from dial peer configuration mode. The prompt should now look like this: **router (config)>** Exiting from dial peer configuration mode finishes the configuration for this POTS. To configure additional dial peers, reenter dial peer configuration mode.

Step 33 For "Remote" users, after completing the above steps, go to step 42.

Step 34 Configure a POTS dial peer for the "Corporate" site's phone. To define a POTS dial peer and enter dial peer configuration mode, use the following command: **router (config)> dial-peer voice 1 pots**

Step 35 Configure the phone's dial peer destination pattern: **router (config-dialpeer)> destination-pattern 2222.**

Step 36 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 1/1**

Step 37 To configure a Voice over ATM dial peer, use the following command: **router (config)> dial-peer voice 2 voatm**

Step 38 Configure the VoATM dial peer's destination pattern: **router (config-dialpeer)> destination-pattern 1111.**

Step 39 Configure the session target for the VoATM line:

router (config-dialpeer)> Session target Serial2 1.

Step 40 **Exit** from dial peer configuration mode. The prompt should now look like this: **router (config)>** Exiting from dial peer configuration mode finishes the configuration for this POTS. To configure additional dial peers, reenter dial peer configuration mode. After you have configured all dial peers, proceed to the next step.

Step 41 **Exit** from configuration mode.

Completion Criteria

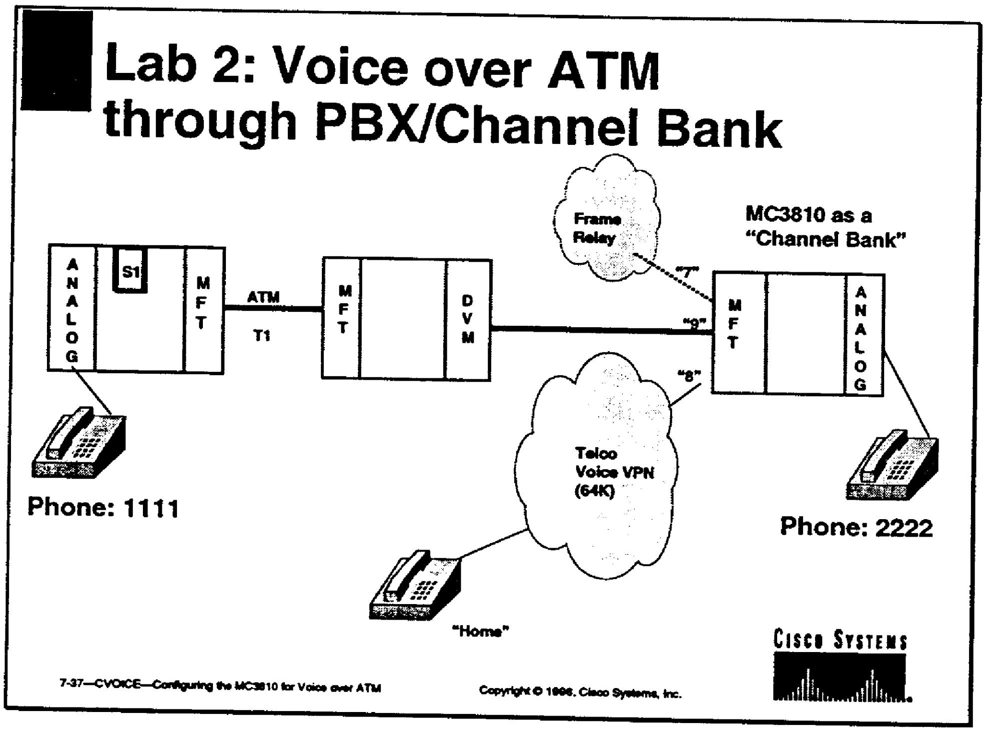
The lab is complete when all Cisco IOS parameters have been configured and a call can be successfully completed between the two back-to-back MC3810s.

If you have not been able to successfully configure the MC3810, consult the following troubleshooting and validation tips, then recheck your configuration, correcting any parameters as needed.

Lab Exercise 2: Pseudo Channel Bank Lab

This lab exercise describes how to configure an MC3810 as a pseudo channel bank or PBX. This lab exercise gives the student the opportunity to see one of the features of the MC3810.

Lab Setup



This lab (see Appendix A for the ATM Lab 2 setup and network diagram) uses an MC3810 as a pseudo channel bank to show how the MC3810 could be connected to either a channel bank or PBX using E&M voice connectivity.

Exercise: Configure and Verify the Cisco IOS Software for Host Information, WAN Services, and Voice Services for the MC3810

Configure the Cisco IOS software to make a phone call to a remote site by first dialing 9 to get to the "channel bank"' or "PBX." The "Corporate" MFT trunk on the "channel bank" MC3810 is a T1 connection to the DVM port on the middle MC3810, a digital voice MC3810. In turn, the digital voice MC3810 is connected by its MFT to the MFT of the "Remote" MC3810 over a T1 ATM link.

Depending at which MC3810 you are sitting, go to the directions for that particular device. The directions for each type of MC3810 are listed on the following pages.

Note Refer to the ATM Lab 2 diagram in Appendix A for complete details such as IP and Ethernet addresses, destination patterns, and other pertinent information.

Directions to Configure an MC3810 as a Channel Bank (Corporate)

Step 1 First, enter **write erase** to clear the configuration from Lab 1.

Step 2 Establish a console link to the MC3810 using an ASCII terminal or a laptop set to terminal mode (i.e., Hyper Term or Procomm).

Step 3 Enter privileged EXEC mode. Enter the following Cisco IOS command: **router> en**

Step 4 Enter global configuration mode: **router> config t**(erminal)

Step 5 Set the host name of your router. Refer to the ATM Lab 2 network design layout in Appendix A. **hostname Team_R__** (The host name is a code to describe your **Team** number, **R**(outer) number, and whether your router has an Analog or **D**igital chassis).

Step 6 Use the controller configuration mode to specify whether your controller is E1 or T1, and enter the controller number. If the DVM is installed, the number can be either 0 or 1. If the MFT is installed, the number must be 0. For this lab exercise, we will be using a T1 MFT: **router(config)#controller T1 0**

Step 7 Configure the controller for channel associated signaling (CAS): **router(config-controller)#mode cas**

Step 8 Configure the channel bank MC3810 for voice group time slots for voice CAS on the T1/E1 controller, and specify E&M signaling: **router(config-controller)#voice-group 1 timeslots 1 type e&m-immediate**

Step 9 Configure the controller clock source for a DS1 link. The default for the DVM is *internal*, while the default for the MFT is *line*. Take the default value: **router(config-controller)#clock source line**. There is another option, *loop-timed* that will take the clock from the Rx line and use it for Tx. See the *Cisco MC3810 Multiservice Concentrator Command Reference Guide* for more details on how to use this option.

Step 10 Enter a description of the controller, such as the destination or its application, for the line value. The description can be up to 80 characters long. Because this MFT controller is for a T1 trunk that will be connected to a DVM port on a digital voice MC3810, it would be meaningful to give it that description: **router(config-controller)#description T1 MFT trunk to DVM**

Step 11 Configure the DS1 link framing format. Extended Super Frame format (**esf**) is required for ATM traffic: **router(config-controller)#framing esf**

Step 12 Configure the line encoding format for the DS1 link. The B8ZS setting is required for Frame Relay and ATM traffic: **router(config-controller)#linecode b8zs**

Step 13 **Exit** from the controller. The prompt should now look like this: **router (config)>**

Step 14 Set up and address the Ethernet interface. At the **router# prompt**, enter the following command to enter the interface configuration mode: **interface Ethernet0**

Step 15 At the **router (config-if)>** prompt, enter the Ethernet IP address and subnet mask: **router (config-if)> ip address 172.16.100.x 255.255.255.0** (The last octet of the Ethernet address will reflect your team and router number. Refer to the ATM Lab 2 sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 16 Enter the following command for the Ethernet interface: **no ip mroute-cache**

Step 17 **Exit** from the Ethernet interface. The prompt should now look like this:

router (config)>

Step 18 Configure the MFT trunk interface. The MFT trunk is designated as Serial2. At the prompt, enter the following command to configure the T1 interface to be an MFT trunk interface: **router (config)> interface Serial2**

Step 19 **"Corporate" site phone configuration commands.** Configure a POTS dial peer for the "Corporate" site's phone. To define a POTS dial peer and enter dial peer configuration mode, use the following command: **router (config)> dial-peer voice 1 pots**

Step 20 Configure the phone's dial peer destination pattern: **router (config-dialpeer)> destination-pattern 2222.**

Step 21 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 1/1**

Step 22 Configure a second POTS dial peer for the "Corporate" site's phone. **This POTS dial peer is to get to an "outside line" by dialing 9: router (config)> dial-peer voice 2 pots**

Step 23 Configure the phone's dial peer destination pattern: **router (config-dialpeer)> destination-pattern 9.**

Step 24 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 0/1**

Step 25 **Exit** from dial peer configuration mode. The prompt should now look like this: **router (config)>** Exiting from dial peer configuration mode finishes the configuration for this POTS dial peer. To configure additional dial peers, reenter dial peer configuration mode. After you have configured all dial peers, proceed to the next step.

Step 26 **Exit** from configuration mode.

Completion Criteria

The lab is complete when all Cisco IOS parameters have been configured and a call can be successfully completed between the two back-to-back MC3810s.

If you have not been able to successfully configure the MC3810, consult the troubleshooting and validation tips in Appendix A, then recheck your configuration, correcting any parameters as needed.

Directions to Configure the DVM MC3810 (Digital "Cloud")

Configure the Cisco IOS software to make an FXS Voice over ATM call between the "Corporate" MC3810 "channel bank" and the "Remote" MC3810 stations. This MC3810 is a digital chassis device, and will act as a "cloud" for this lab exercise.l Refer to the Lab 2 diagram in Appendix A for complete details such as IP and Ethernet addresses, destination patterns, and other pertinent information.

Step 1 Enter **write erase** to clear the MC3810.

Step 2 Enter privileged EXEC mode. Enter the following Cisco IOS command:

router> en

Step 3 Enter global configuration mode: **router> config t**(erminal)

Step 4 Set the host name of your router. Refer to the ATM Lab 2 network design layout in Appendix A. **hostname Team_R__** (The host name is a code to describe your **Team** number, **R**(outer) number, and whether your router has an Analog or Digital chassis).

Step 5 Configure the MFT controller. In this lab, the digital MC3810 has two controllers: the MFT and the DVM. To enter controller configuration mode, specify that your controller is a T1, and enter the controller number. Remember: Because the MFT is installed, the controller number must be 0. The DVM is also installed in this MC3810, so it can only be designated as 1. Configure the MFT first. **router(config)#controller T1 0**

Step 6 Configure the controller clock source as internal. **router(config-controller)#clock source internal**

Step 7 Enter a description of the controller, such as the destination or its application, for the line value. The description can be up to 80 characters long. Because this controller is for an ATM T1 trunk going to the "Remote" site, it would be meaningful to give it that description: **router(config-controller)#description ATM T1 trunk to "Remote" MC3810**

Step 8 Configure the DS1 link framing format. Extended Super Frame format (**esf**) is required for ATM traffic: **router(config-controller)#framing esf**

Step 9 Configure the line encoding format for the DS1 link. The B8ZS setting is required for Frame Relay and ATM traffic: **router(config-controller)#linecode b8zs**

Step 10 **Exit** from the controller. The prompt should now look like this: **router (config)>**

Step 11 Configure the DVM controller. Reenter controller configuration mode. Specify the DVM controller is installed. Enter the number1:

router(config)#controller T1 1

Step 12 Configure the controller for channel associated signaling (CAS): **router(config-controller)#mode cas**

Step 13 Configure the channel bank MC3810 for voice group time slots for voice cas on the T1/E1 controller, and specify E&M signaling:

router(config-controller)#voice-group 1 timeslots 1 type e&m-immediate

Step 14 Configure the controller clock source for a DS1 link. The default for the DVM is *internal*. Take the default value: **router(config-controller)#clock source internal**. There is another option, *loop-timed*, that will take the clock from the Rx line and use it for Tx.

Reference: See the *Cisco MC3810 Multiservice Concentrator Command Reference Guide* for more details on how to use this option.

Step 15 Enter a description of the controller, such as the destination or its application, for the line value. The description can be up to 80 characters long. Because this DVM controller is for a T1 trunk that will be connected to a MFT port on a "channel bank" MC3810, it would be meaningful to give it that description: **router(config-controller)#description T1 DVM trunk to MFT channel bank**

Step 16 Configure the DS1 link framing format. Extended Super Frame format (**esf**) is required for ATM traffic: **router(config-controller)#framing esf**

Step 17 Configure the line encoding format for the DS1 link. The B8ZS setting is required for Frame Relay and ATM traffic: **router(config-controller)#linecode b8zs**

Step 18 **Exit** from the controller. The prompt should now look like this: **router (config)>**

Step 19 Set up and address the Ethernet interface. At the **router#** prompt, enter the following command to enter the interface configuration mode:

interface Ethernet0

Step 20 At the **router (config-if)>** prompt, enter the Ethernet IP address and subnet mask: **router (config-if)> ip address 172.16.100.x 255.255.255.0** (The last octet of the Ethernet address will reflect your team and router number. Refer to the Voice over ATM Lab 1 sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 21 Enter the following command for the Ethernet interface: **no ip mroute-cache**

Step 22 **Exit** from the Ethernet interface. The prompt should now look like this:

router (config)>

Step 23 Configure the MFT trunk interface. The MFT trunk is designated as Serial2. At the prompt, enter the following command to configure the T1 interface to be an MFT trunk interface: **router (config)> interface Serial2**

Step 24 Configure the IP address of the Serial2 interface: **router(config-if)> ip-addr 10.11.1.xx** (The last octet of the S2 interface address will reflect your team and router number. Refer to Appendix A for Ethernet IP address assignments for your specific router).

Step 25 Set the MFT trunk interface for ATM encapsulation:

router (config-if)> encapsulation atm

Step 26 To associate an ATM map list to an interface or subinterface, for either a PVC or SVC, use the following *map-group* command, using "atm1" as the arbitrary name given to the map list:

router (config-if)> map-group atm1

Step 27 To create a PVC that will transport voice traffic on an ATM interface, and to identify the *peak, average,* and *burst* data rates for the voice traffic, use the following command: **router (config-if)> atm pvc 1 1 100 aal5voice 384 192 48.** The values you use will depend on your calculations for peak, average, and burst data rates. To calculate the formula, refer to earlier material for more information.

Step 28 To create a PVC for transporting data over the ATM network, use the following command: **router (config-if)> atm pvc 2 1 200 aal5snap x x x.** The values you use will depend on your calculations for *peak, average,* and *burst* data rates. To calculate the formula, refer to earlier material for more information.

Step 29 To define an ATM map statement for either a PVC or SVC, use the **map-list** global configuration command: **router (config-if)> map-list atm1**

Step 30 To define an ATM map statement for a PVC, use the **atm-vc** command followed by the **map-list** command. The command requires you to identify a protocol and a protocol address along with the PVC's virtual circuit identifier: **router (config-if)> ip 10.11.1.xx atm-vc 2 broadcast**

Because we are using virtual circuit descriptor 1 for the voice PVC, the number for the data virtual circuit descriptor is "2." Broadcast allows this map entry protocol to send broadcast packets to the interface. (The last octet of the S2 interface address will reflect your team and router number. Refer to the VoATM Lab 2 summary sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 31 **Exit** from interface configuration mode. The prompt should now look like this:

router (config)>

Step 32 Set the router for RIP: **router (config)> router rip**

Step 33 Set the router for IP routing capability: **router (config)> ip routing**

Step 34 Because the digital MC3810 is in essence passing calls through it, you will need to configure two POTS dial peers. To define a POTS dial peer and enter dial peer configuration mode, use the following command: **router (config)> dial-peer voice 1 pots**

Step 35 Configure the phone's dial peer destination pattern:

router (config-dialpeer)> destination-pattern 9

Step 36 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 1/1**

Step 37 **Configure a VoATM dial peer.** To configure a Voice over ATM dial peer to the "Remote" site, use the following command:

router (config)> dial-peer voice 2 voatm

Step 38 Configure the VoATM dial peer's destination pattern:

router (config-dialpeer)> destination-pattern 1111.

Step 39 Configure the session target for the VoATM line:

router (config-dialpeer)> Session target Serial2 1.

Step 40 **Exit** from dial peer configuration mode. The prompt should now look like this:

router (config)>

Exiting from dial peer configuration mode finishes the configuration for this POTS dial peer. To configure additional dial peers, reenter dial peer configuration mode.

Step 41 **Exit** from configuration mode.

Completion Criteria

The lab is complete when all Cisco IOS parameters have been configured and a call can be successfully completed between the two back-to-back MC3810s.

If you have not been able to successfully configure the MC3810, consult the troubleshooting and validation tips in Appendix A, then recheck your configuration, correcting any parameters as needed.

Directions to Configure the "Remote" MC3810

Configure the Cisco IOS software to make an FXS Voice over ATM link to the DVM MC3810 digital "cloud." Refer to the Lab 2 diagram in Appendix A for complete details such as IP and Ethernet addresses, destination patterns, and other pertinent information.

Step 1 Establish a console link to the MC3810 using an ASCII terminal or a laptop set to terminal mode (i.e., Hyper Term or Procomm).

Step 2 Entering privileged EXEC mode: **router> en**(able)

Step 3 Enter global configuration mode: **router> config t**(erminal)

Step 4 Set the host name of your router. Refer to the network design layout in Appendix A. **hostname Team_R__** (The host name is a code to describe your **Team** number, **R**(outer) number, and whether your router has an Analog or **D**igital chassis.)

Step 5 Use the controller configuration mode to specify whether your controller is E1 or T1, and enter the controller number. If the DVM is installed, the number can be either 0 or 1. If the MFT is installed, the number must be 0. For this lab exercise, we will be using a T1 MFT: **router(config)#controller T1 0**

Step 6 Configure the controller clock source for a DS1 link. The default for the DVM is *internal*, while the default for the MFT is *line*. Because this lab is back-to-back, the "corporate" side will be *internal*, that is, corporate will set clock. The "remote" site will be *line*, receiving clock from the network: **router(config-controller)#clock source {internal | line}**. There is another option, *loop-timed*, that will take the clock from the Rx line and use it for Tx. See the *Cisco MC3810 Multiservice Concentrator Command Reference Guide* for more details on how to use this option.

Step 7 Enter a description of the controller, such as the destination or its application, for the line value. The description can be up to 80 characters long. Because this controller is for an ATM MFT T1 trunk, going to the MFT trunk on the DVM MC3810, it would be meaningful to give it that description: **router(config-controller)#description ATM MFT T1 trunk to the MFT on the DVM**

Step 8 Configure the DS1 link framing format. Extended Super Frame format (**esf**) is required for ATM traffic: **router(config-controller)#framing esf**

Step 9 Configure the line encoding format for the DS1 link. The B8ZS setting is required for Frame Relay and ATM traffic: **router(config-controller)#linecode b8zs**

Step 10 **Exit** from the controller. The prompt should now look like this: **router (config)>**

Step 11 Set up and address the Ethernet interface. At the **router#** prompt, enter the following command to enter the interface configuration mode:

interface Ethernet0

Step 12 At the **router (config-if)>** prompt, enter the Ethernet IP address and subnet mask: **router (config-if)> ip address 172.16.100.x 255.255.255.0** (The last octet of the Ethernet address will reflect your team and router number. Refer to the Voice over ATM Lab 1 sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 13 Enter the following command for the Ethernet interface: **no ip mroute-cache**

Step 14 **Exit** from the Ethernet interface. The prompt should now look like this:

router (config)>

Step 15 Configure the MFT trunk interface. The MFT trunk is designated as Serial2. At the prompt, enter the following command to configure the T1 interface to be an MFT trunk interface: **router (config)> interface Serial2**

Step 16 Configure the IP address of the Serial2 interface: **router(config-if)> ip-addr 10.11.1.xx**

Note The last octet of the S2 interface address will reflect your team and router number. Refer to Appendix A for Ethernet IP address assignments for your specific router.

Step 17 Set the MFT trunk interface for ATM encapsulation. **router (config-if)> encapsulation atm**

Step 18 To associate an ATM map list to an interface or subinterface, for either a PVC or SVC, use the following *map-group* command: **router (config-if)> map-group atm1**, where "atm1" is the arbitrary name given to the map list.

Step 19 To create a PVC that will transport voice traffic on an ATM interface, and to identify the *peak, average,* and *burst* data rates for the voice traffic, use the following command: **router (config-if)> atm pvc 1 1 100 aal5voice 384 192 48**.

The values you use will depend on your calculations for peak, average, and burst data rates. To calculate the formula, refer to earlier material for more information.

Step 20 To create a PVC for transporting data over the ATM network, use the following command: **router (config-if)> atm pvc 2 1 200 aal5snap x x x**. The values you use will depend on your calculations for *peak, average,* and *burst* data rates. To calculate the formula, refer to earlier material for more information.

Step 21 To define an ATM map statement for either a PVC or SVC, use the **map-list** global configuration command: **router (config-if)> map-list atm1**

Step 22 To define an ATM map statement for a PVC, use the **atm-vc** command followed by the **map-list** command. The command requires you to identify a protocol and a protocol address along with the PVC's virtual circuit identifier.
router (config-if)> ip 10.11.1.xx atm-vc 2 broadcast Because we are using virtual circuit descriptor 1 for the voice PVC, the number for the data virtual circuit descriptor is "2." Broadcast allows this map entry protocol to send broadcast packets to the interface. (The last octet of the S2 interface address will reflect your team and router number. Refer to the VoATM Lab 1 summary sheet in Appendix A for Ethernet IP address assignments for your specific router.)

Step 23 **Exit** from interface configuration mode. The prompt should now look like this:

router (config)>

Step 24 Set the router for RIP: **router (config)> router rip**

Step 25 Set the router for IP routing capability: **router (config)> ip routing**

Step 26 Configure a POTS dial peer for the "Remote" site's phone. To define a POTS dial peer and enter dial peer configuration mode, use the following command: **router (config)> dial-peer voice 1 pots**

Step 27 Configure the phone's dial peer destination pattern: **router (config-dialpeer)> destination-pattern 1111**.

Step 28 Associate this POTS dial peer with a specific logical dial interface. This next command associates a slot/port number of a voice port for this POTS dial peer: **router (config-dialpeer)> port 1/1**

Step 29 To gain access the DVM, configure a Voice over ATM dial peer using the following command: **router (config)> dial-peer voice 2 voatm**

Step 30 Configure the VoATM dial peer's destination pattern: **router (config-dialpeer)> destination-pattern 9**

Step 31 Configure the session target for the VoATM line:

router (config-dialpeer)> Session target Serial2 1.

Step 32 **Exit** from dial peer configuration mode. The prompt should now look like this:

router (config)>

Exiting from dial peer configuration mode finishes the configuration for this POTS dial peer. To configure additional dial peers, reenter dial peer configuration mode.

Step 33 **Exit** from configuration mode.

Completion Criteria

The lab is complete when all Cisco IOS parameters have been configured and a call can be successfully completed between the two back-to-back MC3810s.

If you have not been able to successfully configure the MC3810, consult the troubleshooting and validation tips in Appendix A, then recheck your configuration, correcting any parameters as needed.

Summary

Summary

This chapter covered the following topics:

- **Description of VoATM software and WAN trunk interfaces of the MC3810**
- **List of Cisco IOS commands specific to configuring the MC3810 for VoATM**
- **Configuration of an MC3810 for VoATM using Cisco IOS software**

7-38—CVOICE—Configuring the MC3810 for Voice over ATM

Summary (cont.)

- Configuration of the MC3810 WAN services, ATM voice port interfaces, and dial-peers using Cisco IOS software so a phone call can be successfully completed between two MC3810s
- Configuration of the MC3810 as a "channel bank" using the MC3810's advanced functionality

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7-30—CVOICE—Configuring the MC3810 for Voice over ATM

Configuring Your 2600, 3600, or AS5300 for Voice over IP

Objectives

Objectives

Upon completion of this chapter, you will be able to perform the following tasks:

- Given the need to transport VoIP, identify the issues that must be addressed to transport delay sensitive traffic like voice over an IP network
- Given an existing data network, configure the host and WAN with RSVP, weighted fair queuing, and RTP header compression to carry real-time traffic
- Given a VoIP network, improve voice quality by setting the quality of service options on the VoIP dial peers
- Given a VoIP network and verification commands, verify the configurations necessary for a VoIP call

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Chapter Content

This chapter specifically tells you how to configure your 2600, 3600, or AS5300 for VoIP traffic.

The chapter outline follows:

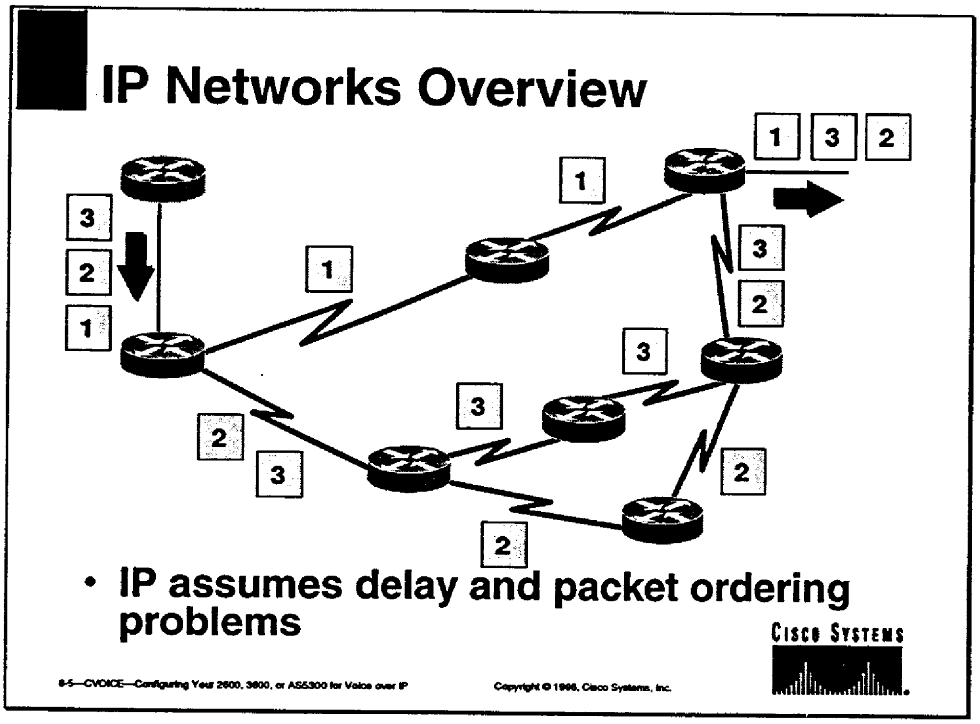
- Introduction
- IP Networks Overview
- Voice over IP Protocols
- Configuring Network for Voice Traffic
- WAN Interface Options
- Configuring RSVP, IP Precedence, Codec, and VAD on VoIP Dial Peers
- Configuring Number Expansion
- Lab Exercise
- Summary

Introduction

Cisco is the proven leader in developing and transporting time-sensitive data across WANs. With the 2600, 3600, or AS5300 family of routers, you can now run telephony over your existing data network.

VoIP offers the following benefits:

- Using the IP network for direct connections within an enterprise
- Reuse of WAN network to transport voice traffic



IP Networks Overview

Unlike the Frame Relay and ATM connection-oriented networks, IP is a connectionless network. Connectionless networks generally do not participate in signaling. In other words, the concept of session establishment exists between end systems, while the connectionless network remains ignorant of the "virtual circuit."

IP resides at the network layer of the OSI protocol stack. Therefore, it can transport IP packets over deterministic and nondeterministic Layer 2 protocols like Frame Relay and ATM; IP can be used to communicate across any set of interconnected networks. It is equally suited for both LAN and WAN communication.

IP information is transferred in a sequence of datagrams. One message may be transmitted as a series of datagrams that are reassembled into the message at the receiving location.

Traditionally, IP traffic has been transmitted on a first-in, first-out basis. Packets have been variable in nature, allowing large file transfers to take advantage of the efficiency associated with larger packet sizes.

User Datagram Protocol (UDP) is the connectionless transport layer protocol used for Voice over IP. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols.

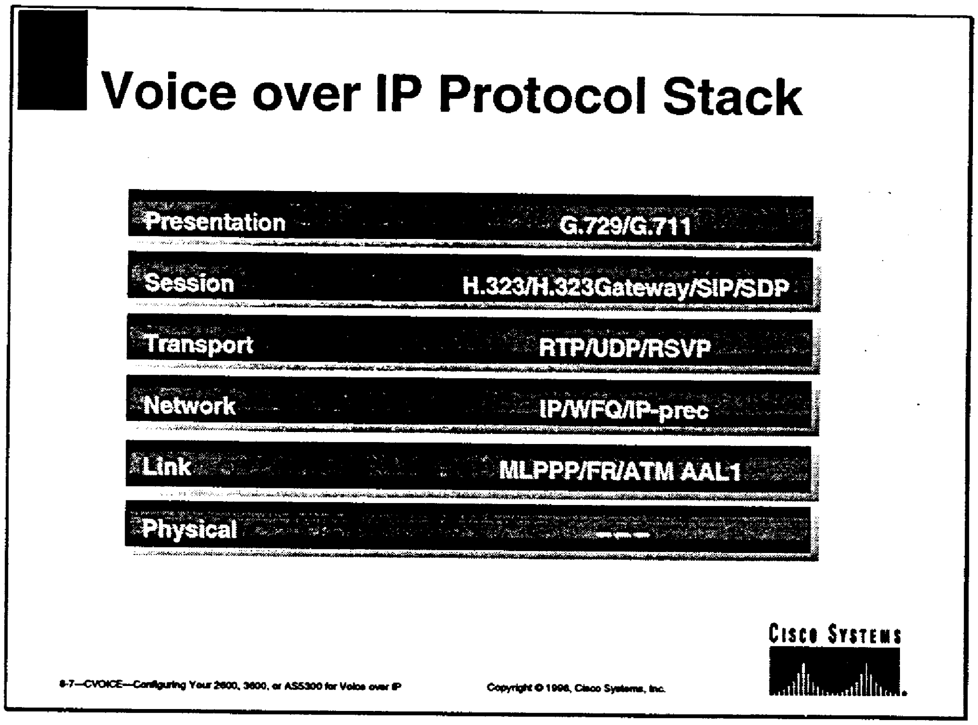
The characteristics of IP mentioned have contributed to large delays and large delay variations in packet delivery. With IP, packet ordering problems may also occur. The challenge to order voice packets properly with no delay so that upon receipt the packets translate into a continuous, verbal stream of sound must be addressed.

Recent efforts have been made through standards to support traffic that is more delay sensitive. These efforts include packet prioritization, resource reservation (i.e., bandwidth), and packet fragmentation. They will be discussed in subsequent sections.

Learn H.323

Delivering Voice over X

Dan Minoli



Voice over IP Protocol Stack

To successfully integrate connection-oriented voice traffic in a connectionless-oriented IP network, enhancements to the signaling stack are required. In some ways we must make the connectionless network appear more connection oriented.

The accepted model for transmitting multimedia like voice across a network like IP that does not guarantee quality of service is H.323. H.323 allows for standards-based interoperability with other vendors' H.323-compatible equipment. H.323 describes terminals, equipment, and services for multimedia communication over LANs. Any H.323-compliant terminal is required to carry voice.

The Cisco 2600, 3600, or AS5300 router acts as an H.323 gateway and assumes some of the functionality of a gatekeeper. An H.323 gatekeeper is required to perform the following:

- Address translation
- Admission control
- Bandwidth management
- Zone management

The H.323 protocol comprises audio, video, data applications, and system control. Available Cisco VoIP audio codecs include G.711 and G.729. As better codecs are developed, the marketplace will determine which codecs are specified.

Other components required for H.323 terminals are H.245, H225.0/Q.931, and Real-Time Transport Protocol (RTP)/RTP Control Protocol (RTCP). Descriptions follow.

The H.245 control channel provides in-band reliable transport for capabilities exchange, mode preference from the receiving end, logical channel signaling, and control and indication. UDP/TCP is used for Voice over IP to provide the reliable transport. H.245 allows H.323 devices to deliver its capabilities to the other H.323 devices. Part of these capabilities are codecs available.

H.225.0 utilizes a scaled-down version of Q.931, a protocol used for call establishment, to set up the connection between two H.323 endpoints.

RTP provides end-to-end network functions and services for delay-sensitive, real-time data like voice. Services include the following:

- Payload type identification
- Sequence numbering
- Time-stamping
- Delivery monitoring

RTCP was established to monitor the quality of the data distribution and provide control information. It provides feedback on current network conditions.

The RTP and RTCP are specified in the H.323 specification. After the H.323 call setup and control process is completed, audio and video packets are sent via UDP. To assist with streaming audio and video, the specification calls for an RTP header. An RTP header contains a time stamp and sequence number, allowing 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.

RTP or H.323 do not provide quality of service guarantees from the network. The Resource Reservation Protocol (RSVP) is the network control protocol that allows Internet applications to obtain special qualities of service for their data flows.

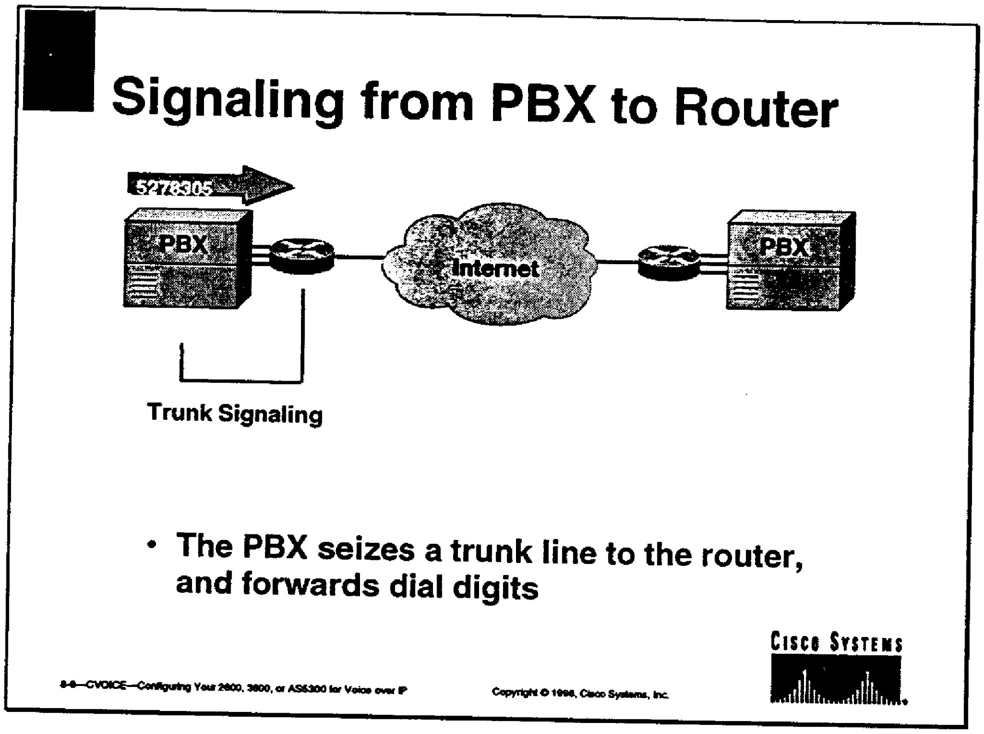
RSVP is not a routing protocol. Instead it works in conjunction with routing protocols to prioritize traffic. In VoIP, RTP works with weighted fair queuing to prioritize voice traffic over other traffic.

Voice over IP signaling takes place between three distinct areas:

- Signaling from the PBX (or other telephony device) to the router
- Signaling between routers
- Signaling from the router to the PBX (or other telephony device)

The signaling is described in subsequent slides.

Reference: For more detail regarding TCP IP, and UDP data packets, refer to the "TCP/IP Overview" chapter in the "Introduction to Cisco Router Configuration" course.



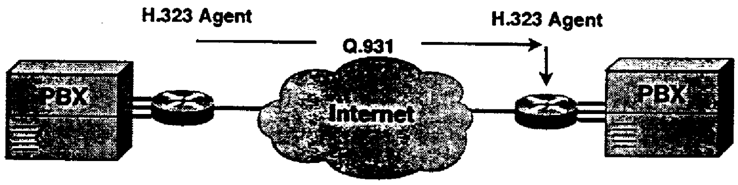
Signaling from PBX to the Router

The user picks up the handset, signaling an off-hook condition.

The connection between the PBX and router appears as a trunk line to the PBX, which will signal the router to seize a trunk. Once a trunk is seized, the PBX then forwards the dialed digits to the router in the same manner the digits would be forwarded to a telephone company switch.

Signaling from the PBX to the router may be any of the common signaling methods used to seize a trunk line (FXS, FXO, or E&M signaling).

Signaling between Routers



- **The Dial Plan Mapper reads the dial digits, and finds the address of the remote IP peer. The H.323 agent initiates a Q.931 call to the remote peer.**

8-8—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

Signaling between Routers

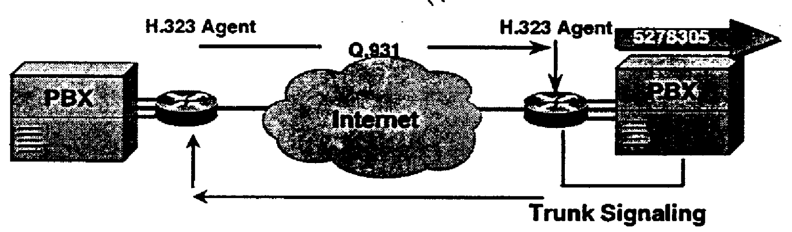
Within the router, the Dial Plan Mapper maps the dialed digits to an IP address. The H.323 agent then signals a Q.931 Call Establishment Request to the remote peer that is indicated by the IP address.

Meanwhile, the control channel is used to set up RTP, audio streams, and the RSVP if it is being used.

RTP

Signaling from Router to PBX

H.235 H.225 H.245



- **The remote H.323 agent seizes a PBX trunk, returns a Q.931 acknowledgment to the origin, and forwards dial digits to the PBX**

8-10—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

Signaling from Router to PBX

When the remote router receives the Q.931 call request, it signals a line seizure to the PBX. After the PBX acknowledges, the router forwards the dialed digits to the PBX and signals a call acknowledgment back to the originating router.

Meanwhile, the control channel is used to set up RTP, audio streams, and the RSVP if it is being used. The voice codecs are turned on for both ends, and the conversation proceeds using RTP/UDP/IP as the protocol stack.

When either end hangs up, the RSVP reservations are torn down, if used, and the session ends, with each end going idle waiting for another off-hook.

Configuring Network for Voice Transport

You must configure your WAN for voice compatibility. This section explains how to configure the WAN interface so it can transport delay sensitive traffic like voice. This section and subsequent sections further discuss the elements in the IP protocol stack that you must configure.

Configuring WAN Interfaces and IP Network for Voice Traffic

This section explains how you can configure your WAN interface to carry real-time traffic like voice. It specifically addresses when and how to configure the following on your WAN IP interface:

- RTP/RTCP and RTP header compression
- QoS and RSVP
- Weighted fair queuing

Before you configure your interfaces for VoIP, you must have them configured for IP.

Reference: This course assumes you already have experience configuring a router for IP. For more detail on basic router configurations, reference the "Introduction to Cisco Router Configuration" (ICRC) course.

To configure the router for IP you must first enter global configuration mode. Then you can create, load, or alter any existing configuration information. To enter global configuration mode, enter the **configure terminal** command in privileged EXEC mode. Once in the global configuration mode, perform the following steps if they were not already performed using the SETUP script when the router was first powered up:

Step 1 Give your router a host name using the **hostname** command.

Step 2 Enter the interface configuration mode for the interface you wish to configure with the **interface** command.

Step 3 You must also specify an IP address for the interface using the **ip address** command.

Step 4 If you have a DCE serial connection, set the clock rate with the **clockrate** command.

Step 5 Repeat steps 2 through 4 for each interface.

Step 6 You must also specify the routing protocol used with the **router** command.

Step 7 While in router configuration mode, specify the networks your router is directly attached to with the **network** command.

Note Use the **show running-config** command to confirm correct configuration.

RTP and RTCP ~~+~~ Transport

- **Real-Time ~~Transfer~~ Transport Protocol**

  Jitter Buffer

  Connectionless environment

  Payload type identification, sequence numbering

  Time-stamping, delivery monitoring

- **Real-Time ~~Transfer~~ Transport Control Protocol**

  Provides feedback on current network conditions

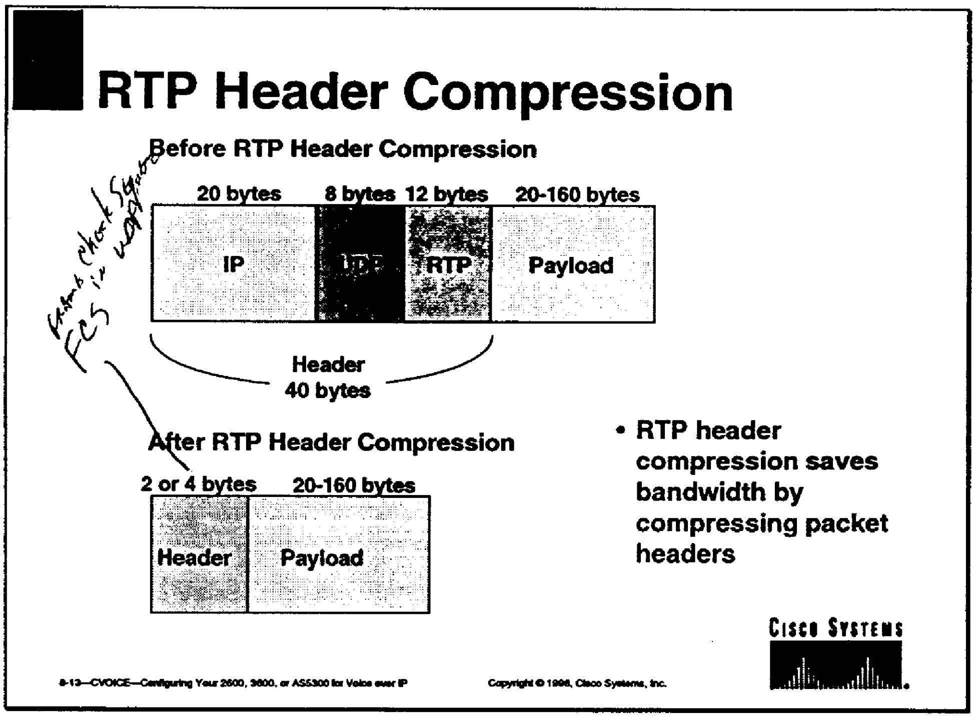
RTP and RTCP

RTP is a host-to-host protocol used for carrying newer multimedia application traffic, including packetized audio and video, over an IP network. It is the Internet standard protocol for the transport of real-time data. RTP provides end-to-end network transport functions intended for applications requiring minimum delay, such as audio, video, or simulation data over multicast or unicast network services.

Because of the connectionless IP environment, RTP provides the following functions:

- Identifies the payload type
- Provides sequence numbering
- Provides time-stamping
- Monitors delivery

RTCP monitors the quality of the data distribution and provides control information. It provides feedback on current network conditions.



RTP Header Compression

Must Enable RTP Header Compression Link By Link Throughout the network

With the multiple protocols necessary to transport voice over an IP network, the header is large. RTP header compression is used on a link-by-link basis to compress the IP/UDP/RTP from 40 bytes to 2 or 4 bytes most of the time. In a packet voice environment when framing speech samples every 20 ms, this scenario generates a payload of 20 bytes. The total packet size comprises an IP header (20 bytes), a UDP header (8 bytes), and an RTP header (12 bytes) combined with a payload of 20 bytes. It is evident that the size of the header is twice the size of the payload. When generating packets every 20 ms on a slow link, the header consumes a large portion of bandwidth.

To avoid unnecessary consumption of available bandwidth, RTP header compression is used on a link-by-link basis. This compression scheme reduces the IP/UDP/RTP header to 2 bytes most of the time when no UDP checksums are being sent, or 4 bytes when UDP checksums are used.

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

- Narrowband links
- Need to conserve bandwidth on your WAN interface

Do not use RTP header compression if you have high-speed interfaces, generally T1 and above.

The figure above illustrates how the packets are compressed.

Configuring RTP Header Compression

Router(config-if)#

```
ip rtp header-compression [ passive ]
```

- **Compresses header from 40 bytes to 2 or 4 most of the time**

Router(config-if)#

```
ip rtp compression connections number
```

- **Specifies the total number of RTP header compression connections supported on an interface**

CISCO SYSTEMS

8-14—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

Configuring RTP Header Compression

To enable RTP header compression on a serial interface, enter the interface configuration mode. You can then use the **ip rtp header-compression [passive]** command.

Use the optional **passive** keyword to request that the software only compresses outgoing RTP packets if incoming RTP packets on the same interface are already compressed. If you use the command without the **passive** keyword, the software compresses all RTP traffic.

Note If you are configuring RTP header compression on a Frame Relay interface, the command to enable RTP header compression is **frame-relay ip rtp header-compression**.

Note Remember to enable RTP header compression on all interfaces at both ends of a serial connection.

Depending on the traffic on the interface, you may need to change the number of header compression connections. By default, the software supports a total of 16 RTP header compression connections on an interface. To specify a different number of RTP header compression connections, use the **ip rtp compression connections** *number* command to specify the number of RTP header compression connections. If the number of real-time compression connections required on an interface is above the default, you will need to use this command to increase the number of connections.

Configuring QoS and RSVP

Router(config-if)#

ip rsvp bandwidth [*interface-kbps*][*single-flow-kbps*]

- **Allows you to reserve bandwidth for real-time traffic**
- **Provides the policy to weighted fair queuing**

CISCO SYSTEMS

8-15—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

By Default 75% will Be Allocated For Full Interface

Configuring QoS and RSVP

(note) make sure we Identify the actual Bandwidth of the interfaces

Quality of service (QoS) is an umbrella term to refer to the network's ability to provide specified service levels. Service levels refer to the actual QoS specification, the ability of a network to deliver end-to-end service requested by specific network applications, with some level of control delay, loss, jitter, and bandwidth.

On relatively low-bandwidth connections, such as narrowband serial links, consider using methods that ensure QoS. If you have a high-bandwidth network, such as Ethernet and Fast Ethernet, and the voice and data traffic together occupy only a small fraction of the bandwidth available, you may not need to specify QoS.

RSVP is one QoS feature you should consider. RSVP allows end-to-end systems to request certain QoS guarantees from the network. Delay-sensitive traffic like voice requires a guaranteed network consistency. Without consistent QoS, real-time traffic can experience jitter, insufficient bandwidth, delay variations, or information loss. RSVP provides guarantees under normal conditions that service between hosts will not vary by reserving resources across the networks traversed.

You should configure RSVP to ensure QoS if the following conditions exist in your network:

- Links with high utilization: links with sustained usage over 60 percent
- Links less than 2 Mbps, or narrowband links
- Need for the best possible voice quality
- You notice that many voice packets are being dropped

To configure RSVP for voice traffic, you must enable RSVP on a WAN interface. From the interface configuration mode, configure RSVP with the **ip rsvp**

bandwidth command. This command starts RSVP and sets the bandwidth and single-flow limits. The default bandwidth is up to 75 percent of the bandwidth available on the interface, but the amount reservable for a flow can be up to the entire reservable bandwidth.

ip rsvp bandwidth Command

interface-kbps	(Optional) Amount of bandwidth on interface to be reserved.
single-flow-kbps	(Optional) Amount of bandwidth allocated to single flow.

Note You must also configure RSVP on the dial peers. The "Configuring RSVP, IP Precedence, Codec, and VAD on VoIP Dial Peers" section in this chapter instructs you how to configure the voice interfaces.

Configuring Weighted Fair Queuing

Router(config-if)#

fair-queue [*congestive-discard-threshold* [*dynamic-queues* [*reservable-queues*]]]

- **Queuing algorithm to sort traffic**
- **Works well with RSVP**
- **Schedules interactive traffic like voice to the front and shares remaining bandwidth for other traffic**

Cisco Systems

8-16—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

Configuring Weighted Fair Queuing

One way that network elements handle an overflow of arriving traffic is to use a queuing algorithm to classify the traffic and determine some method of prioritizing it onto an output link. Queuing will prioritize voice traffic over other data traffic and reduce the delay that is so detrimental for VoIP calls.

There are a few queuing algorithms to choose within Cisco IOS™software. Each was designed to solve a specific network traffic problem.

Weighted fair queuing (WFQ) works well for VoIP when you are also using RSVP because RSVP uses WFQ to allocate buffer space and schedule packets, and guarantees bandwidth for reserved flows.

Note Custom queuing and priority queuing are also alternatives depending on your network configuration. These two Cisco IOS queuing alternatives are not covered in the scope of this section.

WFQ was designed for situations in which it is desirable to provide consistent response time to heavy and light network users alike without provisioning additional, excessive bandwidth. WFQ is a flow-based queuing algorithm that does two things simultaneously: It schedules interactive traffic like voice to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth between high bandwidth flows.

WFQ ensures that the major causes of inconsistent response time, continuous trains of packets, are sorted into separate streams and forced to interleave.

To enable WFQ on an interface, use the **fair-queue** interface configuration command.

congestive-discard-threshold	(Optional) Number of messages allowed in each queue in the range 1 to 512. The default is 64 messages. When the number of messages in the queue for a high-bandwidth conversation reaches the specified threshold, new high-bandwidth messages are discarded.
dynamic-queues	(Optional) Number of dynamic queues used for best-effort conversations in the range 0 to 1000. The default is 0. Reservable queues are used for interfaces configured for the RSVP feature.
reservable-queues	(Optional) Number of reservable queues used for reserved conversations (that is, a normal conversation not requiring any special network services). Values are 16, 32, 64, 128, 256, 512, 1024, 2048, and 4096. The default is 256.

Note WFQ is enabled by default on most serial interfaces configured at E1 speed (2.048 Mbps) or less with Cisco IOS Release 11.0 software.

Configuring the WAN Interface Procedure

To configure the WAN interface, perform the following tasks from the global configuration mode:

Step 1 Enter the interface configuration mode with an **interface** command and prepare to configure the interface to also transport real-time voice traffic.

Step 2 Configure your interface so real-time traffic like voice will experience less congestion and delay. Depending on your network connection, consider enabling RTP header compression, RSVP, and weighted fair queuing.

Step 3 Exit the interface configuration mode and go back to the global configuration mode with the **exit** command.

Step 4 Verify that the voice port was configured properly with the **show running-config** command. If necessary, go back into interface configuration mode and make the proper changes.

Step 5 Repeat steps 1 through 4 for all applicable interfaces.

WAN Interface Options

Because IP is a higher layer connectionless protocol, IP traffic can be transported over data link layer, connection-oriented protocols like Frame Relay, ATM, or PPP, to list a few.

This section illustrates additional recommended configurations you should consider if you are running Frame Relay or PPP.

Frame Relay Option—Additional Configurations for Voice

Router(config)#

interface *interface number*

- **Enters interface or subinterface configuration mode**

Router(config-if)#

encapsulation frame-relay

- **Enables Frame Relay on the interface**

Router(config-if)#

frame-relay interface-dlci *dlci*

- **Assigns a DLCI to a Frame Relay subinterface**

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Frame Relay Option—Additional Configurations for Voice

Frame Relay is one of several Layer 2 data-link protocols that can be used to carry the voice packets from router to router. Because IP is a Layer 3 protocol, you can use it to run your IP traffic over a Frame Relay data link.

Frame Relay is an ITU-T and American National Standards Institute (ANSI) standard that defines the process for sending data over a public data network (PDN). It is a connection-oriented data-link technology that is streamlined to provide high performance and efficiency.

Note Frame Relay defines the interconnection process between your equipment, the router, and the Internet service provider's local access switching equipment. It does not determine how the data is transmitted within the service provider's Frame Relay cloud.

To configure Frame Relay, you must select the interface you want to configure to send voice packets into the cloud. You must then confirm that basic router configurations are made.

To configure Frame Relay on a serial link, enter the interface configuration mode with the **interface** command and specify the serial connection you wish to configure with the **encapsulation frame-relay** command. You must also configure a subinterface. Use the **interface** configuration command to prepare to configure the subinterface as well. To assign a DLCI to a specified Frame Relay subinterface on the router, use the **frame-relay interface-dlci** command.

Reference: This chapter assumes you already know how to configure a Frame Relay link. Other than the commands, no additional instruction is provided here. However, at the end of this section is an example using the Frame Relay commands you should know to run VoIP on a Frame Relay data link. For more information on Frame Relay, reference the section on Frame Relay in Chapter 6. Frame Relay configuration instructions are also in the "Configuring Frame Relay" chapter in the ICRC course.

You need to take certain other factors into consideration when configuring Voice over IP for it to run smoothly over Frame Relay. For Frame Relay links with slow data rates (less than or equal to 64 kbps), where data and voice are being transmitted over the same PVC, you should configure the following:

- QoS and RSVP
- Weighted fair queuing
- RTP header compression
- Lower the MTU size
- Generic traffic shaping

Note No matter what encapsulation you use to carry voice, you should consider enabling weighted fair queuing, RSVP, and RTP header compression on your interface. Setting the MTU size and traffic shaping are more specific to Frame Relay and will be addressed in this section. Reference the sections on weighted fair queuing, RSVP, and RTP header compression in this chapter for information regarding them.

Frame Serialization Delay Matrix

Frame Size

LinkSpeed	1 Byte	64 Bytes	128 Bytes	256 Bytes	512 Bytes	1024 Bytes	1500 Bytes
56 kbps	143 *	9 ms	18 ms	36 ms	72 ms	144 ms	214 ms
64 kbps	125 *	8 ms	16 ms	32 ms	64 ms	128 ms	187 ms
128 kbps	62.5 *	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256 kbps	31 *	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512 kbps	15.5 *	1 ms	2 ms	4 ms	8 ms	16 ms	23 ms
768 kbps	10 *	640 *	1.3 ms	2.6 ms	5.1 ms	10.2 ms	15 ms
1536 kbps	5 *	320 *	640 *	1.3 ms	2.6 ms	5.1 ms	7.5 ms

*** Size in microseconds**

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Frame Serialization Delay Matrix

WFQ and RSVP are important, but they do not solve the entire issue of transporting delay sensitive traffic on slow links like a Frame Relay link. Even with queuing algorithms, delay problems can still exist with delay-sensitive traffic like voice because large frames or packets needed to produce acceptable bulk transmission efficiency cause unacceptable queuing delays for small real-time packets like voice.

As the graphic shows, delay becomes increasingly significant on slow links and large frame sizes. For example, a 1500-byte frame takes 215 ms to transmit on a 56-kbps link. If an RTP packet waits that long it will be late. If a voice packet is held up because a large data packet is transmitting, voice quality will suffer.

Fragmenting the maximum packet or frame size to smaller values so voice and data packets can better interleave is a solution. To fragment the large packets in a Frame Relay network, lower the maximum transmission unit (MTU).

Configuring MTU

Router(config-if)#

```
mtu bytes
```

- **Specifies the maximum packet size a particular interface can transmit**
- **Lowering the MTU size will allow larger data packets to interleave with voice packets**

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Cisco Systems

Configuring MTU

Until FRF.12 is supported in Cisco IOS software (VoFR Frame Relay fragmentation), MTU sizing may be used on any low-bandwidth Frame Relay link.

MTU specifies the maximum packet size, in bytes, that a particular interface will handle. Voice packets are generally small. If the MTU size is lowered to 300 bytes, for example, large data packets can be broken up into smaller data packets that can be interleaved with voice packets and lessen the delay time of the voice packets.

To adjust the MTU size, from the interface configuration mode, use the **mtu** *bytes* command. You will configure this command on the main interface.

Note Use the **mtu** command to reduce the average packet size. You will thus increase the amount of packets in the network.

On low-bandwidth Frame Relay links, it is necessary to fragment larger packets to avoid the delay inherent with large-byte packets. Without a tool to fragment on a link-by-link basis similar to Multiclass Multilink PPP (MCML PPP), it is necessary to set the MTU size on the interface to match the available bandwidth and the total delay budget. This setup solves the large packet issue by fragmenting every packet above the configured MTU size to the "new" MTU size for the interface.

Caution *Configuring the MTU size on your WAN interface is a critical element on designing your network. Care should be taken when using this feature.*

Configuring Generic Traffic Shaping

Router(config-if)#

traffic-shape rate *bit-rate* [*burst-size* [*excess-burst-size*]]

- **Enables traffic shaping for outbound traffic on an interface**
- **Limits chances for voice packets to be discarded**

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Cisco Systems

Configuring Generic Traffic Shaping

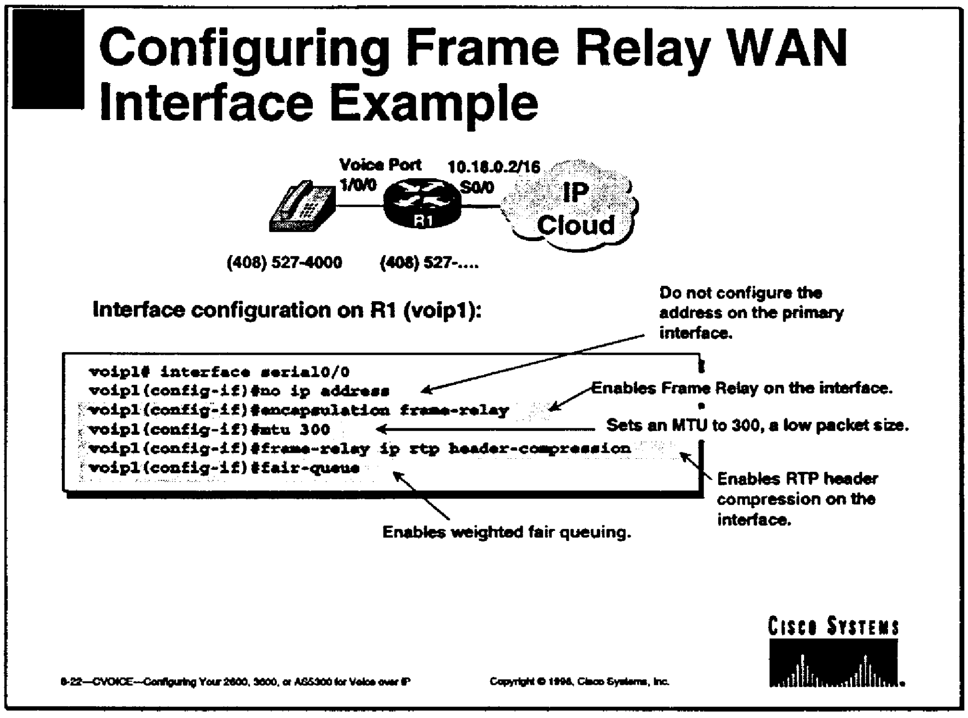
The generic traffic shaping feature is a media and encapsulation-independent traffic shaping tool that helps reduce the flow of outbound traffic when there is congestion within the cloud, on the link, or at the receiving endpoint router. Because a Frame Relay switch does not distinguish between voice and data packets, voice packets may otherwise be discarded if there were a bottleneck. With generic traffic shaping, the Frame Relay switch will not discard packets when the CIR is exceeded. This feature uses RSVP/WFQ weights.

Note The generic traffic shaping feature cannot be enabled at the same time as the Frame Relay traffic shaping feature. Use generic traffic shaping because in Cisco IOS Release 11.3, Frame Relay traffic shaping is not compatible with RSVP.

To configure generic traffic shaping for outbound traffic on an interface, use the **traffic shape rate** command. The values you select depend on your network needs.

traffic-shape rate Command

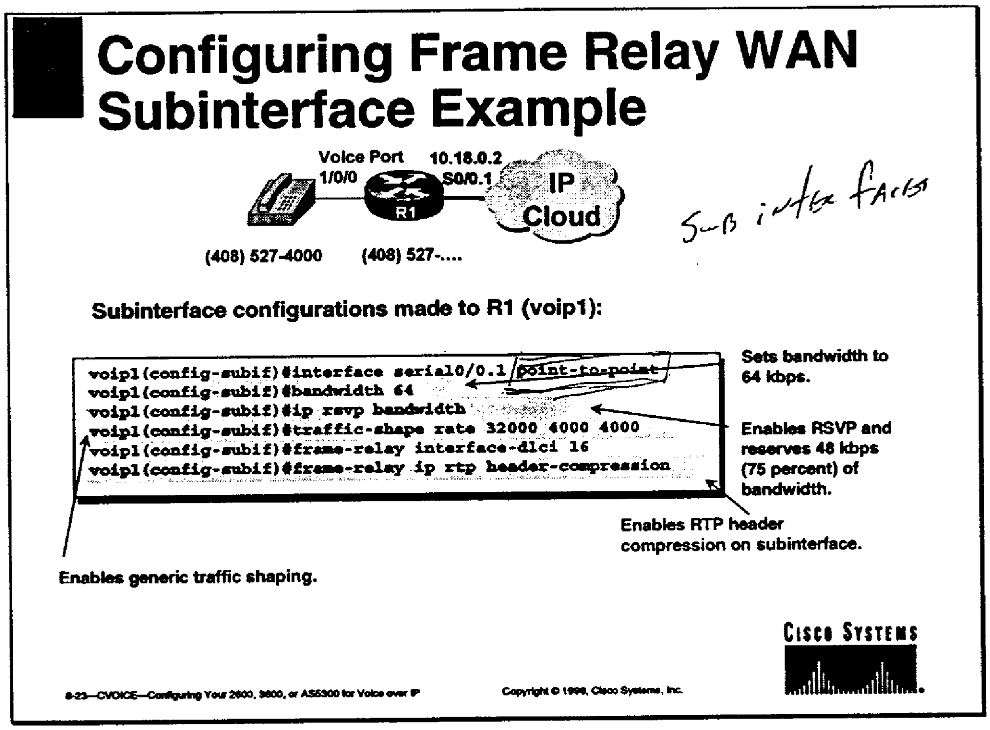
bit-rate	Bit rate that traffic is shaped to in kbps. This is the access bit rate that you contract with your service provider or the service level you intend to maintain.
burst-size	(Optional) Sustained number of bits that can be transmitted per interval. On Frame Relay interfaces, this is the committed burst size contracted with your service provider. The default is the bit-rate divided by 8.
excess-burst-size	(Optional) Maximum number of bits that can exceed the burst size in the first interval in a congestion event. On Frame Relay interfaces, this is the excess burst size contracted with your service provider. The default is equal to the burst-size.



Configuring Frame Relay WAN Interface Example

This example illustrates the configuration commands you use to run VoIP on a Frame Relay network. The graphic illustrates an example of how to configure the Frame Relay interface to carry voice traffic.

Note Pay particular attention to what you configure on the main interface and what you configure on the subinterface.



Sub interfaces

Note) FR if not just (ip RTP Header-comp)

Configuring Frame Relay WAN Subinterface Example

You must also configure a subinterface. Review the configurations in the figure above.

Configuring Frame Relay for VoIP Procedure

To configure Frame Relay for VoIP, from the global configuration mode perform the following tasks:

Step 1 Enter the interface configuration mode and prepare to configure the WAN serial interface using the **interface** command.

Step 2 Encapsulate Frame Relay with the **encapsulation frame-relay** command.

Step 3 For real-time voice traffic, enable any of the following if necessary:

- Smaller packet sizes for easier interleaving with the **mtu** command.
- Weighted fair queuing with the **fair-queue** command.
- RTP header compression with the **frame-relay ip rtp header-compression** command.

Step 4 Enter the subinterface configuration mode with the **interface** command.

Step 5 Specify your IP address with the **ip address** command.

Step 6 Specify the bandwidth on the interface with the **bandwidth** command.

Step 7 If necessary, reserve bandwidth for voice with the **ip rsvp bandwidth** command.

Step 8 Enable generic traffic shaping if necessary with the **traffic shape rate** command.

Step 9 Notify the router of your DLCI with the **frame-relay interface-dlci** command.

Step 10 If you are using RTP header compression, enable it on the subinterface also with the **frame-relay ip rtp header-compression** command.

PPP Option—Configuring Multiclass Multilink PPP for Voice

Router(config-if)#

ppp multilink

- **Enables Multilink PPP**

Router(config-if)#

ppp multilink interleave

- **Enables real-time packet interleaving**

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PPP Option—Configuring Multiclass Multilink PPP for Voice

The theory behind multilink fragmentation and interleaving or MCML PPP is that on slow bandwidth links, there needs to be a method for fragmenting larger packets and then queuing the smaller packets between the fragments of the large packet. When transporting voice packets over a Frame Relay network, packets are fragmented by lowering the MTU size.

With PPP, fragmenting is accomplished using some of the features of Multilink PPP (MP) and tweaking them slightly to allow for interleaving to occur.

Note Multilink PPP should not be used on links greater than 2 Mbps.

MCML PPP builds upon the ability of MP to fragment packets. MCML PPP offers 4 or 16 levels of suspension ("queuing"), while MP offers only one level. MCML PPP does not require both ends of a link to support MCML PPP.

Note MCML PPP can be used only on interfaces that can run PPP, immediately ruling out a large portion of WAN networks (Frame Relay, and so on).

Note MCML PPP specifies only the fragmentation method and suspension levels; it does not specify the queuing technique needed to prioritize the fragments.

Caution *MCML PPP and interleaving can be used only on a dialer interface. Therefore, on a leased-line interface, a virtual template must be used. For more information on configuring a dialer interface or virtual template, refer to the Cisco IOS Release 11.3* Dial Solutions Configuration Guide.

Only on Dial up Interfaces (ISDN) Not on LEASE Lines

To configure MCML PPP and interleaving on a configured and operational interface or virtual interface template, configure the following commands in the interface configuration mode.

- To enable Multilink PPP, enter the **ppp multilink** command.
- To enable real-time packet interleaving, enter the **ppp multilink interleave** command.

Configuring Multiclass Multilink PPP for Voice Example

```
interface virtual-template 1
   ip address 192.168.121.18 255.255.255.248
   no ip mroute-cache
   ppp multilink
   ppp multilink interleave
multilink virtual-template 1
```

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Configuring Multiclass Multilink PPP for Voice Example

The figure above is an example that defines a virtual interface template that enables MCML PPP interleaving and a maximum real-time traffic delay of 20 seconds, and then applies that virtual template to the Multilink PPP bundle.

Configuring MCML PPP Procedure

To configure MCML PPP on your interface, complete the following tasks

Step 1 From the global configuration mode,enter the interface configuration mode with the **interface** command.

Step 2 Set the encapsulation for PPP with the **encapsulation ppp** command.

Step 3 Enable Multilink PPP with the **ppp multilink** command.

Step 4 Enable real-time packet interleaving with the **ppp multilink interleave** command.

Configuring RSVP, IP Precedence, Codec, and VAD on VoIP Dial Peers

In Chapter 5, you were instructed to configure dial peers and voice ports. When transporting VoIP traffic, pay particular attention to your network configuration and configuring voice ports and dial peers accordingly. For example, if you are using analog technology to signal between two PBXs, use E&M interfaces and configure accordingly.

Once you configure your voice ports and dial peers, you may want to optimize dial peer configuration for real-time traffic like voice. To assure real-time voice quality you should configure the dial peers with some of the performance features the Cisco IOS software offers.

Configuring RSVP

Router(config-dial-peer)#

```
req-qos { best-effort | controlled load | guaranteed-delay }
```

- **Specifies the desired QoS on the VoIP dial peer**

Router(config-dial-peer)#

```
acc-qos { best-effort | controlled load | guaranteed-delay }
```

- **Requests an SNMP event if QoS falls below a specified level**

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Configuring RSVP

The "Configuring the WAN Interface Procedure" section recommended that you configure RSVP to maintain certain QoS levels. If you configured your WAN or LAN interfaces for RSVP, you must also specify the level of QoS on your VoIP dial peer.

To specify the level of QoS, enter the VoIP dial peer configuration mode and select the dial peer you wish to configure with QoS; use the **dial-peer voice** *number* **voip** command to enter the VoIP dial peer configuration mode.

Once in dial peer configuration mode, use the **req-qos** command to specify the quality of service you desire.

req-qos Command

best-effort	Indicates that RSVP makes no bandwidth reservation.
controlled-load	Indicates that RSVP guarantees a single level of preferential service, presumed to correlate to a delay boundary. The controlled load service uses admission (or capacity) control to assure that preferential service is received even when the bandwidth is overloaded.
guaranteed-delay	Indicates that RSVP reserves bandwidth and guarantees a minimum bit rate and preferential queuing if the bandwidth reserved is not exceeded.

The default level is **best-effort.** In addition, two types of dynamic reservations can be made with RSVP: **controlled-load** and **guaranteed-delay.**

According to the RSVP RFC, controlled load is defined as follows: The end-to-end behavior provided to an application by a series of network elements that provide controlled-load service tightly approximates the behavior visible to applications that receive best-effort service under unloaded conditions, or conditions not heavily loaded or congested. It does not mean the absence of all other traffic from the same series of network elements. If the network functions correctly, these applications may assume that:

- A very high percentage of transmitted packets will be successfully delivered by the network to the receiving end nodes (the percentage of packets not successfully delivered must closely approximate the basic packet error rate of the transmission medium).
- The transit delay experienced by a very high percentage of the delivered packets will not greatly exceed the minimum transmit delay experienced by any successfully delivered packet (this minimum transit delay includes speed-of-light delay plus the fixed processing time in routers and other communications devices along the path).

To ensure that these conditions are met, clients who request controlled-load service provide the intermediate network elements with an estimation of the data traffic they will generate, the TSpec. In return, the service ensures that network element resources adequate to process traffic falling within this descriptive envelope will be available to the client. If the client's traffic generation properties fall outside of the region described by the TSpec parameters, the QoS provided to the client may exhibit characteristics indicative of overload, including large numbers of delayed or dropped packets. The service definition does not require that the precise characteristics of this overload behavior match those that would be received by a best-effort data flow traversing the same path under overloaded conditions.

Guaranteed service is defined as follows: The end-to-end behavior provided by a series of network elements that conform to this document is an assured level of bandwidth that, when used by a policed flow, produces a delay-bounded service with no queuing loss for all conforming datagrams (assuming no failure of network components or changes in routing during the life of the flow).

If you wanted to specify a **guaranteed-delay** QoS for a call leg, enter VoIP dial peer configuration mode if not already there by entering the **dial-peer voice** *tag* **voip** command. Then enter **req-qos guaranteed-delay**.

Use the **acc-qos** dial peer command to generate a Simple Network Management Protocol (SNMP) event if the quality of service for a specified dial peer drops below the specified level; when the **acc-qos** command is in use, the Cisco IOS software generates a trap message when the bandwidth required to provide the selected quality of service is not available.

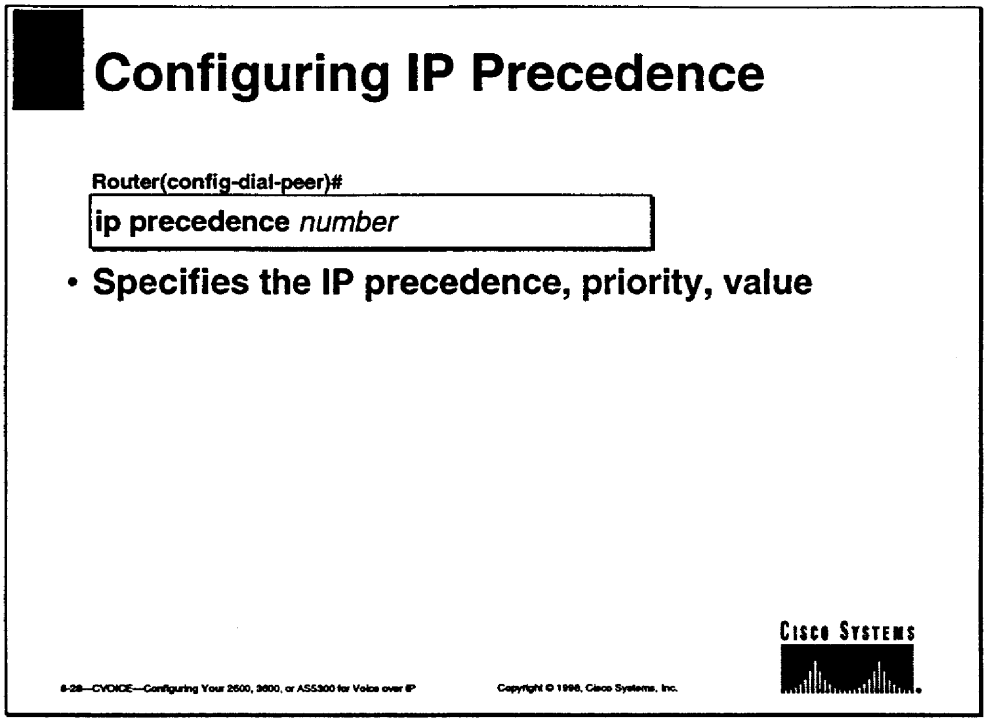
Note SNMP is a protocol that provides a means to monitor and control network devices and to manage configurations, statistics collection, performance, and security in IP networks.

When a dial peer is used, the Cisco IOS software reserves a certain amount of bandwidth so that the selected quality of service can be provided. The software uses RSVP to request quality of service guarantees from the network. To select the most appropriate quality of service value for this command, you will need to be familiar with the amount of traffic your connection supports and what kind of impact you are willing to have on it.

To generate an SNMP event, confirm that you are in VoIP dial peer configuration mode. Then use the **acc-qos {best-effort | controlled-load | guaranteed-delay}** command to request an SNMP event if the quality of service for a specified dial peer drops below the specified level.

Reference: Descriptions of the QoS levels are the same as those in the **req-qos** command. See the previous **req-qos** command table for descriptions.

Note RSVP reservations are only one-way. If you configure RSVP, the VoIP dial peers on both ends of the connection must be configured for RSVP.



Configuring IP Precedence

Instead of requesting best-effort, controlled-load, or guaranteed-delay levels, you may wish to forward packets by prioritizing classes of service. Use the **ip precedence** command to prioritize voice packets over other packets. This command will set the IP precedence or priority for the packets sent by the dial peer.

Note To enable IP precedence, a queuing algorithm like WFQ must be enabled.

---

ip precedence Command

number	Integer specifying the IP precedence value. Valid entries are 0 to 7. A value of 0 means that no precedence (priority) has been set.

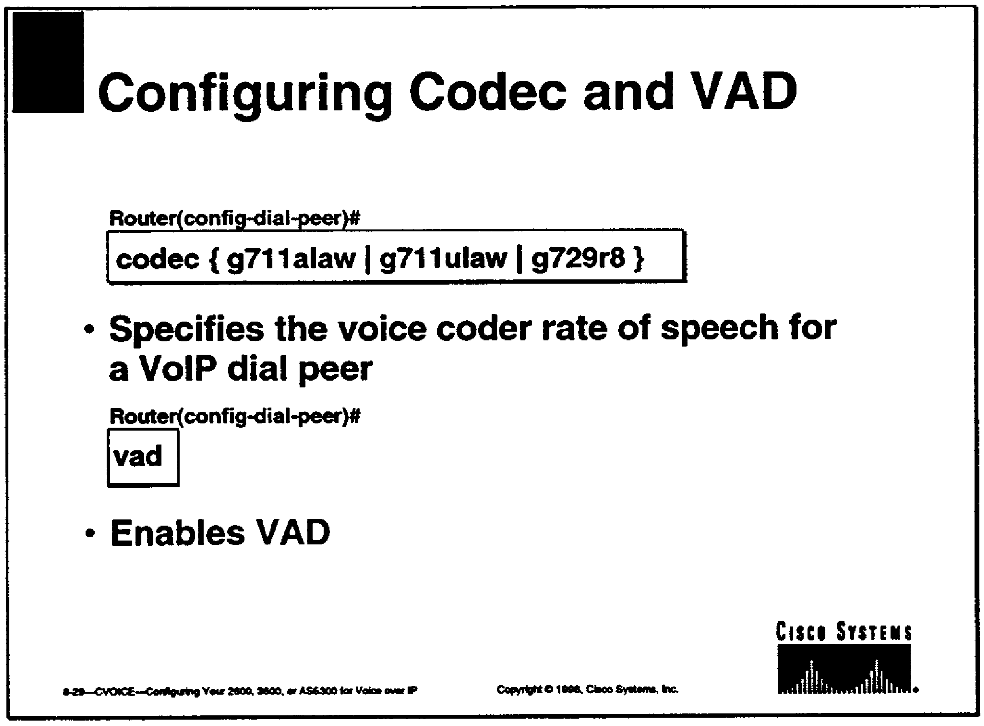
Use the **ip precedence** command to configure the value set in the IP precedence field when voice data packets are sent over the IP network. This command should be used if the IP link utilization is high and the QoS for voice packets needs to have a higher priority than other IP packets. The **ip precedence** command should also be used if RSVP is not enabled and the user would like to give voice packets a higher priority over the IP data traffic.

The available IP precedence settings are:

routine	Set routine precedence (0)
priority	Set priority precedence (1)
immediate	Set immediate precedence (2)
flash	Set flash precedence (3)
flash-override	Set flash-override precedence (4)
critical	Set critical precedence (5)
internet	Set internetwork control precedence (6)
network	Set network control precedence (7)

never use 6 or 7

Setting 0 is the lowest priority and 7 is the highest. IP precedence settings 6 and 7 are reserved for network control information. So, the IP precedence for delay-sensitive applications like voice should be set to a higher setting like 4 or 5.



Configuring Codec and VAD

Codec and VAD for a dial peer determine how much bandwidth the voice session uses. Codec, typically used to transform analog signals into a digital bit stream and digital signals back into analog signals, specifies the voice coder rate of speech for a dial peer.

Caution *When configuring the 2600, 3600, and AS5300 series of routers for VoIP, codec and VAD are configured on the dial peer. If you are configuring the MC3810 series router for VoFR or VoATM, codec and VAD are configured on the voice port. The codec options are also different on the MC3810. See Chapter 5 for more information on configuring codec and VAD on MC3810s.*

To specify the voice coder rate of speech for a dial peer, use the **codec** dial peer configuration command. The codec choices are in the following table.

codec Command

g711alaw	G.711 A-law 64,000 bps
g711ulaw	G.711 Mu-law 64,000 bps
g729r8	G.729 8000 bps

For toll quality, use **g711alaw** or **g711ulaw**. These values provide high-quality voice transmission but use a significant amount of bandwidth. For almost toll quality and a significant savings in bandwidth, use the **g729r8** option.

The default for the **codec** command is **g729r8**. This level of quality is acceptable most of the time unless you are operating in a high bandwidth environment and voice quality is of the highest importance.

To enable silence suppression and VAD for calls using a dial peer, use the **vad** dial peer configuration command. With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth. Enable VAD if bandwidth requirements are an issue. Disable VAD with **no vad** if you are operating in a high bandwidth network and voice quality is of the highest importance.

Note Enable VAD with the **vad** command. Disable it with the **no vad** command. The **vad** command has no other arguments, options, or keywords associated with it.

---

show dial-peer voice Output on VoIP Dial Peer

```
VoiceOverIpPeer110
        tag = 110, destination-pattern = `1101101....',
        answer-address = `',
        group = 110, Admin state is up, Operation state is up
        type = voip, session-target = `ipv4:10.1.1.1',
        ip precedence: 0        UDP checksum = disabled
        session-protocol = cisco, req-qos = guaranteed,
        acc-qos = guaranteed,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10, Icpif = 30,
        VAD = enabled, Poor QOV Trap = disabled
        Connect Time = 44739, Charged Units = 0
        Successful Calls = 35, Failed Calls = 0
        Accepted Calls = 35, Refused Calls = 0
        Last Disconnect Cause is "10  "
        Last Disconnect Text is "normal call clearing."
        Last Setup Time = 34461803
```

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show dial-peer voice Output on VoIP Dial Peer

You can check your new VoIP dial peer configurations with the same **show dial-peer voice** command you learned in Chapter 5. To verify new configurations, perform the following tasks:

- You can use the **show dial-peer voice** command to verify that the data configured is correct. Use this command to display a specific dial peer or to display all configured dial peers. See the graphic for configured output. The optimization features you learned are highlighted.
- If you configured a **codec** value, there can be a problem if VoIP dial peers have incompatible codec values. Make sure that VoIP peers at both ends of the connection have been configured with the same codec value.

Configuring Dial Peers Procedure

To configure dial peers, perform the following tasks:

Step 1 Enter the dial peer configuration mode and specify the POTS dial peer you wish to configure.

Step 2 Set the destination pattern of the POTS peer.

Step 3 Specify the voice port the POTS peer is connected to.

Step 4 Exit the dial peer configuration mode for the specific POTS peer.

Step 5 Repeat for all POTS dial peers.

Step 6 Enter the dial peer configuration mode and specify the VoIP dial peer you wish to configure.

Step 7 Set the destination pattern of the VoIP peer.

Step 8 Specify the session target of the peer you wish to attach to.

Step 9 Optimize the VoIP peer with RSVP, IP precedence, codec, and VAD.

Step 10 Exit VoIP dial peer configuration mode.

Step 11 Repeat for all VoIP dial peers.

Step 12 In the global configuration mode, configure number expansion to allow for prefix dialing.

Step 13 Verify proper configuration with the **show** commands.

Configuring Number Expansion

Router(config)#

num-exp *extension-number expanded-number*

- **Allows you to dial an extension only on a PBX**

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Configuring Number Expansion

In most corporate environments, office PBXs are usually configured so the user can dial a local call (within the same PBX) by dialing the extension only. For example, you may call another telephone number, 1 (408) 555-0000, within the PBX with the four-digit extension 0000 or the five-digit extension 50000.

You can provide the same shortcut on a Voice over IP network by using the **num-exp** global configuration command. This command tells the router to expand a particular sequence of dialed numbers into a complete telephone number (destination pattern).

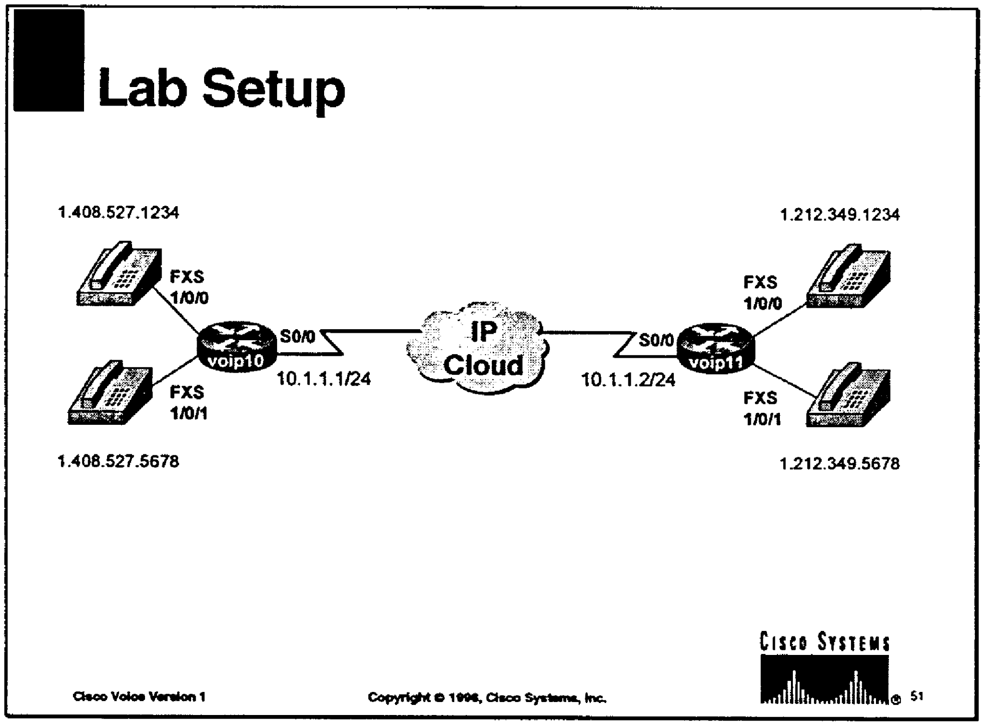
Note Remember, you can use the "." as a wildcard to replace a digit. For example, to expand 50000 into 1 (408) 555-0000, enter the **num-exp 50000 14085550000** command. To expand all the numbers with the 408 area code and 555 prefix on the PBX, enter **num-exp 5.... 1408555.....**

Lab Exercise: Establishing a VoIP Call between Two 2600 or 3600 Routers

Use the following laboratory exercises to practice what you have learned in this lesson.

Lab Setup

Use the following information to prepare for this exercise. The graphic illustrates the lab configuration for group 1. Use information supplied in the lab setup table to make router-specific connections.



Group	Routers	VoIP Dial Peer Tag	Your Router's IP Address	POTS Dial Peer Tags and Phone Number	POTS Dial Peer Tags and Ports	DLCI
Group 1	voip10	1000	10.1.1.1/24	101–14085271234 102–14085275678	101–1/0/0 102–1/0/1	100
	voip11	1100	10.1.1.2/24	111–12123491234 112–12123495678	111–1/0/0 112–1/0/1	110
Group 2	voip12	1200	10.1.1.3/24	121–12029651234 122–12029655678	121–1/0/0 122–1/0/1	120
	voip13	1300	10.1.1.4/24	131–13034691234 132–13034695678	131–1/0/0 132–1/0/1	130
Group 3	voip14	1400	10.1.1.5/24	141–14152831234 142–14152835678	141–1/0/0 142–1/0/1	140
	voip15	1500	10.1.1.6/24	151–15036431234 152–15036435678	151–1/0/0 152–1/0/1	150
Group 4	voip16	1600	10.1.1.7/24	161–16022921234 162–16022925678	161–1/0/0 162–1/0/1	160
	voip17	1700	10.1.1.8/24	171–17143421234 172–17143425678	171–1/0/0 172–1/0/1	170

The instructions represent commands specific to router voip10. From the global configuration mode, perform the following tasks:

Step 1 Name the router with the **hostname voip10** command.

Step 2 Enter router configuration mode and turn on RIP with the **router rip** command.

Step 3 Specify the network by typing the **network 10.0.0.0** command.

Step 4 Exit the router configuration mode with the **end** command.

Step 5 Copy the configurations to memory with the **copy running-config startup-config** command.

Scenario

Assume you are a network technician who has been given the task to connect two remote offices. These remote offices are networked using Frame Relay. Use the existing Frame Relay network, your 2600 or 3600 router, and FXS VICs to place a VoIP call.

Directions

Perform the following tasks for each router.

Exercise: Configuring and Verifying Frame Relay to Carry VoIP Packets

To configure Frame Relay WAN services, perform the following tasks:

Step 1 Enter the global configuration mode using the **configure terminal** command.

Step 2 Enter the interface configuration mode and prepare to configure the WAN serial interface 0/0 using the **interface serial 0/0** command.

Step 3 You will only want to specify the IP address for the subinterface. To make certain no IP address is already configured on your serial interface, enter **no ip address.**

Step 4 Set encapsulation to Frame Relay by entering **encapsulation frame-relay**.

Step 5 Set the MTU size to 300 bytes by entering **MTU 300.**

Step 6 Enable weighted fair queuing and set it to the default values by typing the **fair-queue** command.

Step 7 Enable RTP header compression by entering **frame-relay ip rtp header-compression.**

Step 8 Activate your serial interface with the **no shutdown** command.

Step 9 Now you are ready to configure the subinterface. Prepare to configure the Frame Relay subinterface to run VoIP by entering the **interface serial 0/0.1 multipoint** command.

Step 10 Specify the IP address in the subinterface by entering **ip address 10.1.1.1 255.255.255.0.** (Refer to the lab setup table for your IP address.)

Step 11 Specify the bandwidth as 64 kbps with the **bandwidth 64** command.

Step 12 Enable RSVP to the default value of 75 percent with the **ip rsvp bandwidth** command.

Step 13 Enable traffic shaping with the **traffic-shape rate 32000 4000 1000** command.

Step 14 Enable RTP header compression on the subinterface with the **frame-relay ip rtp header-compression** command.

Step 15 Identify the logical connection to router voip11 with the **frame-relay interface-dlci 110** command. (A Frame Relay switch, simulating your IP network, has been configured so anyone establishing a logical connection from another router to voip10 uses DLCI 100, any connecting to voip11 uses DLCI 110, and so forth.)

Step 16 Activate your serial interface with the **no shutdown** command.

Step 17 Exit all configuration modes by entering **end.**

Step 18 Save your configurations with the **copy run start** command.

Step 19 Enter the **show running-config** command to confirm proper configuration.

Verify that you configured your router correctly Your output should resemble the following:

```
voip10#show running-config
!<Output Omitted>
interface Serial0/0
 mtu 300
 no ip address
 ip rsvp bandwidth 1158 1158
 encapsulation frame-relay
 frame-relay ip rtp header-compression
!
interface Serial0/0.1 multipoint
 mtu 300
 ip address 10.1.1.1 255.255.255.0
 ip rsvp bandwidth 48 48
 bandwidth 64
 traffic-shape rate 32000 4000 1000 1000
 frame-relay interface-dlci 110
 frame-relay ip rtp header-compression
!<Output Omitted>
```

Step 20 Once everyone in the group has configured the interface, **ping** each other to confirm connectivity.

Exercise: Configuring POTS Dial Peers

To configure the POTS dial peers, perform the following tasks from the global configuration mode:

Step 1 Enter the local dial peer configuration mode and prepare to configure the local dial peer using the **dial-peer voice 101 pots** command.

Step 2 Specify the phone number using the **destination-pattern 14085271234** command.

Step 3 Associate the local dial peer with voice port 0 using the **port 1/0/0** command.

Step 4 Exit the local dial peer configuration mode using the **exit** command.

Step 5 Enter the local dial peer configuration mode and prepare to configure the local dial peer using the **dial-peer voice 102 pots** command.

Step 6 Repeat steps 2 through 4 of this exercise, making the specific configurations to your telephony device attached to port 1/0/1.

Step 7 Exit configuration modes with the **end** command and verify proper POTS dial peer configurations with the **show dial-peer voice** command you used in Chapter 5's lab exercise.

Step 8 Lift the receiver and check for dial tone. Key in phone numbers to check for DTMF detection.

Step 9 Place calls between the two telephones.

Exercise: Configuring and Optimizing VoIP Dial Peers

To configure the VoIP dial peers, perform the following tasks from the global configuration mode:

Step 1 Enter the VoIP dial peer configuration mode and prepare to configure the VoIP dial peer from voip10 to voip11 using the **dial-peer voice 1100 voip** command. (Use the tag numbers assigned in the lab. If you are creating a VoIP dial peer to contact voip10, use tag number 1000. If you are creating a VoIP dial peer to contact voip11, use tag number 1100, and so forth.)

Step 2 Specify the VoIP dial peer you wish to call using the **destination-pattern 1212349....** command.

Note Remember, the "...." are wildcard placeholders so you can call anyone on voip11 whose number starts with 1212349 plus four additional numbers. If you use the telephone numbers in this exercise and the wildcard placeholders, voip10 connects to both telephony devices on voip11, for example, with one VoIP dial peer.

Step 3 Designate a network-specific address for the dial peer you are routing by using the **session target ipv4:10.1.1.2** command.

Step 4 Exit all configuration modes by entering **end**.

Step 5 Save configurations with the **copy run start** command.

Step 6 Verify proper configuration by entering the **show dial-peer voice** command. Your output should look like the following:

```
<Omitted POTS dial peer output.>

VoiceOverIpPeer1100
        tag = 1100, destination-pattern = `+1212349....',
        answer-address = `',
        group = 1100, Admin state is up, Operation state is up
        type = voip, session-target = `ipv4:10.1.1.2',
        ip precedence: 0        UDP checksum = disabled
        session-protocol = cisco, req-qos = best-effort,
        acc-qos = best-effort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10, Icpif = 30,
        VAD = enabled, Poor QOV Trap = disabled
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        lLast Disconnect Cause is ""
        Last Disconnect Text is ""
        Last Setup Time =
```

Step 7 Once a group has finished configuring its routers, take turns placing calls within the group.

Note You can send data traffic with the voice traffic by entering **ping** in the privileged EXEC mode and following the terminal's prompts. Use the defaults, in brackets, except when asked for the repeat count. When asked for the repeat count, key in a higher number, like 300.

Step 8 Reenter the dial peer configuration mode with the **dial-peer voice 1100 voip** command and specify the desired QoS for the VoIP dial peer as guaranteed-delay with the **req-qos guaranteed-delay** command.

Step 9 Make another call while sending data traffic across the network using the **ping** command.

Step 10 Specify the voice coder rate of speech for the VoIP dial peer as g711ulaw with the **codec g711ulaw** command.

Step 11 Take turns within your group making more calls to the telephone you configured your router to call.

Q1) What happened, and why?

Step 12 Reset codec to the default value with the **no codec** command.

Step 13 Enable VAD for the calls using the **vad** command.

Step 14 Exit the VoIP dial peer configuration mode using **end**.

Step 15 Use the **show dial-peer voice** command to display all configured dial peers. This time, pay particular attention to the QoS configurations.

Step 16 Enter global configuration mode with the **config terminal** command.

Step 17 Prepare to allow prefix dialing with the **num-exp 7.... 1408527....** command. After configuring number expansion place calls between your local telephones using the five-digit extension (i.e., for telephone 14085271234 on voip10, the extension is 71234).

Q2) What would happen if the first five digits of an external number were the same as the last five digits of a telephony device connected to your router?

Step 18 If you finish early and your instructor allows, team with a new group of students that is configuring a different router and configure dial peers to call them.

Note To complete step 18, you must also identify the logical connection in the subinterface serial 0/0.1 with the **frame-relay interface-dlci** *number* command.

Completion Criteria

You have completed the lab when you have successfully called members of your group and placed calls to the telephones attached to the same router.

Answers

Q1) What happened and why?

You are experiencing poor speech quality due to bandwidth congestion. The codec you configured is not compressed.

Q2) What would happen if the first five digits of an external number were the same as the last five digits of a telephony device connected to your router?

The call would go through to the local telephony device directly attached to your router.

Summary

Summary

In this chapter you learned how to complete the following tasks:

- **Identify the issues that must be addressed to transport delay sensitive traffic like voice over an IP network**
- **Configure the host and WAN with RSVP, weighted fair queuing, and RTP header compression to carry real-time traffic like voice**
- **Improve voice quality by setting the quality of service options on the VoIP dial peers**
- **Verify the configurations necessary for a VoIP call**

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8-42—CVOICE—Configuring Your 2600, 3600, or AS5300 for Voice over IP

Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

A Six-Step Network Design Process

Objectives

Objectives

- Design, configure, integrate, and optimize voice in remote branch and regional offices by using technology that combines voice and data transmission over Frame Relay, ATM, and IP connections
- Appraise existing branch and regional office voice traffic, delay budget, and services, and choose the optimum transmission method for voice traffic: Frame Relay, ATM, or IP

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Objectives (cont.)

- **Analyze existing voice hardware/software, and select which Cisco multiservice access device (2600, 3600 or MC3810) would best serve the needs**
- **Connect branch and regional offices with Voice over FR/ATM/IP**
- **Cover case studies of networks using VoFR, VoATM, and VoIP**

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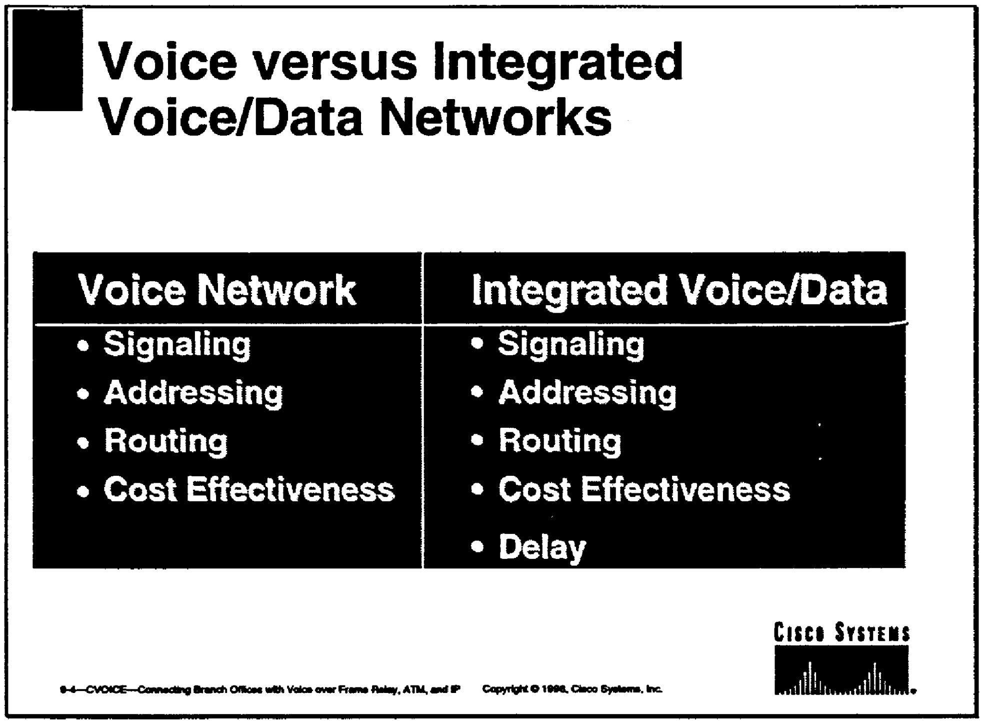
9-3—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

Introduction

This chapter presents a six-step network design process to help students perform an audit of their existing voice equipment, set objectives and goals for their network, evaluate available technologies, and formulate a technical network design using Cisco's multiservice access devices of the 3620 and 3810.

Chapter Content

This chapter covers all the essential elements to connect branch and regional offices with Voice over FR/ATM/IP. The chapter covers: the concepts of data communications that will affect voice data integration; how to evaluate voice traffic for Frame Relay, ATM, or IP; how to plan branch office connectivity using a six-step process for designing an integrated voice and data network; and some real-world examples and solutions.



Voice versus Integrated Voice/Data Networks

This section starts by discussing characteristics common to both voice networks and integrated voice/data networks, plus the one characteristic that is not: delay.

Cost efficiencies gained by an integrated network are of little value if they come at the expense of unsatisfactory voice quality. The shared resources must be properly engineered to accommodate the critical requirements of both traffic types.

Voice traffic is not necessarily delay sensitive; for example, fax, voice mail, and modem traffic need not be handled in real time. Adding fax and modem traffic to a data network may be justification alone for creating a voice over data network.

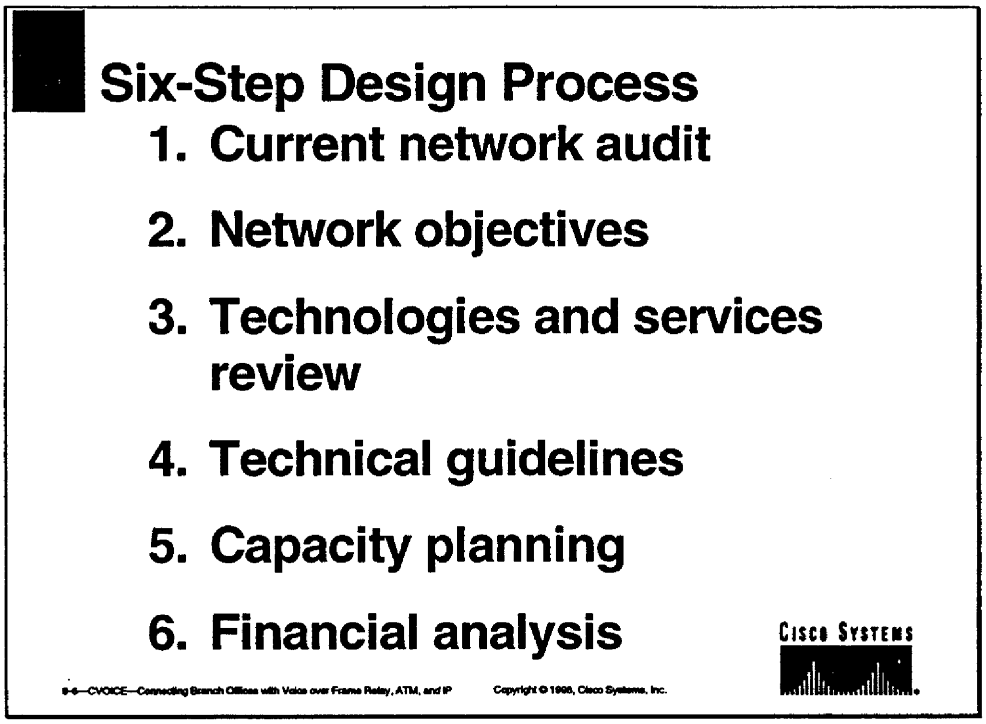
In a voice network, delay is not a major issue. In existing voice networks, delay is recognized as a factor affecting voice quality. The largest contributor to delay is propagation delay, which is the period of time required for the electronic signal carrying voice to travel the distance across the physical network medium. When distances are short, propagation delay is negligible. For example, a phone call from Munich to Frankfurt has a 1-ms delay, but a call from Munich to Los Angeles has a much higher delay, 32 ms.

However, in an integrated voice and data network, transmission priorities can cause delay. The addition of equipment for "packetizing" voice, and the delays inherent in the data network, make managing delay a critical factor in integrating voice and data networks.

Current voice and data networks utilize signaling, addressing, and routing to establish communications between end users:

- Voice networks have been designed to limit delay and delay variation.
- Data traffic, because it is not as delay sensitive as voice traffic, has been designed with throughput efficiency in mind.

These two different goals can be accommodated in an integrated voice/data network by reducing delay and delay variation as delay sensitive and delay insensitive traffic is mixed onto the same network. The next section describes in detail the six-step network design process for analyzing your current network and for assisting you in creating an integrated voice/data network.



Six-Step Design Process

The six-step network design process starts with an evaluation of your current network, helps set objectives and goals, evaluates available technologies, provides a technical design engineered for the unique characteristics of voice communications, and ends with a financial analysis.

Step 1: Current Network Audit

- **Existing equipment**
  - **Operating costs**
  - **Capabilities**
- **Facility costs**
- **Upcoming projects**
- **Service quality**
- **Traffic study**

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9-7—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

Step 1: Current Network Audit

The first step in designing a network is to take stock of what currently exists. Look at the existing equipment and evaluate its capabilities and operating costs.

Determine your existing facility costs and whether the networks will meet your planned voice and data needs.

In addition, identify upcoming projects that will be required in the network—and determine their impact on the network as best you can.

What has been the service quality of both voice and data? Does it need improving?

A traffic study may be necessary to look at current traffic patterns. The traffic survey will determine if some links can be removed, while others may need to be increased.

Step 2: Set Network Objectives

Identify:

- **Dominant traffic type**
- **Required voice quality**
- **Performance**
- **Cost**

Step 2: Set Network Objectives

Once you understand your current networking situation, the next step is to set the integrated network objectives.

Determine the dominant traffic type you expect to carry on the integrated network. Also consider how closely you wish to tie voice and data functionality. Estimating these points will help you select the appropriate technologies.

Setting voice quality objectives will help you design networks with acceptable amounts of delay and compression.

Determine the level of performance expected out of the integrated network. Is it permissible to allow the addition of voice to an existing data network to slightly degrade the performance of the data traffic? If not, the data network will require some degree of upgrading.

When setting financial objectives you may ask what return on investment or payback period you are trying to meet.

Step 3: Technologies and Services Review

- Voice over Frame Relay
- Voice over ATM
- Voice over IP

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9-8—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

Step 3: Technologies and Service Review

Once you have set your network objectives, step 3 is to evaluate available technologies and services, and pick the model and technology that best meets your objectives.

Integrating voice and data networks should first include an evaluation of these three technologies: Voice over Frame Relay, ATM, and IP. These three topics have already been covered in the preceding chapters.

Voice over Data Summary— The Best Technology Fit

- Frame Relay
- ATM
- IP

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The different technology alternatives are as follows:

- Frame Relay. FRF.11 has provisions for supporting voice over a public Frame Relay network. It is also relatively inexpensive and quite common in many parts of the world. Frame Relay may provide SVCs services and support QoS in the future, but its lack of sophisticated signaling, addressing, and routing functionality will prevent users from moving beyond the transport model.
- ATM. ATM is connection oriented. It was designed to handle time-sensitive traffic, like voice. Its signaling, addressing, and routing allow us to build a network that follows the translation model. The routing function in particular is quite robust, allowing users to build connections based on meeting a certain delay and delay variation.
- IP. IP is connectionless. Development in areas of prioritization, resource reservation, and packet fragmentation are all relatively new.

IP, like ATM, has robust signaling, addressing, and routing functionality, which makes the translation model a possibility. The most compelling argument for IP is its integration with current data applications.

Step 4: Technical Guidelines

- Voice quality guidelines
- Voice delay guidelines

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Step 4: Technologies Guidelines

Steps 1 through 3 of the design process included a recommendation to review existing network infrastructure, as well as current available voice over data network technologies.

In step 4 through 6 we will discuss technical guidelines, capacity planning, and financial analysis.

It is now time to look at the technical design guidelines that should be considered.

Step 4 will review factors that could impact voice quality and provide guidelines for you to follow. We also will provide further direction on how much delay you can expect as you plan your network.

Voice Quality Factors

- **Coding**
- **Compression**
- **Multiple conversions**
- **Delay/delay variation**
- **Line quality**

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Coding and voice compression methods are the first factors that could affect voice quality.

We use the term coding to refer to the entire process of converting between an analog voice signal and its digital counterpart. Pulse Code Modulation (PCM) is the standard for representing voice as a 64-kbps bit stream.

Compression is the method of reducing the amount of digital information below the traditional 64 kbps. Advances in technology have greatly improved the quality of compressed voice and resulted in a spectrum of compression algorithms.

Multiple conversions from analog to digital or changes in compression schemes will affect the quality of the original voice signal.

Two common network characteristics that affect quality are delay and delay variation, also known as digital jitter.

Delay can cause two potential impairments to speech:

- Long delays in conversation cause the receiver to start to talk before the sender is finished.

  Delay exacerbates the problem of echo, which is the reflection of the original signal to the sender. Echo is indiscernible under low delay conditions. It is noticeable to the point of distraction when the delay becomes too great.

- Digital jitter causes gaps in the speech pattern that cause the quality of voice to be "jerky."

  Line quality also affects voice quality. The details of how line quality affects voice quality are outside the scope of this course.

Voice Compression

Compression Type	Data Rate
G.711—PCM	64 kbps
G.726—ADPCM	32, 24, 16 kbps
G.728—LD CELP	16 kbps
G.729—CSA-CELP	8 kbps

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When dealing with voice compression, a trade-off occurs between the level of voice quality delivered and the amount of bandwidth savings achieved. Through the use of voice compression and therefore the optimization of bandwidth, significant monthly cost savings are possible.

PCM is the most common form of digital voice coding available. PCM runs at 64 kbps, provides no compression, and therefore no opportunity for bandwidth savings. It is also the sound quality benchmark against which we compare voice compression schemes.

Adaptive Differential Pulse Code Modulation (ADPCM) provides various levels of compression. The quality difference between 32-kbps ADPCM and 64-kbps PCM is virtually imperceptible. Some fidelity is lost as the compression increases. Depending upon traffic mix, cost savings generally run 25 percent for 32-kbps ADPCM, 30 percent for 24-kbps ADPCM, and 35 percent for 16-kbps ADPCM.

Low delay code-excited linear-predictive (LD CELP) algorithms model the human voice. Dependent upon traffic mix, cost savings generally run 35 percent for 16-kbps LD CELP.

Conjugate structure algebraic code-excited linear predictive (CSA-CELP) provides eight times the bandwidth savings over PCM and four times that of 32-kbps ADPCM. CSA-CELP is a more recently developed algorithm modeled after the human voice. It delivers quality comparable to LD CELP and 24-kbps ADPCM. Dependent upon traffic mix, cost savings generally run 40 percent for 8-kbps CSA-CELP.

Voice Quality Guidelines

Compression Method	MOS Score	Delay (ms)
PCM (G.711)	4.1	0.75
32K ADPCM (G.726)	3.85	1
16K LD CELP (G.728)	3.61	3–5
8K CSA-CELP (G.729)	3.92	10
8K CSA-CELP (G.729a)	3.9	10

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9-14—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

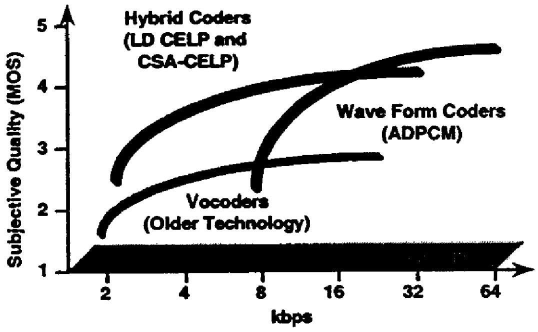
The table provides a guideline for you to use in measuring voice quality.

Mean opinion score (MOS) is a widely used subjective measure of voice quality. Scores of 4 to 5 are deemed toll quality, 3 to 4 communication quality, and less than 3 synthetic quality.

The graphic shows today's MOS for varying compression algorithms such as ADPCM and CSA-CELP. These high MOS scores are a result of improvements in the algorithms along with dramatic increases in the power of the digital signal processors (DSPs).

An opportunity exists to integrate voice and data networks while maintaining high-quality voice. Note the trade-off between bandwidth compression and higher delay. When you design networks, bandwidth compression and higher delay must be balanced to ensure overall voice quality.

Voice Quality Guidelines



Score	Quality	Description of Impairment
5	Excellent	Imperceptible
4	Good	Just Perceptible, Not Annoying
3	Fair	Perceptible and Slightly Annoying
2	Poor	Annoying but Not Objectionable
1	Bad	Very Annoying and Objectionable

Source: A.M. Kondoz, *Digital Speech Coding for Low Bit-Rate Communications Systems*, 1995

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9-15—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

Though a little out of date given the speed of change, the upper part of this graphic is another way of looking at voice quality guidelines as of 1995. Since that time, the MOS scores for hybrid coders have risen another 0.2 of a point because of improvements in the technology.

The table in the lower part of the graphic further explains the MOS 1 through 5 ratings.

Voice Delay Guidelines

One-Way Delay (ms)	Description
0–150	Acceptable for most user applications
150–400	Acceptable provided that administrations are aware of the transmission time impact on the transmission quality of user applications
400+	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded

ITU's G.114 Recommendation

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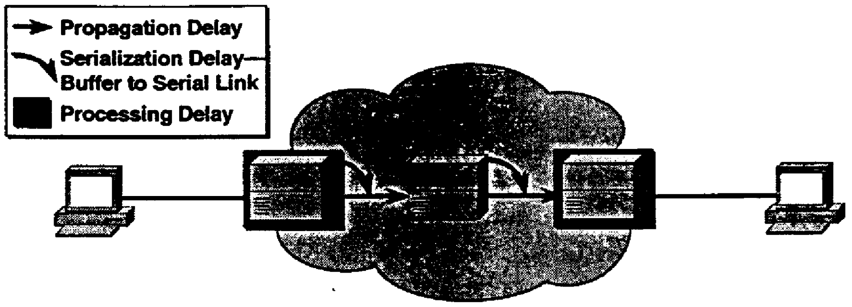
9-17—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

This table summarizes the ITU's recommendations for voice delay guidelines. You can see that below 150 ms is considered acceptable for most applications. Delays ranging from 150 to 400 ms are also acceptable subject to current voice quality.

For example, a 200-ms delay from Chicago to New York City is unacceptable, given experiences with public networks.

However, a 200-ms delay from Chicago to Singapore will likely be acceptable given current conditions. Furthermore, higher delays may be acceptable if cost savings are taken into account.

Fixed Delay Components



- **Propagation—six microseconds per kilometer**
- **Serialization**
- **Processing**
  - **Coding/compression/decompression/decoding**
  - **Packetization**

The delay components on this graphic are fixed in nature and add very little to delay variation.

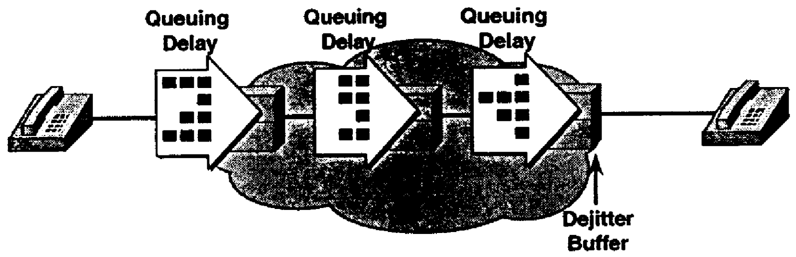
Propagation delay is based on the total distance between source and destination. As a planning number, 6 microseconds per kilometer can be used.

Serialization is the process of placing bits on the circuit. The higher the circuit speed, the less time it takes to place the bits on the circuit. So, the higher the speed, the less serialization delay. For example, it takes 125 microseconds to place one byte on a 64-kbps circuit. The same byte placed on an OC-3 circuit will take half a microsecond.

Processing delays can be broken down as follows:

- Coding, compression, decompression, and decoding delay will be based on the algorithm employed. It is important to note that these functions can be performed in either hardware or software. Using specialized hardware such as DSPs will dramatically improve the quality and reduce the delay associated with different voice compression schemes. Current voice over Internet products using software-based compression should not be confused with the practicality of carrying Voice over IP.
- Packetization delay is the process of holding the digital voice samples for placement into the payload until enough samples are collected to fill the packet or cell payload. To reduce excessive packetization delay associated with some compression schemes, partial packets could be sent.

Variable Delay Components



- **Queuing delay**
- **Dejitter buffers**
- **Variable packet sizes**

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9-18—CVOICE—Connecting Branch Offices with Voice over Frame Relay, ATM, and IP

The delay components depicted on this graphic are variable in nature and result in higher delay variation; they are also more controllable.

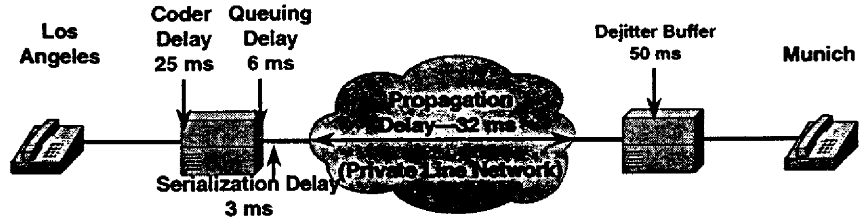
Queuing delay occurs when a packet is waiting for others to be serviced first on the trunk. This waiting time is statistically based on the arrival of traffic; hence, the more inputs, the more likely we will encounter contention for the trunk. It is also based on the ***size of the packet*** currently being serviced.

Dejitter buffers are used at the receiving end to smooth out delay variability and allow time for decoding and decompression. They also help on the first talk spurt to provide smooth playback of voice traffic. Setting these buffers too low will cause overflows and loss of data. Setting them too high will cause excessive delay.

In effect, a dejitter buffer reduces or eliminates delay variation by converting it to fixed delay.

The next few graphics will show how to calculate delay.

Calculate Delay Budget



	Fixed Delay	Variable Delay
Coder Delay G.729 (5 ms Look-Ahead)	5 ms	
Coder Delay G.729 (10 ms per Frame)	20 ms	
Packetization Delay—Included in Coder Delay		
Queuing Delay 64-kbps Trunk		6 ms
Serialization Delay 64-kbps Trunk	3 ms	
Propagation Delay (Private Lines)	32 ms	
Network Delay (e.g., Public Frame Relay Svc)		
Dejitter Buffer	50 ms	
Total	110 ms	

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Calculate Delay Budget

Now that the fixed and variable delay components have been identified, you can calculate your delay budget. The delay budget is the amount of delay permissible in your planned network while still holding to your voice quality objectives.

For the following example, assume that our delay budget is 200 ms. This example is of a private Frame Relay network running over leased lines. For a public service such as public Frame Relay you should contact the service provider for its guaranteed delay figures and use those figures in your delay budget.

We are filling a Frame Relay frame with two 10-byte packets. The coder delay for G.729 voice compression is an initial 5 ms for a look-ahead, plus 20 ms for the two 10-byte frames.

Packetization delay is the rate at which we fill a packet, which is typically governed by the speed at which voice samples are played out. Standard voice is transmitted at 64 kbps or one byte every 125 microseconds. The delay for this transmission is included in the coder delay.

Queuing delay is variable, and with a 64-kbps trunk will equal 3 ms per 20-byte packet already in queue. We used two voice packets in queue for a total queuing delay of 6 ms. However, this assumption will be variable, and is cumulative based on the number of devices in the network.

Serialization delay is the time it takes to place a 20-byte packet onto the 64-kbps trunk.

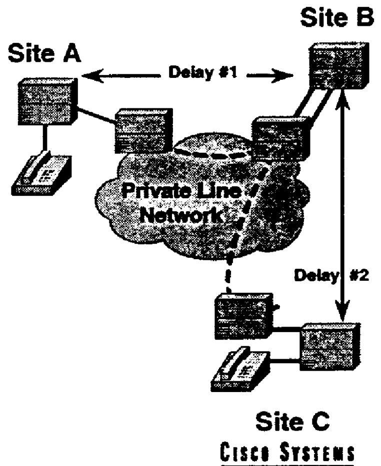
As calculated in an earlier slide, propagation delay on a 64-kbps private line from Los Angles to Munich is 32 ms.

The dejitter buffer was set to twice the coder delay as a general rule, which accounts for the decoding and decompression delay as well as any variable delays in the network. Our variable delay is only 6 ms, but a public packet network such as Frame Relay or the Internet can have much higher variable delays. It pays to be able to vary the size of the dejitter buffer.

Totaling up these figures we get a 110-ms delay, which is well within the delay guidelines set forth by the ITU and our planning number of 200 ms.

Calculate Delay Budget—Multiple Conversions

	Fixed Delay	Variable Delay
DELAY #1		
Coder Delay G.729	25 ms	
Packetization Delay (Included in Coder Delay)		
Queuing Delay 64-kbps Trunk		6 ms
Serialization Delay 64-kbps Trunk	3 ms	
Propagation Delay (Private Lines)	32 ms	
Dejitter Buffer	50 ms	
Tandem Switch	—	
Delay #1 Total	110 ms	



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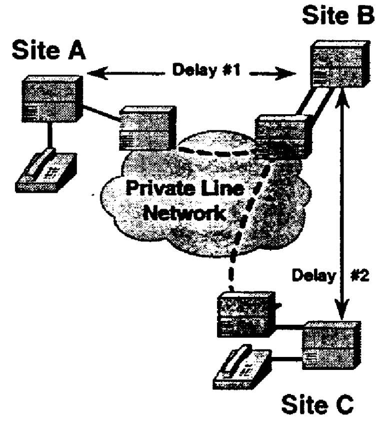
This example shows the delay effects of switching voice calls through a central site tandem PBX.

This example is of a private Frame Relay network running over leased lines. The speed of the site A to site B link is 64 kbps. The link from site B to site C is E1. The first hop is a duplicate of the first example. So the total for delay #1 is 110 ms.

Now we will look at Delay #2.

Calculate Delay Budget—Multiple Conversions (cont.)

	Fixed Delay	Variable Delay
DELAY #1 Total	110 ms	
DELAY #2		
Coder Delay G.729	25 ms	
Packetization Delay (Included in Coder Delay)		
Queuing Delay 2-mbps Trunk		0.2 ms
Serialization Delay 2-mbps Trunk	0.1 ms	
Propagation Delay (Private Lines)	5 ms	
Dejitter Buffer	50 ms	
Delay #2 Total	80 ms	
Total Delay	190 ms	



For delay #2, notice that the numbers are calculated in the same manner and total 80 ms. The delay is less because the distance is shorter from site B to site C and because the speed of the private line is 2 Mbps versus 64 kbps.

The total for both hops equals 190 ms, which is in the "acceptable provided administrations are aware" category as shown in the previous ITU recommendation chart, and also meets our planning delay budget of 200 ms.

These last two graphics show the impact of using a tandem PBX at site B. If you were to switch the voice without needing to break it out at the tandem PBX, the aggregate delay could be reduced by 75ms, lowering the total delay to 115 ms.

Note Delay associated with the tandem PBX itself has not been included in this estimate.

Technical Guidelines Summary

- **Balance: voice quality, delay, and bandwidth**
- **Determine acceptable delay and delay variation thresholds**
- **Calculate delay for the chosen model**
- **Avoid tandem (multiple) conversions**

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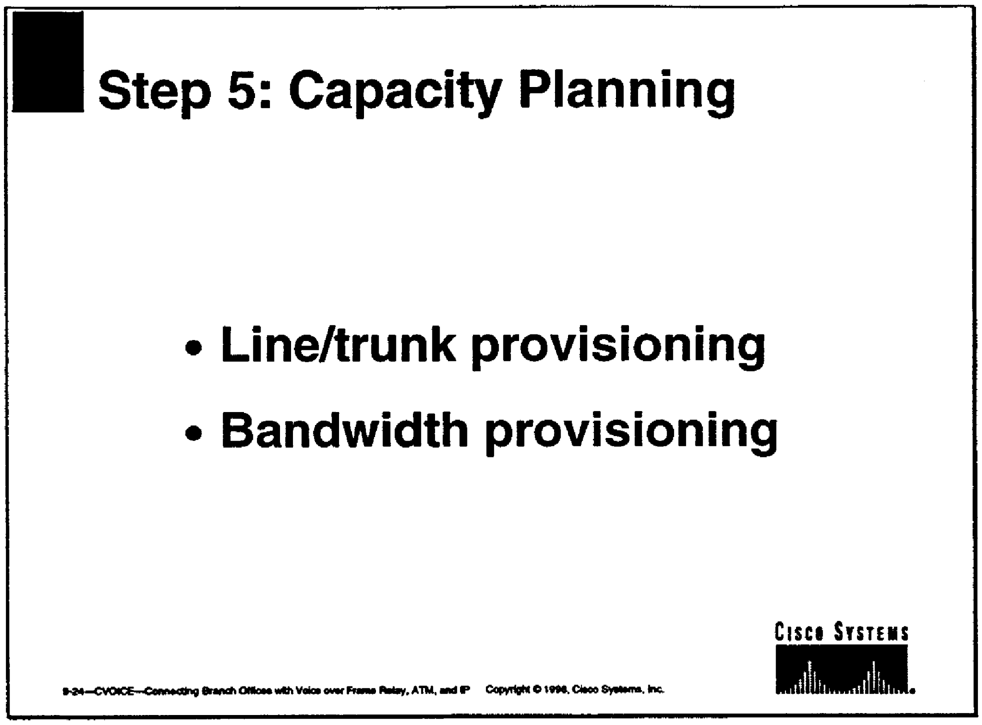
To summarize, step 4 discussed elements to consider when designing an integrated voice data network:

- Requirements
- Method for calculating budget delay
- Technical guidelines:
  - Balance voice quality, delay, and bandwidth
  - Determine acceptable delay and delay variation thresholds
  - Calculate delay for the chosen model
  - Avoid tandem (or multiple) conversions

As in all designs, a balance must be struck between quality and cost. Given the large MOS quality improvements in today's compression techniques, this balance is easier than ever.



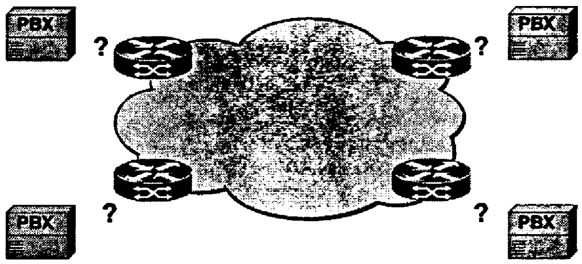
Step 5: Capacity Planning

Step 5 covers capacity planning for the integrated voice data network.

Line trunk provisioning is establishing the number of trunks from the PBX to the integrated voice data network.

Once we establish the number of trunks, the next step is to translate that to the required network bandwidth.

Provisioning—Trunking



- **Determine traffic volume and flow**
- **Calculate required number of trunks**
- **Grade of service**

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Provisioning—Trunking

The correct number of PBX or key system trunks will be determined by:

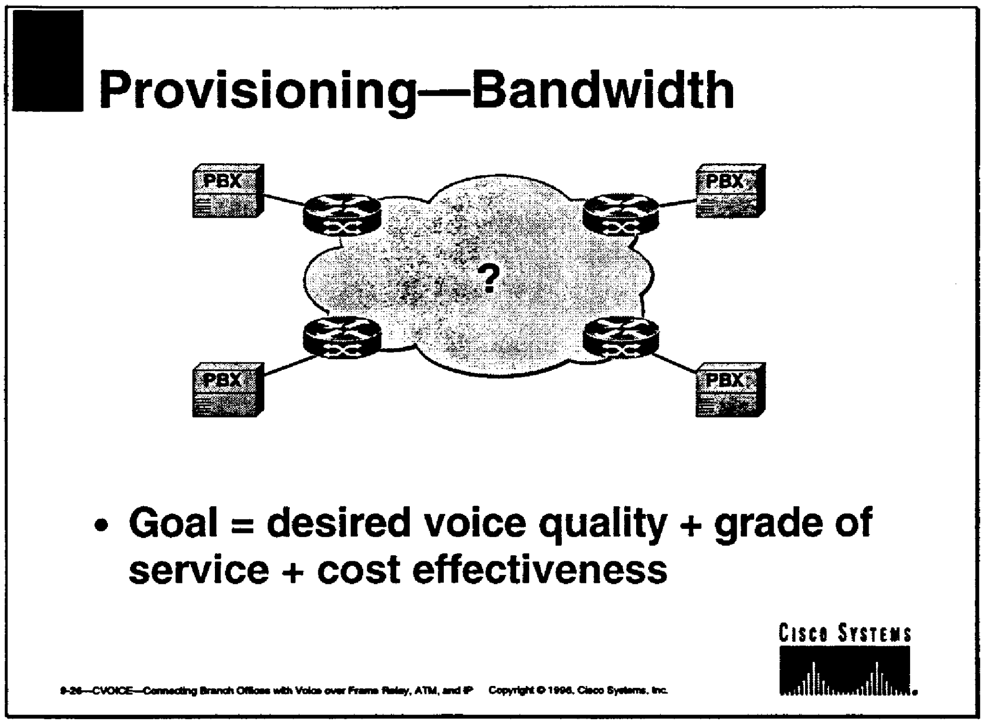
- Traffic volume and flow
- The selected grade of service (or blocking factor)
- Your objectives

Some organizations will simply move certain trunks from their current network to the integrated network. Others will take this opportunity to update their traffic engineering information and choose to conduct a traffic study. Either approach can work and is very familiar territory for voice engineering professionals.

The use of the transport or translate model can have an impact on the number of trunks simulated by the network. The transport network model matches a virtual connection for a tie line on a one-for-one basis. From a voice engineer's standpoint, nothing has changed.

However, the translate model uses the network to simulate a tandem PBX, thereby potentially reducing the number of trunks required.

Routing of calls over specific trunk groups in which one group could be the integrated voice data network is left to the voice engineer. One engineer may choose to use the network as the first option while another may use it as the last choice.



Provisioning—Bandwidth

Based on the proposed network design and the required number of trunks between locations, the required bandwidth can be calculated.

Bandwidth calculations should take into account:

- Compression
- Overhead
- Utilization

Each of these elements will vary, depending on which voice over data network technology is chosen.

Calculate the delay matrix between locations and confirm that the delay calculations meet the delay budget requirements. If they do not, adjust the bandwidth, or select a different voice over data network technology.

Now that the voice/data objectives have been set, a technology chosen, the capacity planning completed, and the trunks sized to properly handle the delay budget, it is time to perform the last step: financial analysis.

Step 6: Financial Analysis

- Cost/benefit analysis
- Financial justification
- Return on investment

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Step 6: Financial Analysis

Part of the step 6 financial analysis design process is to compare benefits against costs.

Once you have completed the previous five steps, you must determine if the cost is justified in relation to the benefits received. After working through some different aspects of financial analysis, the bottom line of return on investment can be calculated.

In order to demonstrate how the six-step design process can be used in real-world situations, case studies will be provided in the next section as an exercise.

Exercise: Case Studies

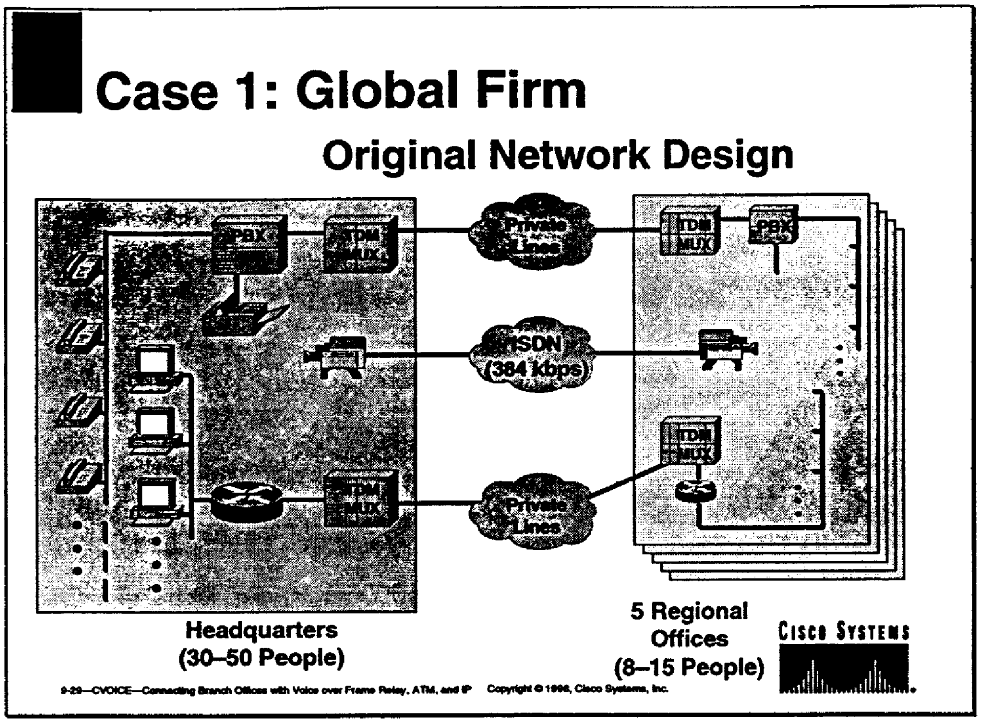
In this section, three different case studies are used to show the concepts covered in this chapter. By using three different technologies (Frame Relay, ATM, and IP), standard voice engineering principles, actual PSTN and VPN toll charges, and private line or Frame Relay service costs, students will see the financial benefits of the various solutions.

For all three case studies, the traffic volume, patterns, and assumptions have been held constant. Only the regional PSTN, Frame Relay, or private line prices change within each case.

Case 1: Global Firm

Scenario

A global firm has a corporate headquarters and five smaller offices using E1s, T1s, and fractional private lines. For historical reasons it has two separate private TDM-based networks: one transporting data traffic and the other transporting voice traffic. Additionally, a relatively new requirement, video conferencing, has been implemented using the public ISDN network at 384-kbps rates.



Case 1: Original Network Design

At headquarters, there are approximately 30 to 50 people working at the corporate site. In the five regional offices, there are 8 to 15 people.

Objective: Reduce costs for the network by merging the three networks as efficiently and cost-effectively as possible while maintaining voice quality. The company would also prefer to reduce reliance on tandem PBXs for switching its voice calls. Reducing reliance on PBXs would avoid the decompression/recompression cycle associated with the tandem PBX, allowing higher voice compression ratios and resulting in further reductions in bandwidth expenses.

Decision: After reviewing the different voice over data alternatives, an ATM network was selected because of the large capacity requirements, the video conferencing needs, and the intent for the network to add voice switching. ATM reduces the company's reliance on tandem PBXs.

Maintaining separate networks is very costly, particularly when the networks are very large. Focusing only on bandwidth costs, the following tables show the voice network and data network expenses, respectively.

Note The costs of the ISDN network were not captured for this exercise and consequently are not shown, although the network was sized to carry the video traffic.

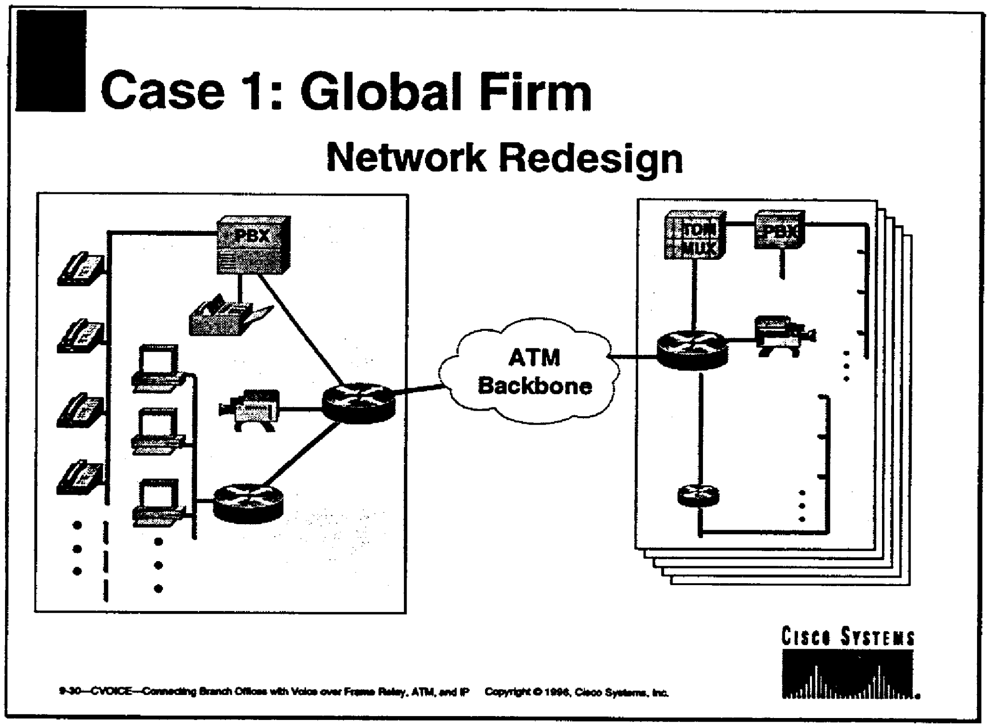
Case 1: Voice Backbone Configuration and Expense

Source	Destination	Quantity	Speed	Monthly Expense
London	Manchester	1	FT768K	$4460
Paris	Lyon	1	FT384K	$6516
Frankfurt	Cologne	1	F-1920	$10,314
Frankfurt	Munich	1	E1	$9156
Milan	Rome	1	FT512K	$13,338
London	Frankfurt	1	T1	$40,452
London	Philadelphia	1	E1	$47,950
London	Toronto	1	FT512K	$21,822
London	Dublin	1	FT256K	$8276
Amsterdam	Paris	1	FT512K	$21,672
Amsterdam	Zurich	1	FT512K	$17,179
Amsterdam	Philadelphia	1	F-1984	$48,581
Amsterdam	Rome	1	FT768K	$32,837
Amsterdam	Manchester	1	FT768K	$22,374
Amsterdam	Cologne	1	FT768K	$19,228
Amsterdam	Budapest	1	FT512K	$29,001
Amsterdam	Oslo	1	FT384K	$12,573
Frankfurt	Milan	1	FT768K	$30,967
Frankfurt	Copenhagen	1	FT384K	$17,377
Frankfurt	Zurich	1	FT384K	$17,362
Frankfurt	Lyon	1	FT512K	$20,259
Copenhagen	Oslo	1	FT256K	$5275
Philadelphia	Toronto	1	FT512K	$8829
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	FT768K	$19,228
TOTAL		**25**		**$507,119**

The following table displays the data backbone configuration and monthly expense.

Source	Destination	Quantity	Speed	Monthly Expense
Cologne	Frankfurt	1	F-1920	$10,314
Cologne	Munich	1	F-1920	$15,158
Paris	Lyon	1	FT256K	$4117
Frankfurt	Munich	1	E1	$9156
Milan	Rome	1	FT256K	$8927
Amsterdam	Philadelphia	1	F-1984	$38,581
Amsterdam	London	1	FT512K	$17,281
Amsterdam	Copenhagen	1	FT256K	$10,130
Amsterdam	Zurich	1	FT512K	$17,179
Amsterdam	Philadelphia	1	F-1984	$48,581
Amsterdam	Rome	1	FT256K	$15,785
Amsterdam	Manchester	1	FT512K	$17,281
Amsterdam	Budapest	1	FT512K	$29,001
Amsterdam	Oslo	1	FT256K	$8867
Philadelphia	Toronto	1	T1	$12,277
Cologne	Paris	1	FT256K	$12,269
Cologne	Zurich	1	F-1920	$41,382
Cologne	Lyon	1	FT256K	$12,269
London	Dublin	1	FT256K	$8276
Milan	Munich	1	FT256K	$18,017
Copehagen	Oslo	1	FT256K	$5275
Philadelphia	Toronto	1	T1	$12,277
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	FT768K	$19,228
TOTAL		**24**		**$413,721**

The combined monthly expense for both voice and data networks is $920,840. In the next section, the new, redesigned network is shown, including the features and the financial cost/benefit analysis.



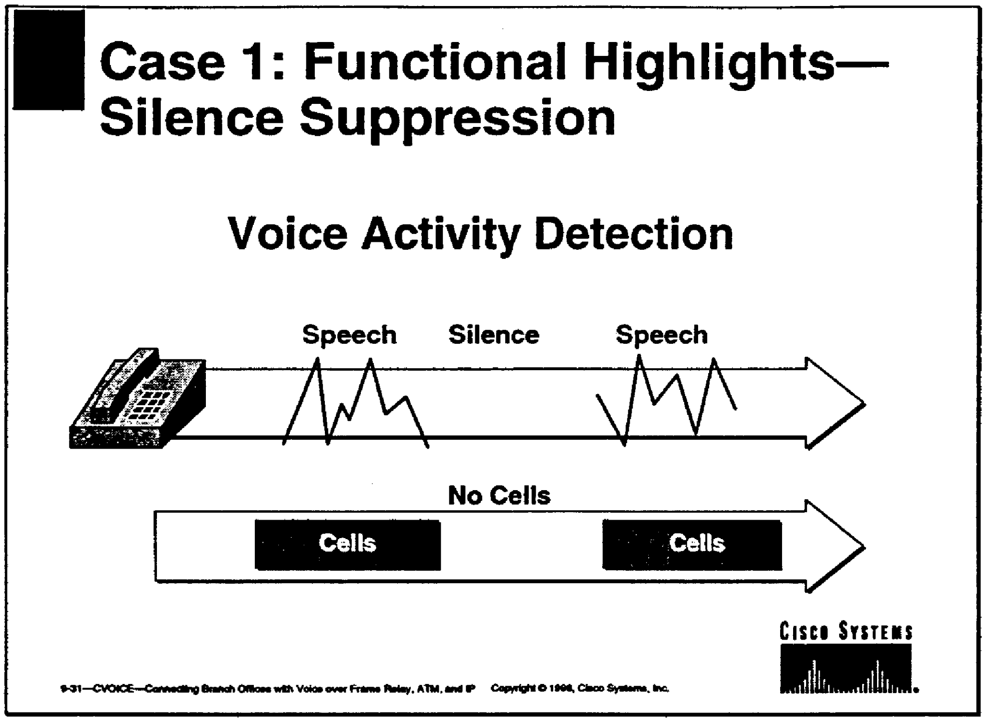
Case 1: Network Redesign

By redesigning the network, the different types of traffic can be consolidated across a single redundant infrastructure based on ATM cell switching.

In the process, the firm migrated from TDM networks to take advantage of the efficiencies an ATM network affords. Both voice compression and closed-loop congestion traffic management are implemented to maximize bandwidth utilization. The firm also employed techniques such as VAD to further increase utilization.

Given the efficiency gained by deploying this network, voice traffic essentially rides for free.

Numerous card/port types and speeds are supported across the network. The company in Case 1 now has the flexibility to deploy new applications to users quickly.



Case 1: Functional Highlights

Voice conversations are half duplex by nature. For telephone conversations, that translates to 60 percent of a 64-kbps voice channel containing silence, because we listen and pause between sentences..

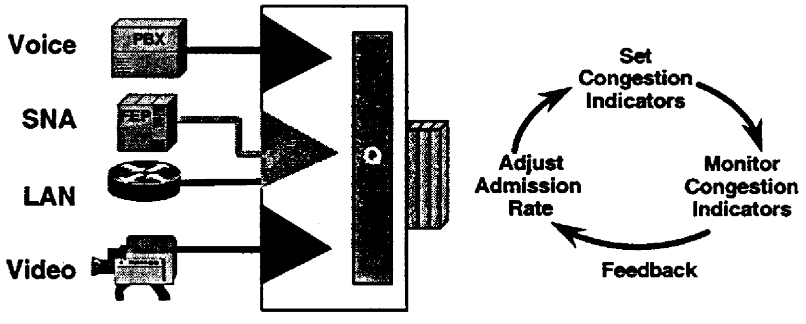
VAD allows traffic from other voice channels or data circuits to make use of the silence on expensive WAN bandwidth and provides for additional savings beyond that achieved by voice compression.

The effects of VAD are highest with the addition of more channels, because the statistical probability of silence increases with the number of voice conversations being combined.

The company in Case 1 has implemented VAD to detect the presence of speech. If a cell does not contain speech, it is not transmitted through the network. So, by implementing VAD together with voice compression, the firm realized even more savings.

Case 1: Functional Highlights—Optimal Bandwidth Utilization

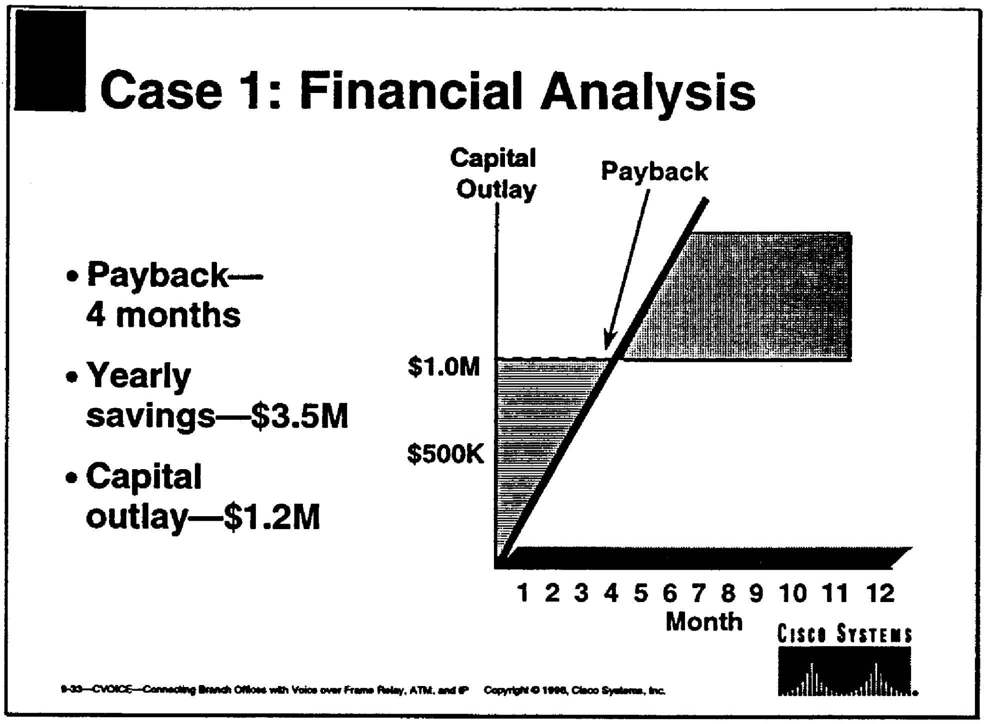
Traffic Management



Rate-based traffic management helps prevent network congestion from occurring and serves to minimize cell discards and retransmissions.

The switch monitors the network for congestion every 40 ms and adjusts the data admission rate to a speed the network can support.

Rate-based traffic management ensures optimal bandwidth utilization.



Case 1: Financial Analysis

By redesigning the network to carry voice, data, and video, and implementing special functions in the network, the company reduced its costs by more than 30 percent while preparing itself for growth.

The graphic summarizes the details of the financial analysis. The capital cost for redesigning the existing network to an ATM network is about $1,205,000 including equipment installation.

The old network cost $920,840 per month to operate. The new combined voice and data ATM network costs $620,280 per month (see the following table), which represents a savings of $300,560 per month.

Bottom line: By combining its voice network with its data network, the company reduced its original yearly network costs from $11 million to about $7.5 million.

The graphic shows us that the payback period is four months.

The following table is the now redesigned ATM network's monthly bandwidth cost.

Case 1: Redesigned ATM Network Expenses

Source	Destination	Quantity	Speed	Monthly Expense
Paris	Lyon	1	FT768K	$11,835
Frankfurt	Munich	1	F-1920	$12,776
Milan	Rome	1	FT384K	$19,124
Amsterdam	Philadelphia	1	F-1984	$48,581
Amsterdam	Zurich	1	F-1984	$44,952
Amsterdam	Rome	1	FT512K	$35,993
Amsterdam	Manchester	1	FT768K	$22,374
Amsterdam	Budapest	1	F-1024	$23,456
Amsterdam	Oslo	1	F-1984	$31,357
Philadelphia	Toronto	1	T1	$12,277
Cologne	London	1	F-1024	$38,073
Cologne	Paris	1	F-1024	$32,831
Cologne	Milan	1	FT512K	$32,082
Cologne	Copenhagen	1	F-1920	$41,578
Cologne	Zurich	1	F-1920	$41,382
London	Toronto	1	T1	$44,090
London	Dublin	1	FT768K	$16,412
Frankfurt	Lyon	1	FT768K	$27,054
Copenhagen	Oslo	1	T1	$17,280
Manchester	Dublin	1	E1	$22,093
Budapest	Munich	1	F-1920	$44,680
TOTAL		**21**		**$620,280**

Case 1: Summary

- 30 percent reduction in annual expenses
- Voice quality maintained
- Increased network availability
- Simplified management
- Optional tandem replacement

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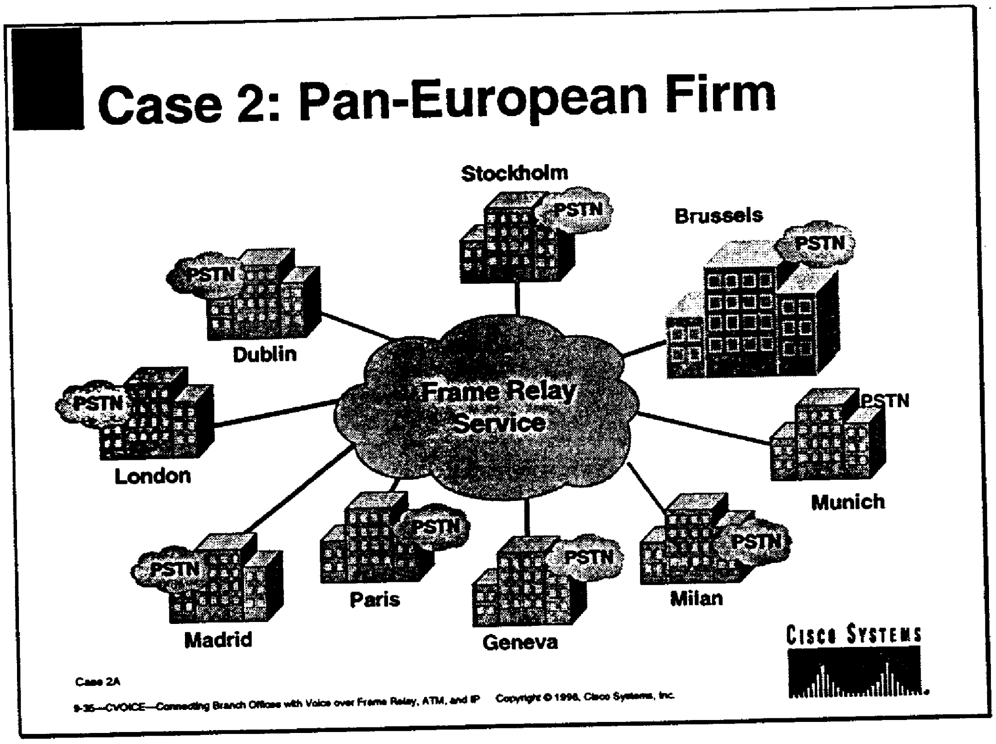
Case 1: Summary

To summarize, Case 1 has shown that after redesigning the network, a 30 percent annual expense reduction was achieved with no reduction in voice quality.

Because of the reduction in equipment and improved robust network design, network availability and manageability has increased.

Case 2: Pan-European Firm



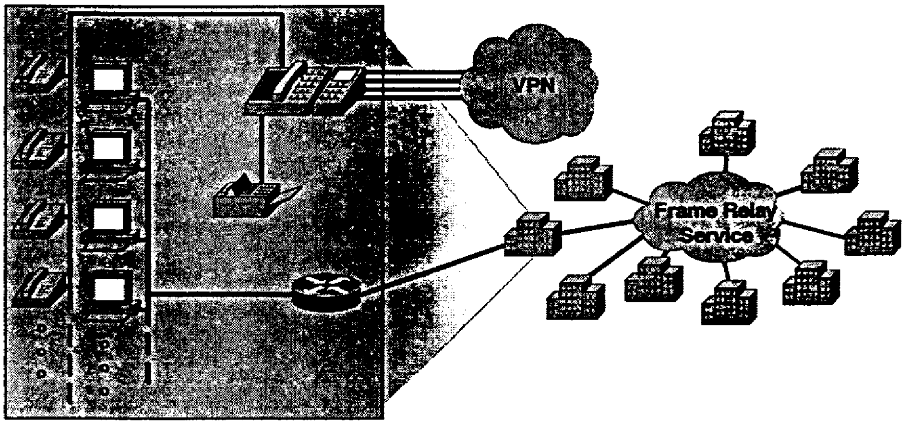
Scenario

A medium-sized firm has locations throughout Europe. Its headquarters is in Brussels and it has offices in eight different countries. Each office is connected to a public Frame Relay network for data and a VPN for voice networking. To be conservative, we have assumed the firm generates enough call volume to obtain a VPN contract from a carrier at about a 15 percent discount off standard PSTN rates.

The majority of calling is between branch employees and outside local customers. Most of its international long distance traffic is to Brussels. This is the most expensive traffic on a per-minute basis.

The company's current Frame Relay network is adequate to handle today's data traffic only.

Case 2: Pan-European Firm
Original Network Design



Regional Offices
40–70 People

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Case 2: Original Network Design

In the regional offices, there are 40 to 70 people in each office. On average, each person uses the phone 1.5 hours per day. The majority of branch calls are between branch employees and local outside customers.

Only 20 percent of the calls is between the offices and headquarters. The interoffice calls amount to about a 15 minutes per day between the headquarters and each branch per month per person. The calls also include fax traffic. Although the branch-to-headquarters traffic volume is lower, it is also the most expensive on a per-minute basis because it is billed at international rates. As a result of these high rates, this firm pays more than \$50,000 per month or about \$600,000 per year for international long distance services.

The firm is using a public Frame Relay service. Voice services are provided by small PBXs connected over the PSTN. To ensure the financial analysis is conservative, it is assumed that the firm's call volume is large enough to obtain a VPN contract from a carrier at about a 15 percent discount from standard PSTN rates.

Objective: Reduce long distance toll charges between the offices and headquarters. The firm is also interested in migrating to an ATM service when it becomes available.

Decision: Any of the packet voice technologies could be used in this case to build the network. But, given its existing Frame Relay network, the company elected to use Voice over Frame Relay technologies. The company opts for an integrated access solution for Voice over Frame Relay functionality to bypass the PSTN for its interoffice calls.

The following table shows the estimated on-net voice and fax traffic volume and PSTN expense for the existing network.

Case 2: PSTN Volume and Expenses

Location	Purpose	Number of People	Average Minutes per Person per Day	On-Net % to HQ	Workdays per Month	Total Minutes per Person per Month	Total Minutes per Office per Month	Cost per Minute*	Monthly Cost per Office
Brussels	HQ								
Stockholm	Branch	45	90	20	21.67	390	17,553	$0.39	$6846
Milan	Branch	55	90	20	21.67	390	21,453	$0.34	$7294
London	Branch	50	90	20	21.67	390	19,503	$0.23	$4486
Madrid	Branch	40	90	20	21.67	390	15,602	$0.39	$6085
Paris	Branch	70	90	20	21.67	390	27,304	$0.26	$7099
Munich	Branch	70	90	20	21.67	390	27,304	$0.26	$7099
Geneva	Branch	65	90	20	21.67	390	25,354	$0.35	$8874
Dublin	Branch	45	90	20	21.67	390	17,553	$0.36	$6319
TOTAL							**171,626**		**$54,101**

* This is the average cost per minute and assumes 50 percent of calls are to headquarters and 50 percent from headquarters. Cost of a voice call based on carrier quote, off-net pricing with discount.

The Data Network

The exact configuration of the data network and the ongoing monthly expenses, or run-rate, are shown in the next table. Note there is a single PVC between each branch and headquarters.

Case 2: Current Data Network Configuration

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CIR	Frame Relay Charges*
Brussels	Headquarters	E1	768	—	$2800
Stockholm	Branch	E1	128	64	$2690
Milan	Branch	E1	128	64	$3010
London	Branch	E1	128	64	$3010
Madrid	Branch	E1	128	64	$3170
Paris	Branch	E1	128	64	$3010
Munich	Branch	E1	128	64	$3010
Geneva	Branch	E1	128	64	$3110
Dublin	Branch	E1	128	64	$3110
TOTAL					**$26,920**

* The port cost and the PVC cost are included in the Frame Relay Charges column.

Network Redesign

The first step is to determine the additional bandwidth required on the data network to support the voice and fax traffic. Determining the required bandwidth can be done in one of two ways:

- **Method 1.** The best way is to collect traffic information from both the key system or PBX and the router, and graphically add the voice and data traffic to see how often the combined voice and data traffic exceeds the current available bandwidth. However, this kind of traffic information is often unavailable or difficult to obtain.
- **Method 2.** If this information is not available, the easiest method to determine the proper upgrade is to estimate whether (and how much) extra bandwidth is required, provision that, and then add the voice traffic to the data stream while tracking two measures of performance: user-reported voice quality, and data latency. If either performance measure appears to be suffering, then more bandwidth should be added.

Often, data and voice traffic will peak at different times during the day. Consequently, the data network will frequently benefit from the added bandwidth.

Assumptions. The network was redesigned considering the following assumptions.

- The total voice and fax call volume per person equals about 90 minutes per day.
- The call volume between headquarters and branch personnel represents about 20 percent of the total call volume or about 15 minutes. Plus, a peak load hour loading factor of 17 percent is appropriate.
- The Cisco voice compression module uses 8 kbps plus 3 kbps overhead (11 kbps per voice channel); thus, each 64-kbps trunk supports five voice channels.
- The branch PBXs require additional trunk modules.

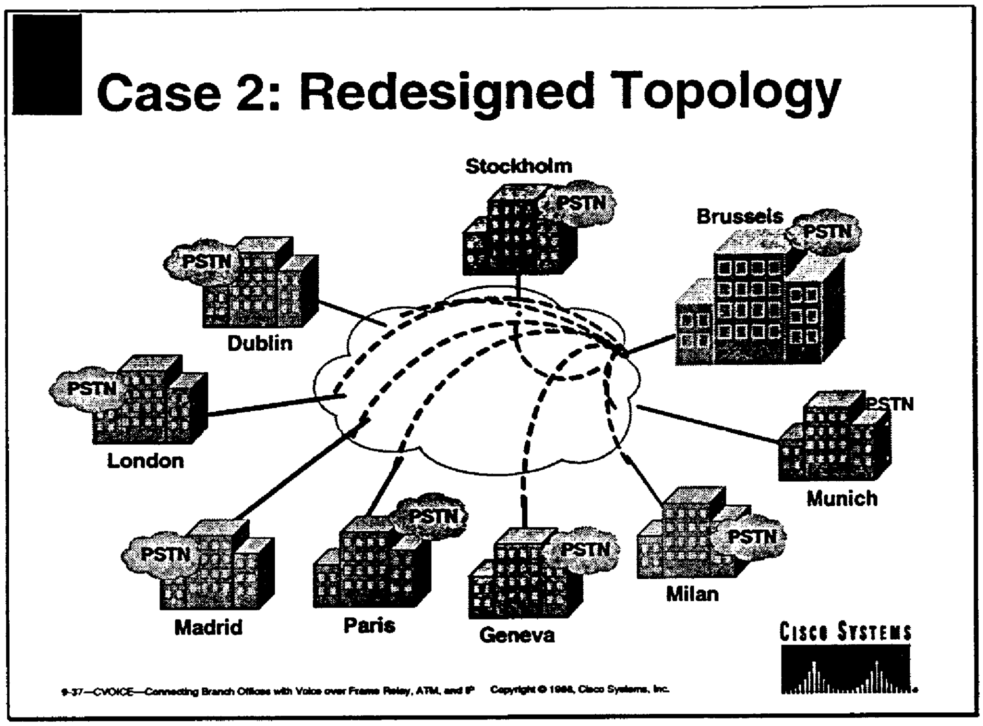
Trunking Analysis

The amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is determined using the assumptions and the appropriate traffic engineering tables, given the desired P grade of service. This firm chose a P.05 grade of service. The following table summarizes the calculations and provides the number of PBX trunks that would be required at each site.

Case 2: Trunking Analysis

Number of Users	Number of Sites	Hours/Day on Phone	Minutes/ Day	Busy Hour (17%)	Minutes/ Busy Hour	% Traffic to HQ	Total Erlangs to HQ	Required Lines for 0.05 Blocking Probability
40	1	1.5	3600	0.17	612	20	2.04	6
45	2	1.5	4050	0.17	689	20	2.30	6
50	1	1.5	4500	0.17	765	20	2.55	6
55	1	1.5	4950	0.17	842	20	2.81	7
60	0	0	—	0	—	0	0	0
65	1	1.5	5850	0.17	995	20	3.32	8
70	2	1.5	6300	0.17	1,071	20	3.57	8
						Total Erlang to HQ	22.44	31

Based on this trunking analysis, the company determined the additional bandwidth required on the Frame Relay network to support the voice and fax traffic. From the assumptions, each 64 kbps of bandwidth can provide a minimum of five voice channels.



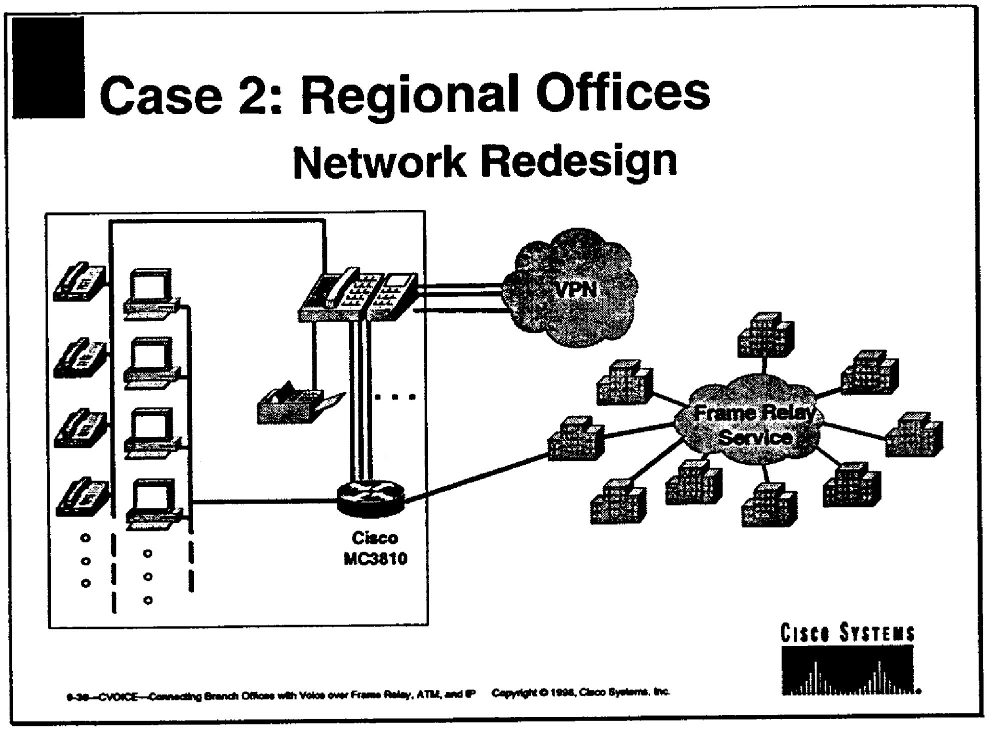
Case 2: Redesigned Topology

The company opted for a multiservice integrated access solution. It chose an MC3810 to transport interoffice voice traffic across the Frame Relay network.

From this picture of the Frame Relay topology you can see that the network was redesigned to use a single PVC configured to carry both data and voice traffic. The network was redesigned to keep expenses down because public Frame Relay carriers typically charge on a per-PVC basis. To successfully accomplish the redesign, the MC3810 has been configured to prioritize voice and data traffic.

All circuit and port speeds were increased, as was the PVC CIR. An additional headquarters access line was installed to support the added traffic.

Note that voice traffic does not pass through the router portion of the MC3810, but instead through the switch. Consequently, in this case, the voice traffic is not encapsulated in IP. Of course, the data traffic does go through the router because it is IP traffic.



Case 2: Regional Offices (Network Redesign)

At the regional offices, the lines going from the key system to the MC3810 are the PBX trunks that are moved from the VPN to the MC3810 to accommodate the voice traffic. The MC3810 is configured for Voice over Frame Relay using bundled DLCIs and fax relay.

The PBX least-cost routing tables are configured to preferentially select MC3810 trunks, for on-net voice and fax.

Meanwhile, the PSTN continues to carry local calls and back up voice traffic on the Frame Relay network. On-net fax traffic is also routed across the Frame Relay network.

As a conservative measure, to accommodate added voice traffic, the Frame Relay network access line speeds were increased, as were the CIRs. These expenses were included in the financial model along with estimates for additional PBX or key system modules that might be required.

An estimated installation charge for PBX, router, etc., also was included to cover that expense.

Network Upgrade

Recall that each branch office was originally linked to the Frame Relay network via a 128-kbps Frame Relay port (using an E1 access line), with the PVC CIR set to 64 kbps. The original link was more than sufficient to handle the original data traffic. However, it will not be enough to carry the added compressed voice traffic. The bandwidth at each location also must be upgraded.

The company decided that because it was redesigning its voice/data network, this would be an ideal time to upgrade its network at all regional and branch offices.

Upgrade Considerations and Strategy

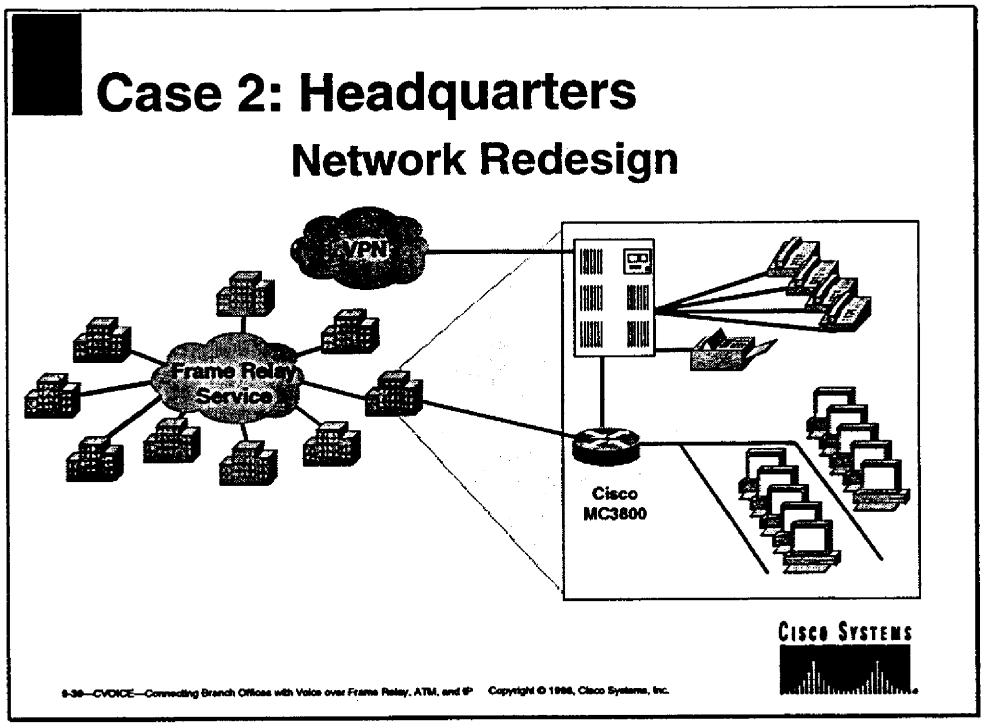
Because the company will make a major change in its voice and data network, and rather than conduct traffic studies to determine the minimum upgrades each location would require, it was felt that upgrading all locations to the same high bandwidth level would provide a conservative approach and leave room for growth in either voice or data traffic. The following facts and assumptions serve as the company's upgrade formula.

Upgrade Formula

The upgrade is based on the largest branch offices in the company. The largest branch offices, containing 70 people, were reengineered to direct a maximum of eight voice lines of traffic over the Frame Relay network. At its maximum, this data stream could reach 88 kbps (because each voice line can load a maximum of 11 kbps). The branch port speeds were doubled from 128 kbps to 256 kbps, far more than the maximum added voice load, to provide improved performance (decrease buffer delay at the port) for the more sensitive voice traffic, and to leave room for growth. The PVC CIR was increased from 64 to 128 kbps to ensure that delay due to vendor network congestion would be minimized (although as always with a Frame Relay network, close watch must be kept on latency measurements to ensure that the PVC CIR is set properly).

This added Frame Relay bandwidth is also certain to improve current data traffic performance. The voice traffic, even at its maximum, still takes up less bandwidth than the new doubled port speed delivers. And voice traffic, being exceptionally variable over time, leaves the new doubled Frame Relay pipe open for more and faster data traffic for most of the day. Although the voice traffic at its peak adds an extra 88-kbps load to the Frame Relay circuit, on average it adds less than 30 kbps of data throughout the day. The rest of the added 128-kbps bandwidth is available for improved data performance. If more savings are required, the smaller branch locations' circuit and port speed could be upgraded in a smaller increment. Half the branches under consideration needed only five or six voice circuits to be added to the data stream; adding 64 kbps to the port size (upgrading from 128 kbps to 192 kbps) would have been more than sufficient.

As long as the business case could support it, however, it was preferred to upgrade the network once and then reduce bandwidth, and expenses, over time as experience is gained with the new traffic patterns.



Case 2: Headquarters (Network Redesign)

At headquarters, tandem switching is configured on the Cisco MC3810 to allow voice traffic from one office to transit through, and terminate at a different regional office's PBX. Switching through the router eliminates the need to tandem though the PBX.

Upgraded Network and Expenses

The company had the MC3810 installed in the branches and the headquarters. The port speeds and the CIR of the PVCs were increased to accommodate the voice traffic. Adding a separate PVC for the voice was unnecessary because the MC3810 prioritizes the traffic on a single PVC, thus avoiding the cost of another PVC.

Each branch office connected the appropriate number of PBX trunks to the Cisco MC3810. The PBX can direct approximately 95 percent of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco MC3810. The remaining overflow traffic, an estimated 5 percent, is directed to the PSTN.

By using G.729 encoding, the MC3810 can compress each 64-kbps voice channel to a data stream of approximately 11 kbps (8 kbps or less plus 3 kbps overhead) and forward the compressed voice traffic over the Frame Relay network.

Note The 11 kbps is a conservative measure because it does not include the benefits of the silence suppression techniques implemented in the MC3810. The MC3810 can support up to 24 channels of 8-kbps compressed voice that can be transported over public or private Frame Relay, ATM, or leased line networks.

Case 2: Functional Highlights

- Voice prioritization within a single PVC (bundled DLCI)
- Tandem switching
- ATM service ready
- Multiflex trunk

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Case 2: Functional Highlights

The graphic shows the highlights of the upgraded network using an MC3810 for Frame Relay. There are additional trunking options not explored here, called multiflex trunking.

Options: A Second PVC

The MC3810 was configured to support tandem switching of the branch to branch voice traffic. Tandem switching relieves the central site PBX of this task and avoids the PBX recompression cycle to maintain high quality voice. This savings was not included in the financial analysis.

For those areas of the world preparing to use ATM services, the Cisco MC3810 can be modified via a software change to support ATM.

A number of different network designs were available to this firm. One option would be to add a second PVC to transport the voice traffic. However, the second PVC option would also result in higher expenses because public Frame Relay providers commonly charge for each PVC.

Another, less expensive method is to utilize the same PVC for both data and voice traffic. The dual use PVC option means sharing a properly engineered and sized PVC between voice and data. But PVC sharing can be done only if the access device supports the prioritization of real-time traffic, voice, and video, over nonreal-time traffic. This design is available when using the MC3810 because it can prioritize the delay-sensitive traffic, voice, while ensuring that the delay-insensitive traffic, data, is also serviced properly. Prioritizing real-time traffic would avoid to some degree the potential contention between the two different traffic types.

In addition, Cisco IOS™technology provides voice switching and advanced call management. For example, the headquarters MC3810 can be configured to switch calls between branches. Although the call volume may be low, the per-minute cost in this case is very high. By off-loading this tandem function from the PBX, the delay budget is reduced and the quality is maintained by avoiding the compression/decompression cycle required by the PBX. This additional savings was not included in this case study.

The final upgrade and the cost to upgrade are shown in the following table.

Case 2: Upgraded Equipment and Costs

Location	Purpose	Access Line Speed	Initial Port Speed	Initial PVC CIR	Frame Relay Charges	Upgraded Port Speed	Upgraded PVC CIR	Frame Relay Charges	Cost to Upgrade
Brussels	HQ	E1	768		$2800	2 x1024		$14,300	$11,500
Stockholm	Br	E1	128	64	$2690	256	128	$4958	$2268
Milan	Br	E1	128	64	$3010	256	128	$6238	$3228
London	Br	E1	128	64	$3010	256	128	$4958	$1948
Madrid	Br	E1	128	64	$3170	256	128	$6238	$3068
Paris	Br	E1	128	64	$3010	256	128	$4958	$1948
Munich	Br	E1	128	64	$3010	256	128	$4958	$1948
Geneva	Br	E1	128	64	$3110	256	128	$6238	$3128
Dublin	Br	E1	128	64	$3110	256	128	$6238	$3128
TOTAL					**$26,920**			**$59,084**	**$32,164**

Futures

The MC3810 can support E1/T1 ATM services via a software change as these services become available. Plus, the CBR and VBR support in the unit prepares this firm for video conferencing when it becomes required.

Case 2: Financial Analysis

The capital costs for the branches and the headquarters are shown in the following table. The required additional bandwidth indicated in the preceding table costs $32,000 per month.

Case 2: Capital Costs

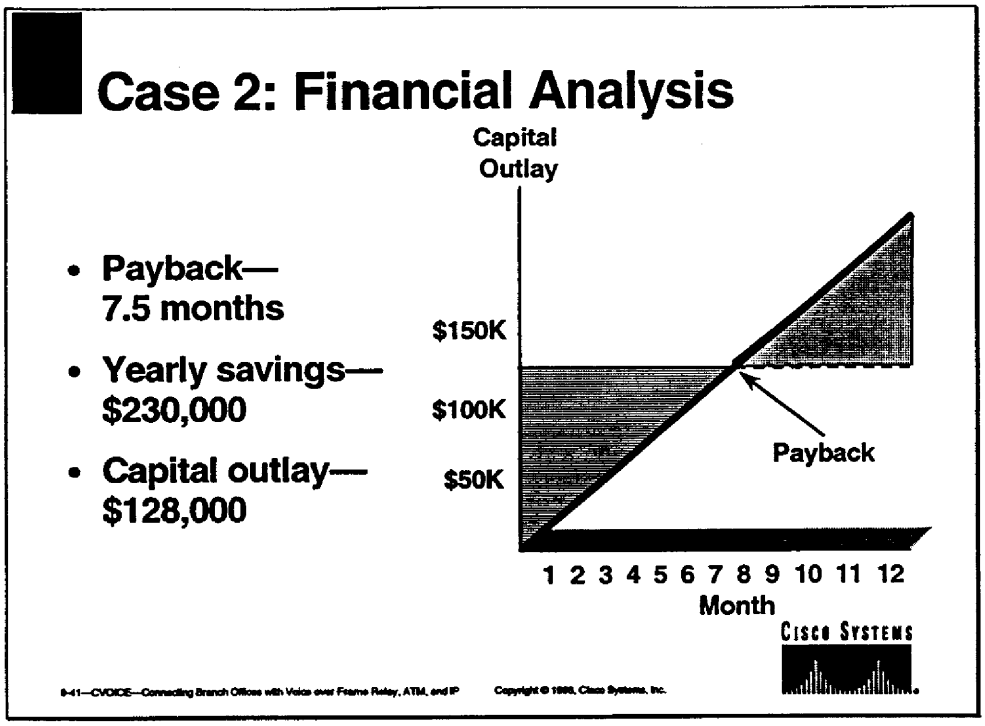
Equipment/Location	Cost
Cisco MC3810s	$72,000
Branch PBX trunk modules	$5400
Number required	8
Total	**$43,200**
Headquarters PBX trunk modules	$12,300
Number required	1
Total	**$12,300**
Total capital cost	**$127,500**

Comparing the savings to these additional expenses, the following table illustrates the net monthly savings. This firm saves more than $230,000 per year by moving on-net voice traffic onto the Frame Relay network.

Case 2: Savings and Payback

Savings/Costs/Payback	Amount
Monthly PSTN voice expenses	$54,101
Multiply by 95 percent (P.05)	0.95
Monthly PSTN voice savings	$51,396
Required added bandwidth	$(32,164)
Net total monthly savings	**$19,232**
Net annual savings	**$230,787**
Total capital costs	$127,500
Installation	$20,000
Total capital costs	**$147,500**
Payback period (months)	**7.7**

The payback period is less than eight months. The following figure graphically shows this payback period and summarizes the important numbers.



This firm reduced its annual networking expenses by over $230,000 by redesigning its network to carry voice. The company paid for the equipment in less than eight months.

The redesign costs include additional Frame Relay service expenses required to transport voice traffic.

Cost per Minute

An organization's voice cost per minute is a common expense measurement. The previous table provides the cost details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 171,626 minutes and the total monthly cost is $54,101. Dividing total cost by total minutes yields an average cost per minute of $0.32 for all calls between headquarters and the branches. To carry 95 percent of this traffic over the data network, that network must be upgraded at a cost of $32,164 per month. Ninety-five percent of 171,626 equals 163,044 minutes. Dividing the upgrade costs by this amount yields an average cost per minute of $0.20. In this case, the average cost per minute was reduced from $0.32 to $0.20, a 38 percent reduction.

Case 2: Summary

- **Annual savings over $230,000**
- **Voice quality maintained using voice priority**
- **Voice decompression/ recompression cycle avoided**

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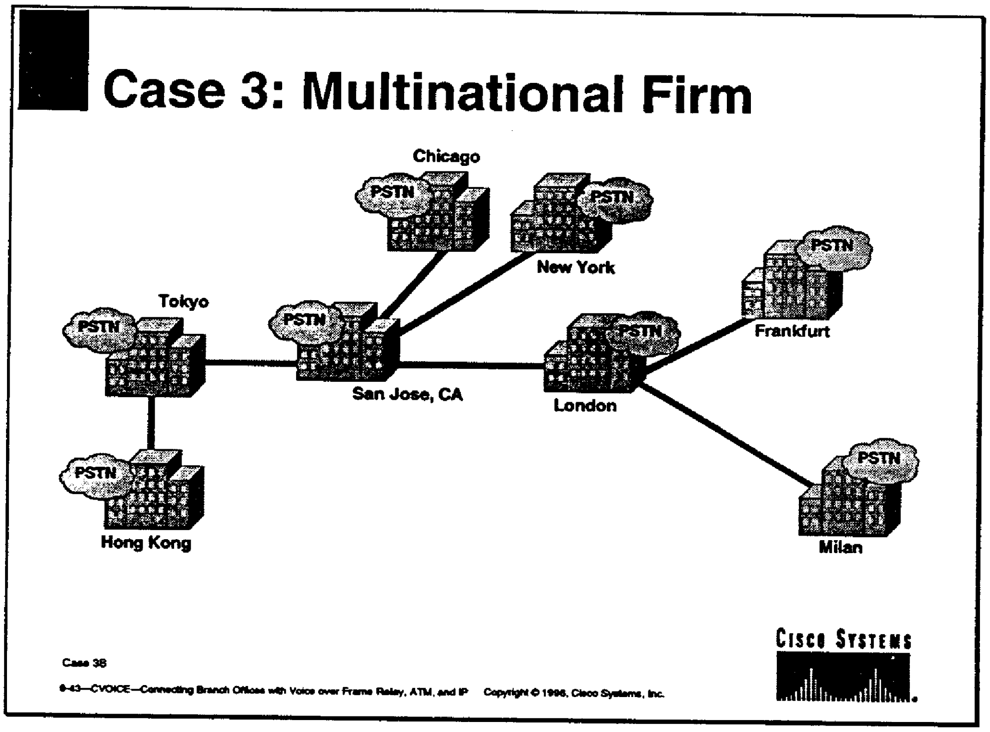
Case 2: Summary

In this case, annual savings of over $230,000 was realized while maintaining voice quality.

The branch to branch calling volume was not figured in. Though the traffic volume is likely to be small, the savings could be quite large, given the distance between the offices.

These savings could easily be realized by utilizing the tandem functions of the MC3810.

Case 3: Multinational Firm

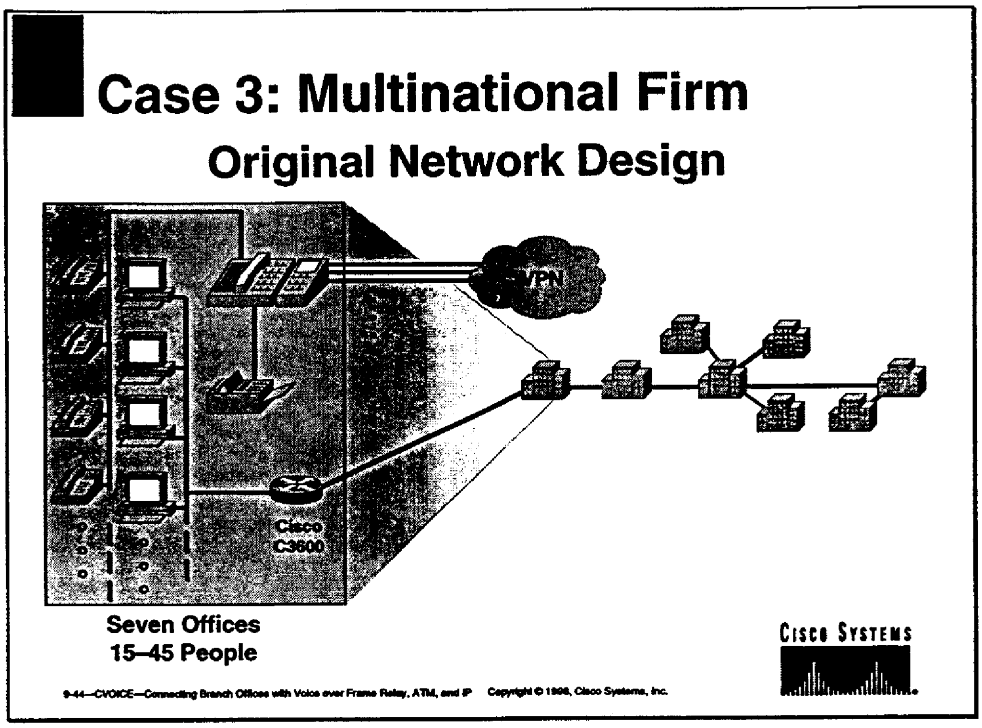


Scenario

This relatively small yet fast-growing organization has seven offices. Besides the San Jose headquarters it has offices in London, Frankfurt, Milan, Hong Kong, Tokyo, Chicago, and New York.

For data communications, it has a routed network using private lines. An eight-node data network, leased by the firm, utilizes routers and is hubbed out of its San Jose headquarters. A number of branches connect through other branches to reach the headquarters location. This arrangement was set up to hold down leased line costs.

Key systems and small PBXs connected over the PSTN provide voice services. Again, it is assumed the firm has a VPN contract at about a 15 percent discount off list prices from standard PSTN rates.



Case 3: Original Network Design

The branches use key systems or small PBXs to serve their phone needs, and a separate router for their leased line IP network. Each branch office has 15 to 45 people.

The majority of calls are between the branch staff members and local customers. Branch employees talk on the phone and use the fax an average of 2.5 hours per day. Only about 20 percent of this traffic is back and forth with headquarters.

Although the branch-to-headquarters traffic volume is small, it is also the most expensive on a per-minute basis because it is billed at international rates. Consequently, for international long distance services, this firm pays approximately $32,000 per month, or about $390,000 annually.

Objective: Reduce long distance toll charges back to headquarters.

The company does not plan to make the transition to Frame Relay but would like to retain that option. It is also considering aggregating its local dial traffic for telecommuters, especially in New York and Hong Kong, where there is a great demand for telecommuting.

Decision: This firm has chosen to implement VoIP to leverage its current infrastructure and provide quality voice connections. The data network may be capable of handling the increased traffic, but to be sure, we have increased the bandwidth of most circuits. The increased bandwidth required on the data network costs much less than the long distance charges.

Existing Voice Network

The following table shows the current on-net voice and fax traffic volume and expense from the branch offices.

Case 3: PSTN Voice and Fax Volume and Expenses

Location	Purpose	No. of People	Avg Min. per Pers. per Day	On-Net % to HQ	Workdays per Month	Tot. Min. per Pers per Mo.	Tot. Min. per Office per Mo.	Cost per Minute*	Monthly Cost per Office
San Jose	HQ								
Frankfurt	Br	15	150	20	21.67	650	9752	$0.54	$5266
Milan	Br	15	150	20	21.67	650	9752	$0.48	$4681
London	Br	45	150	20	21.67	650	29,255	$0.29	$8484
New York	Br	45	150	20	21.67	650	29,255	$0.07	$2048
Chicago	Br	15	150	20	21.67	650	9752	$0.07	$683
Tokyo	Br	15	150	20	21.67	650	9752	$0.52	$5071
Hong Kong	Br	15	150	20	21.67	650	9752	$0.63	$6095
TOTAL							**107,267**		**$32,326**

* This is the average cost per minute and assumes 50 percent of calls are to headquarters and 50 percent from headquarters.

Case 3: Network Redesign

It is essential that redesign of the data network to support the added voice traffic be accomplished without adversely affecting performance. The plan is to have the PSTN cost reductions pay for the redesign. Though any of the packet voice technologies could be used to build this multiservice network, given the firm's infrastructure and expertise in IP, a Voice over IP network is chosen.

Assumptions

In redesigning the network the following assumptions were used:

1. There are approximately 15 people per small branch, 45 per large branch.
2. The bidirectional voice and fax call volume totals about 2.5 hours per person per day per branch.
3. About 20 percent of the total call volume is between headquarters and each branch location.
4. A peak load hour loading factor of 17 percent is appropriate.
5. The Cisco voice compression module uses 8 kbps, plus 1 kbps overhead per call. It was assumed that a 64-kbps trunk circuit supports five calls, rather than seven. This is a conservative estimate.
6. In the small branches, one key system trunk module would be required, whereas two cards would be necessary for the large branches.

Using the assumptions and the following calculations, the amount of voice and fax traffic at each branch office that would optimally be diverted from the PSTN to the multiservice network is as follows:

- 2.5 hours call volume per user per day X 15 users = 37.5 hours daily call volume per office
- 37.5 hours X 60 minutes per hour = 2250 minutes per day
- 2250 minutes X 17 percent (peak load hour) = 382.5 minutes per peak load hour
- 382.5 minutes per peak load hour X 1 Erlang/60 minutes per peak load hour = 6.375 Erlangs
- 6.375 Erlangs X 20 percent of traffic to headquarters = 1.275 Erlangs volume proposed

Considering the assumptions, the following redesign is conservative. The additional bandwidth required on the data network to support the voice and fax traffic is determined. As indicated in Case 2, there are two approaches to determining bandwidth requirements. The best way is to collect traffic information from both the key system or PBX and the router, and then graphically add the voice and data traffic. Collecting traffic information enables one to see how often the combined voice and data traffic would exceed the available bandwidth. But this kind of traffic information is often unavailable. If this information is not available, the easiest method to establish the proper upgrade would be to estimate whether (and how much) extra bandwidth would be required, and provision that amount. Then the voice traffic can be added to the data stream while tracking two measures of performance: user-reported voice quality and data latency. If either performance measure appears to be insufficient, then bandwidth should be added. Data and voice traffic frequently peak at different times in the day. The data network will frequently benefit from the added bandwidth.

Trunking Analysis

To determine the appropriate number of trunks required to carry the traffic, traffic engineering tables are consulted next, given the desired P grade of service. This firm chose a P.05 grade of service. Shown in the following table are the applicable sections of the Erlang C tables. Using the calculated Erlangs and the following table, it turns out that four trunks are required in the smaller offices and eight trunks in the larger ones. In the following table the calculations and trunking requirements are summarized, along with the figures for the larger branch office.

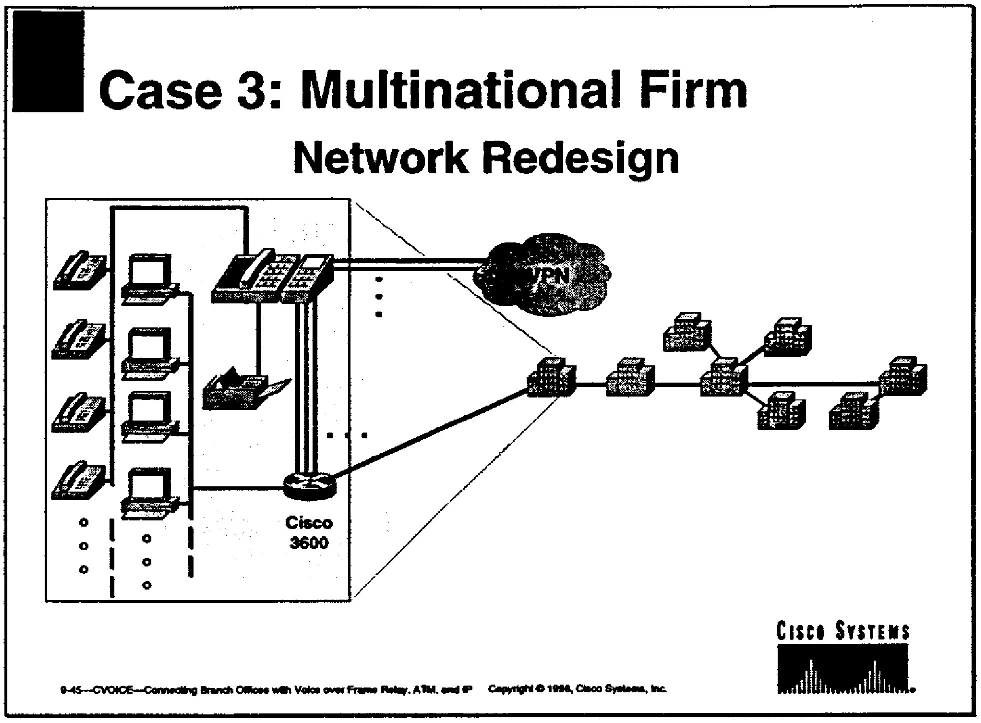
Case 3: Erlang C Table

Blocking Probability (Grade of Service)	Small Branch Traffic to HQ (1.275 Erlangs)	Large Branch Traffic to HQ (3.825 Erlangs)
P = 0.01	5 trunks	10 trunks
P = 0.05	4 trunks	8 trunks
P = 0.10	3 trunks	7 trunks
P = 0.20	3 trunks	6 trunks

Case 3: Trunking Analysis

Site	No. of Users	Call Volume per Day	Minutes per Day	Busy Hour (17%)	Minutes per Busy Hour	% of Traffic to HQ	Total Erlangs to HQ	Required trunks for 0.05 Blocking Probability
Large	45	2.5	6750	0.17	1147.5	20	3.825	8
Small	15	2.5	2250	0.17	382.5	20	1.275	4
						2 Large	7.65	
						5 Small	6.375	
					TOTAL:	**Erlangs for HQ**	**14.025**	**23**

Using the calculated Erlangs in the table above, it turns out that four trunks are required in the smaller offices and eight trunks in the larger ones.

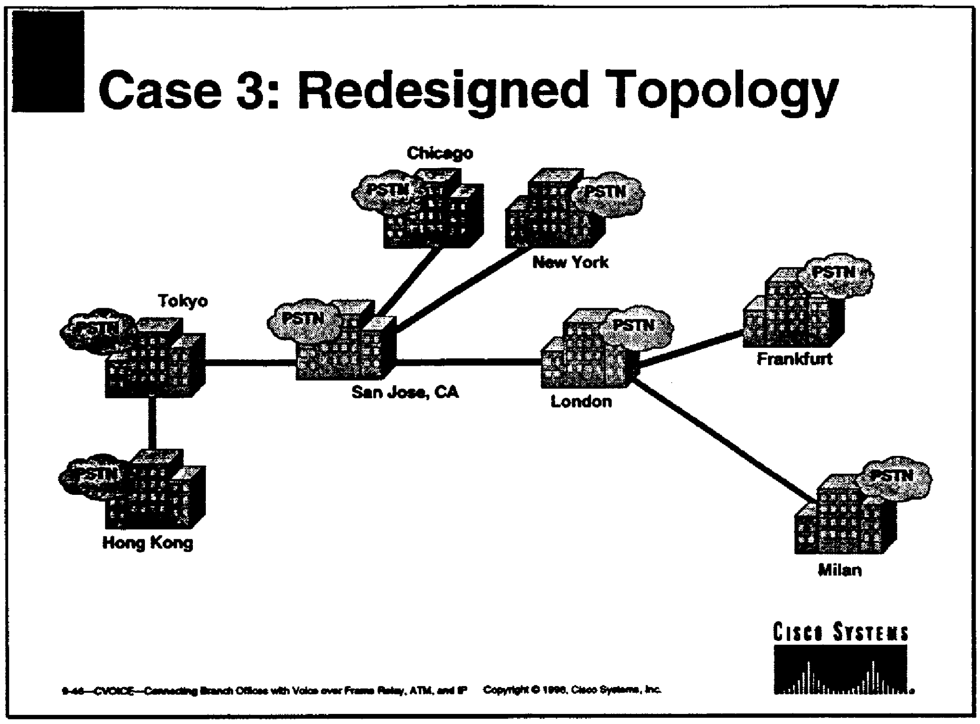


At the smaller branches, a Cisco 3620 is installed and connected to the key system. The central office trunks to the VPN remain for local and off-net voice traffic, but the quantity could be reduced. This reduction is not included in the financial model and represents additional savings.

If required, the necessary key system interfaces are expanded. Just in case, we have added a key system interface module to each office and included it in the financials.

The key systems have been reprogrammed to include the router in the least cost routing tables, thereby routing the calls to the voice-enabled Cisco 3600, when the destination is an interoffice number. Because this firm is leveraging its existing infrastructure, it can actually call any other office as well, but this traffic is not accounted for in this model.

At the two larger offices a Cisco 3640 is installed and connected to the small PBX.



Case 3: Redesigned Topology

Bandwidth was increased to handle the additional voice traffic on the private line network.

To support four simultaneous voice calls, only 36 kbps of additional bandwidth is needed. But to be conservative, bandwidth on most circuits was increased from 64 kbps to 128 kbps. The San Jose to London circuit was increased further to accommodate London's own voice traffic as well as that transiting though to San Jose.

The VoIP section of the design steps noted that certain mechanisms are available to ensure voice quality. These mechanisms, like RSVP and weighted fair queuing, have been configured in the Cisco routers.

Because this is a routed network, any office can now call any other office and the traffic will be routed over the IP network. However, the delay budget should first be calculated to ensure the quality is within the firm's requirements. For those calls that do not meet the requirement, the key system or PBX would be configured to use only the VPN.

Note that on-net fax traffic is also routed across the IP network.

Customer Premises Equipment

A Cisco 3620 router was installed in the smaller branches and a Cisco 3640 router was installed in the two larger branches. Each smaller branch office connected four key system FXO trunks to the Cisco 3620 router (eight trunks to the Cisco 3640 in the larger branch locations). The reprogrammed key system would then direct approximately 95 percent of traffic destined for headquarters preferentially to one of the trunks connected to the Cisco router. The remaining overflow on-net voice traffic, an estimated 5 percent, is directed to the PSTN.

The leased lines terminate at the San Jose headquarters, where the voice channels are decompressed, and routed to the headquarters PBX. Because 23 channels have been removed from the PSTN and are now sent over the router network, the San Jose headquarters can remove one of the PSTN T1 access lines.

The Cisco 3600 enables high performance and low latency, which result in very high voice quality. This firm is planning a possible migration to Frame Relay and support for telecommuting. In Hong Kong and New York, where commuting to work can be very difficult, the Cisco 3600 will eventually be used as an access server for employees telecommuting. If the firm decides to make the transition to Frame Relay, the Cisco 3600 can support that protocol.

It should be noted that not all key systems provide automatic route selection for preferential voice traffic switching. If you are considering this type of configuration, contact the key system or PBX vendor to ensure this capability exists in the equipment.

Case 3: Network Redesign Incremental Expense

The additional bandwidth is not free. The following table furnishes the incremental expense details.

Case 3: Incremental Expense

San Jose to:	Original 64 kbps (Monthly Cost)	Redesigned 128 kbps (Monthly cost)	Redesigned 192 kbps (Monthly cost)	Incremental Cost (Monthly cost)
Tokyo	$8700	$13,400		$4700
Tokyo to Hong Kong	$5500			—
Chicago	$1250	$2050		$800
New York	$1400		$3100	$1700
London	$6400		$13,400	$7000
London to Milan	$5100	$8150		$3050
London to Frankfurt	$4250			—
TOTAL				**$17,250**

Case 3: Functional Highlights

- Extend the routed network architecture
- Only 9 kbps per call
- Migration path to Frame Relay
- Future interoperability with H.323 applications

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Case 3: Functional Highlights

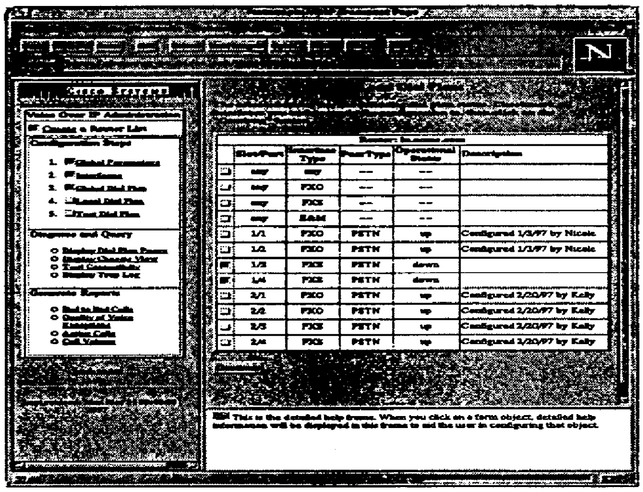
This design has extended the firm's IP network, thereby leveraging the expertise it has already built within the company.

The compression algorithm used here was CSA-CELP, which, including overhead, utilizes only 9 kbps per call.

The Cisco 3600 itself is modular and provides a migration path from private lines to Frame Relay, one of the firm's potential requirements.

The VoIP is standards based to allow future integration with H.323 applications such as video delivery and Web-based applications such as Microsoft's NetMeeting.

Case 3: Functional Highlights—Dial Plan



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The Java-based manager shown in the graphic is a complete tool offering configuration of the voice network, Call Detail Reports, and QoS. This tool allows simple dial plan configuration and easy deployment of the dial plan throughout the network.

Case 3: Financial Analysis

The following table shows the capital costs for the branches and the headquarters in Case 3. The required additional bandwidth indicated in the preceding section costs $22,050 per month.

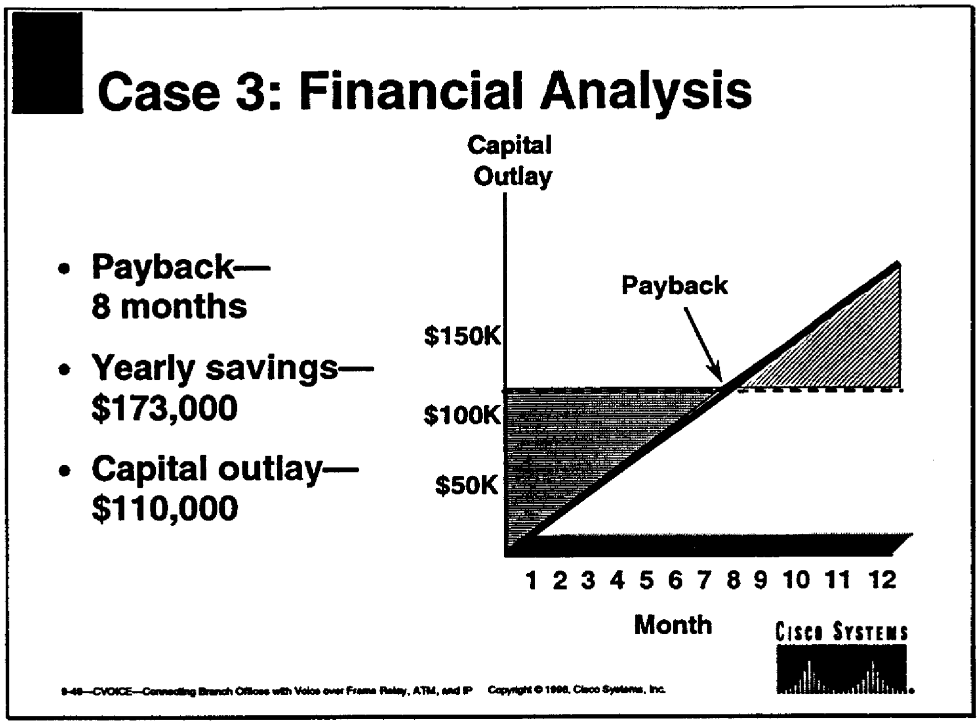
Case 3: Capital Costs

Equipment/Location	Cost
Cisco 3620 and 3640 routers	$84,000
Branch key system modules	$700
Number required (1/small branch 2/large branch)	9
Subotal	**$6300**
HQ PBX trunk modules	$5418
Number required	1
Subtotal	**$5418**
Total capital cost	**$95,718**

Comparing the savings to these additional expenses, the net monthly savings are illustrated in the following table. This firm saves nearly $175,000 per year by moving its internal voice traffic onto its router backbone. The payback period is less than eight months.

Case 3: Savings and Payback

Expenses/Savings/Payback	Amount
Monthly PSTN voice expenses	$32,326
Multiply by 95 percent (P0.05)	0.95
Monthly PSTN voice savings	$30,710
E1 removed from HQ to PSTN	$950
Required added WAN links	–$17,250
Net total monthly savings	14,410
Net total annual savings	$172,919
Capital costs	$95,718
Installation (estimate)	$15,000
Total capital costs	**$110,718**
Payback period (months)	**7.7**



The network redesign saves over $173,000. The payback period of eight months is conservative given the installation of additional private line bandwidth.

After new traffic patterns are established, it may be possible to reduce bandwidth.

Cost per Minute

The previous table provides the details required to calculate the original cost per minute. The total on-net minutes per month for all offices equals 107,267 minutes and the total monthly cost is $32,326. Dividing $32,326 by 107,267 yields an average cost per minute of $0.30 for all calls between headquarters and the branches. To carry 95 percent of this traffic over the data network, that network must be upgraded at a cost of $17,250 per month. Ninety-five percent of 107,267 equals 101,904 minutes. Dividing the monthly upgrade costs by the carried minutes yields an average cost per minute of $0.17. In this case, the firm was able to reduce its average cost per minute from $0.30 to $0.17, a 43 percent reduction.

Case 3: Summary

- **Annual savings over $173,000**
- **Management simplified using Java-based manager**
- **Voice decompression/ recompression cycle avoided**

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Case 3: Summary

In our final case, the company was able to save over $173,000 by adjusting bandwidth and extending its IP network.

The Java-based manager simplified configuration and deployment.

The Cisco 3600 family of routers anchors its network and allows for growth.

Note The networks presented in the case studies were overengineered and have more bandwidth than required. Overengineering the network was done to show that the added voice traffic would not negatively affect data traffic. In fact, given that voice and data traffic often peak at different times of the day, the data network will likely perform better.

The solution you choose will be based on current and future needs as well as the existing environment and expertise.

Chapter 9 provided a background on those aspects of voice and data networking that must be considered when designing a multiservice integrated voice data network. It covered the available technologies—VoIP, VoFR, and VoATM—while trying to provide a framework to help analyze the optimum solution.

The case studies showed examples, using actual scenarios and costs, of how the technologies can be implemented and the benefits obtained.

For future reference: analyze traffic patterns, account for quality requirements, calculate delay budget, and decide upon the right design for your organization's individual needs.

Summary

Summary

- Design, configure, integrate, and optimize voice in remote branch and regional offices by using technology that combines voice and data transmission over Frame Relay, ATM, and IP connections
- Appraise existing branch and regional office voice traffic, delay budget, and services, and choose the optimum transmission method for voice traffic: Frame Relay, ATM, or IP

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Summary (cont.)

- Analyze existing voice hardware/software, and select which Cisco multiservice access device (2600, 3600 or MC3810) would best serve the needs
- Recommend ways to connect branch and regional offices with Voice over FR/ATM/IP
- Cover case studies of networks using VoFR, VoATM, and VoIP

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A

Supporting Materials and Labs

Voice-Port Commands

Enter global configuration mode to configure voice-port commands. The following table gives a list of voice port commands, whether they can be used for analog or digital voice calls, and which type of voice call signaling is supported.

Table A-1 Cisco IOS™ Voice-Port Commands

Voice-Port Commands	Type of Voice Port	FXO	FXS	E&M
auto-cut-through *	Analog and digital	—	—	Y
codec	Analog and digital	Y	Y	Y
comfort-noise	Analog and digital	Y	Y	Y
compand-type	Analog and digital	Y	Y	Y
connection	Analog and digital	Y	Y	Y
cptone *	Analog	Y	Y	Y
description	Analog and digital	Y	Y	Y
dial-type	Analog and digital	Y	—	Y
echo-cancel coverage	Analog and digital	Y	Y	Y
echo-cancel enable *	Analog and digital	Y	Y	Y
impedance	Analog	Y	—	—
input gain *	Analog and digital	Y	Y	Y
non-linear	Analog and digital	Y	Y	Y
operation	Analog	—	—	Y
output attenuation *	Analog and digital	Y	Y	Y
playout-delay	Analog and digital	Y	Y	Y
ring cadence	Analog	—	Y	—
ring frequency	Analog	—	Y	—
ring number	Analog and digital	Y	—	—
ring time out *		—	Y	—
shutdown	Analog and digital	Y	Y	Y
signal	Analog	Y	Y	Y
snmp trap link-status	Analog and digital	Y	Y	Y
supervisory disconnect *	Analog and digital	Y	—	—
timeouts initial	Analog and digital	Y	Y	Y
timeouts interdigit	Analog and digital	Y	Y	Y
timing clear-wait	Analog and digital	—	—	Y
timing delay-start	Analog and digital	—	—	Y
timing delay-with-integrity	Analog and digital	—		Y
timing digit	Analog and digital	Y	Y	Y
timing inter-digit	Analog and digital	Y	Y	Y
timing pulse	Analog and digital	Y	—	Y
timing pulse inter-digit	Analog and digital	Y	—	Y
timing wink-duration	Analog and digital	—	—	Y
timing wink-wait	Analog and digital	—	—	Y
type	Analog	—	—	Y
vad *	Analog and digital	Y	Y	Y

* Important commands. See *Command Reference Guide.*

Voice Port Planning Forms

To help you keep track of the voice ports while configuring them, you can use the analog and digital voice port planning forms in the *Cisco MC3810 Multiservice Concentrator Software Configuration Guide* appendix titled "Planning Forms." Using the displays from the **show voice port** or **show voice port summary** commands, write down the voice module or signaling type next to the slot/port number to help you configure each port. Documenting this information will help you in configuring the voice ports because you must enter the correct slot/port number to enter voice-port configuration mode. Also, you can use the planning forms to plan the connection type and destination phone number for each voice port.

Relevant Frame Relay Definitions

Relevant Frame relay terms are as follows:

- Virtual circuits (VCs)—Virtual circuits are logical paths through the Frame Relay network. They can be permanently assigned (PVC) or switched (SVC). A single, physical interface can support multiple PVCs, and a single PVC can support multiple traffic types.

  Virtual circuits are identified by means of a data-link connection identifier (DLCI), which may have local significance only.

- Permanent virtual circuit (PVC)—PVCs are virtual circuits that can be fixed and are typically defined during network setup, with each pair of locations to be connected being provided a PVC, and each PVC in turn being assigned a specific transmission rate. Each PVC is identified at each end by means of a DLCI.
- Switched virtual circuit (SVC)—SVCs are virtual circuits that can be set up or torn down on a demand basis. SVCs are not widespread at this time, but that trend is expected to change. SVCs can be useful for locations with low-volume, intermittent traffic.
- Data-link connection identifier (DLCI)—DLCI is the addressing mechanism of Frame Relay. Several restrictions are placed on the assignment of DLCIs. The general rule is to avoid using those DLCIs reserved for other express purposes such as LMI communications, call control signaling, and future use. The total number of DLCIs available for use are 992. Frame Relay PVCs may be assigned values ranging from 16 through 1007.
- Local Management Interface (LMI)—LMI is an optional extension that allows "keepalive" messages, configuration information, and congestion status to be exchanged across the User-Network Interface (UNI) between the access and network devices.
- Latency—Is the time interval between a request for data and the actual moment a response is sent. Layer 2 (OSI) switching improves latency and network throughput. The 8 kbps G.729 and G.729a data algorithms reduce voice latency sensitivity by about 60 percent across the WAN. The basic

compression causes about 40 ms of delay, but does not cause noticeable degradation of quality end-to-end.

- Voice activity detection (VAD)/silence suppression —VAD is integral with voice CSA-CELP standards. This type of silence suppression typically frees up 30 percent of the bandwidth required for a call. This bandwidth may then be allocated, in real time, to other applications.

  If the MC3810 detects silence, 4 bytes are sent with 11 bytes of header indicating changes in the levels of background noise.

Note If VAD is not configured, silence will be sent and bandwidth will be wasted.

Troubleshooting MC3810 Hardware and Cables

The following table is for troubleshooting MC3810 initial startup problems.

Table A-2. Troubleshooting MC3810 Hardware and Cables

Symptom	Possible Cause	Corrective Action
Power LED and fan are off	Power source switched off Faulty power cable Faulty power source Faulty internal power supply	Switch power source on Check/replace power cable Check/correct input power Contact Cisco Technical Assistance Center or your Cisco reseller
Power LED on; fan off	Faulty Cisco MC3810	Contact Cisco Technical Assistance Center or your Cisco reseller
Power LED off; fan on	Faulty Cisco MC3810	Contact Cisco Technical Assistance Center or your Cisco reseller
No initialization response from Cisco MC3810	Faulty modem console terminal Faulty cabling to terminal Faulty Cisco MC3810	Check/replace modem/terminal Check/replace cable Contact Cisco Technical Assistance Center or your Cisco reseller
Unit shuts off after operating for some time	Overheating Faulty Cisco MC3810	Check ventilation Contact Cisco Technical Assistance Center or your Cisco reseller
Console screen display freezes	Console fault Software error Faulty Cisco MC3810	Reset/replace console Repeat power-up procedure Contact Cisco Technical Assistance Center or your Cisco reseller

Dial Peer Validation Tips

Verify that the voice connection is working by doing the following:

- Pick up the handset on a telephone connected to the configuration and verify that you can get a dial tone.
- Make a call from the local telephone to a configured dial peer and verify that the call attempt is successful.

You can check the validity of your dial peer and voice port configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the data configured is correct.
- To show the status of the voice ports, use the **show voice port** command.
- To show the call status for all voice ports, use the **show voice call** command.
- To show the current status of all DSP voice channels, use the **show voice dsp** command.

Dial Peer Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with the dial peer configuration, you can try to resolve the problem by performing the following tasks:

- Use the **show dial-peer voice** command on the local and remote routers to verify that the data is configured correctly on both.
- **ping** the ATM voice-network peer to make sure the peer can be reached.
- Make sure that the line-code and framing commands on the T1 controller are configured correctly. The line-code command should be set to Extended Super Frame format, and the framing command should be set to **b8zs**.
- Use the **show interface** command to verify that serial port 2 is up.
- Make sure that encapsulation ATM is set on serial port 2.
- Check the LED for serial port 2 and make sure it is green, indicating the port is active.

Voice Port Validation Tips

You can check the validity of your voice-port configuration by performing the following steps:

Step 1 Pick up the handset of an attached telephony device and check for a dial tone.

Step 2 If you have dial tone, check for DTMF detection. If the dial tone stops when you dial a digit, then the voice port is most likely configured properly.

Step 3 Use the **show voice port [slot/port]** and **show voice port summary** EXEC commands to verify that the voice-port configuration is correct.

Step 4 Use the **show voice dsp** EXEC command to verify the current status of all DSP voice channels.

Step 5 Use the **show voice call summary** EXEC command to verify the call status for all voice ports.

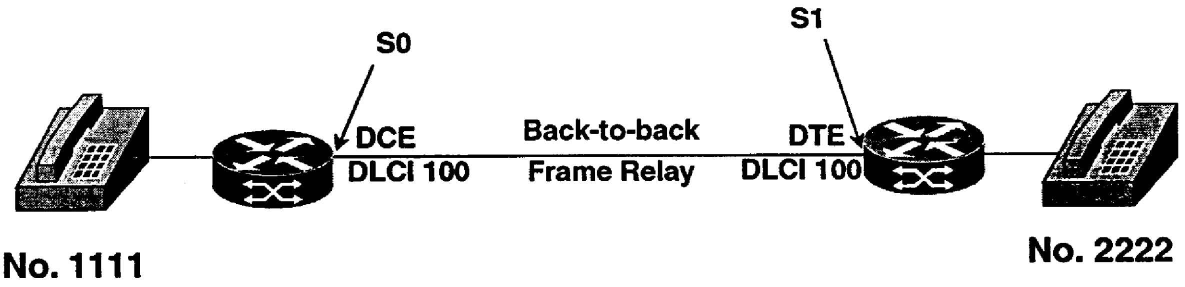
Voice Port Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with voice-port configuration, you can try to resolve the problem by performing the following tasks:

- **ping** the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the "Configuring IP" chapter in the *Cisco IOS Release 11.3 Network Protocols Configuration Guide, Part 1.*
- Use the **show voice port** command to make sure that the voice port is enabled. You can display information for a single voice port, for all voice ports, or a summary report. If the voice port is off line, use the **no shutdown** command.
- Confirm the dial peer configuration (as configured in the "Voice over Frame Relay Configuration," "Voice over ATM Configuration," "Voice-Port Configuration," or "Voice over HDLC Configuration" chapters of the *Cisco MC3810 Multiservice Concentrator Software Configuration Guide.*
- Confirm the Frame Relay, ATM, or HDLC configuration.
- Check that the voice network module has been correctly installed. For more information, refer to the *Cisco MC3810 Multiservice Concentrator Hardware Installation Guide.*

Voice over Frame Relay Lab

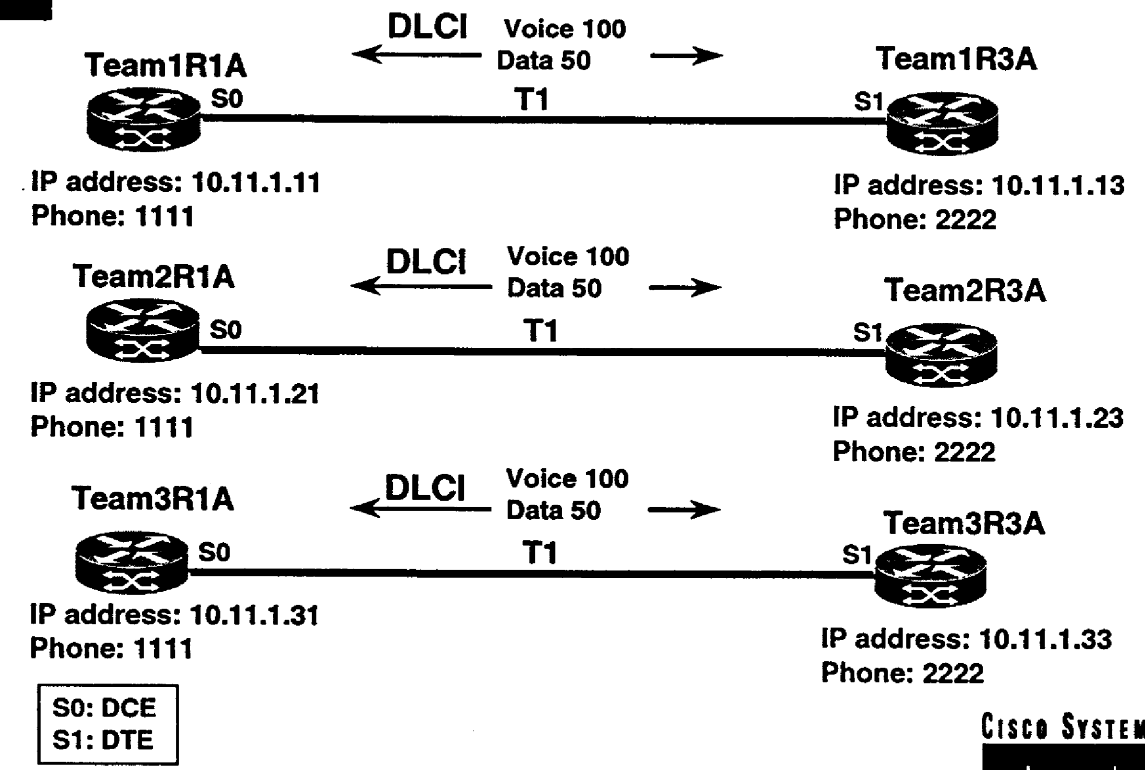
VoFR Lab: Back-to-Back Connection



A-3—CVOICE—Supporting Material and Labs

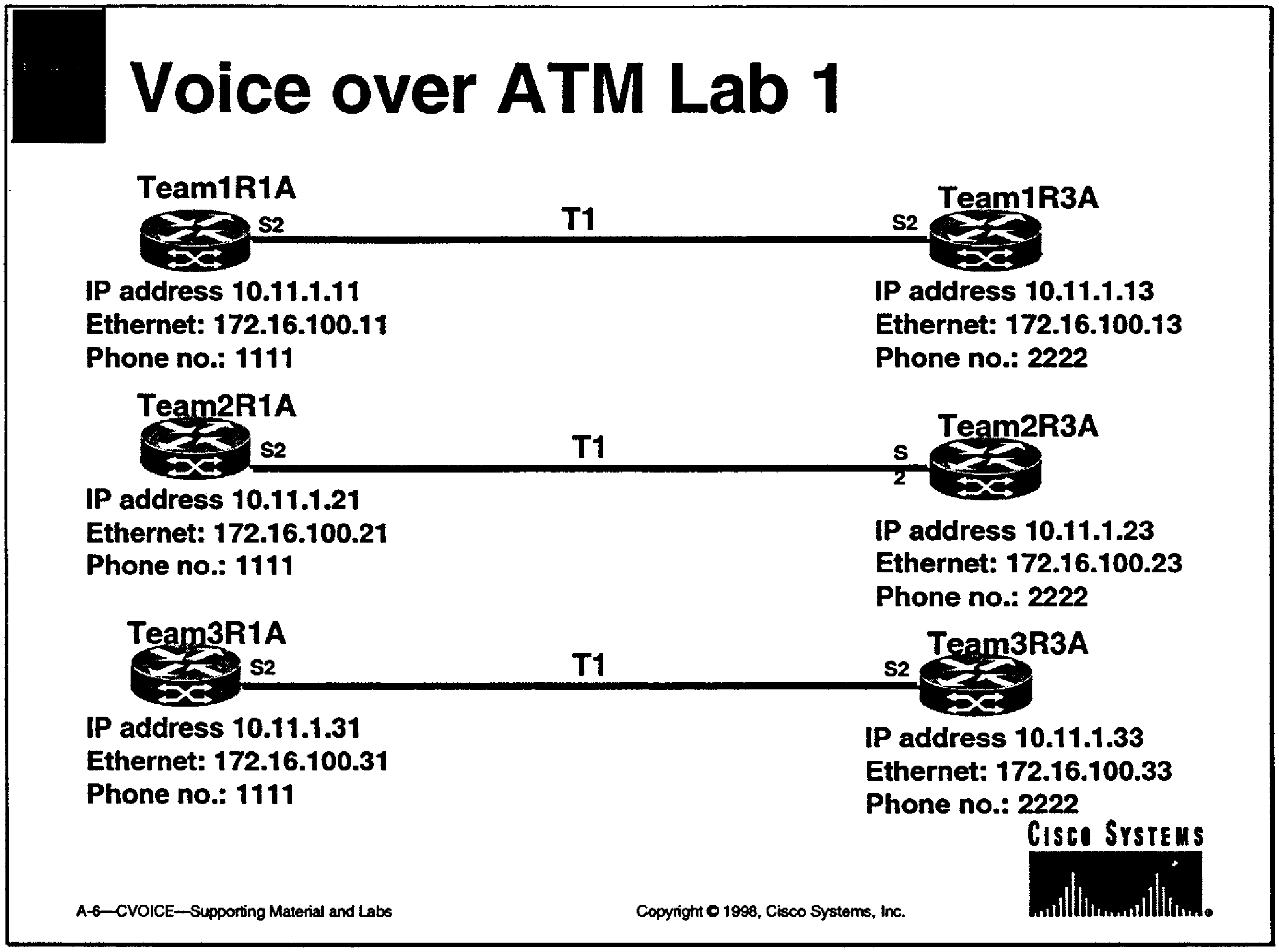
CISCO SYSTEMS

VoFR Frame Relay Lab Setup



A-4—CVOICE—Supporting Material and Labs

Voice over ATM Lab 1



VoATM Lab 1: "Remote" Site's Cisco IOS Configuration

```
! LAB 1 CONFIGURATION ("Remote" Site)
!
version 11.3
no service pad
no service password-encryption
!
hostname team1r1a
!
enable secret 5 $1$gCmM$s/16ooOJFIDkWuDpsyhn90
enable password cisco
!
controller T1 0
 framing esf
 clock source internal
 linecode b8zs
 description line
!
interface Ethernet0
 ip address 172.16.100.11 255.255.255.0
 no ip mroute-cache
!
interface Serial0
 no ip address
 shutdown
!
interface Serial1
 no ip address
 shutdown
!
interface Serial2
 ip address 10.11.1.11 255.255.255.0
 encapsulation atm
 no ip route-cache
 no ip mroute-cache
 map-group atm1
 atm pvc 1 1 100 aal5voice 384 192 48
 atm pvc 2 1 200 aal5snap
 serial restart-delay 0
!
interface FR-ATM20
 no ip address
 no ip mroute-cache
!
router rip
 network 10.0.0.0
!
ip classless
!
map-list atm1
 ip 10.11.1.13 atm-vc 2 broadcast
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
!
voice-port 1/1
!
voice-port 1/2
!
```



```
voice-port 1/5
!
voice-port 1/6
!
dial-peer voice 1 pots
 destination-pattern 1111
 port 1/1
!
dial-peer voice 2 voatm
 destination-pattern 2222
 session target Serial2 1
!
end
```

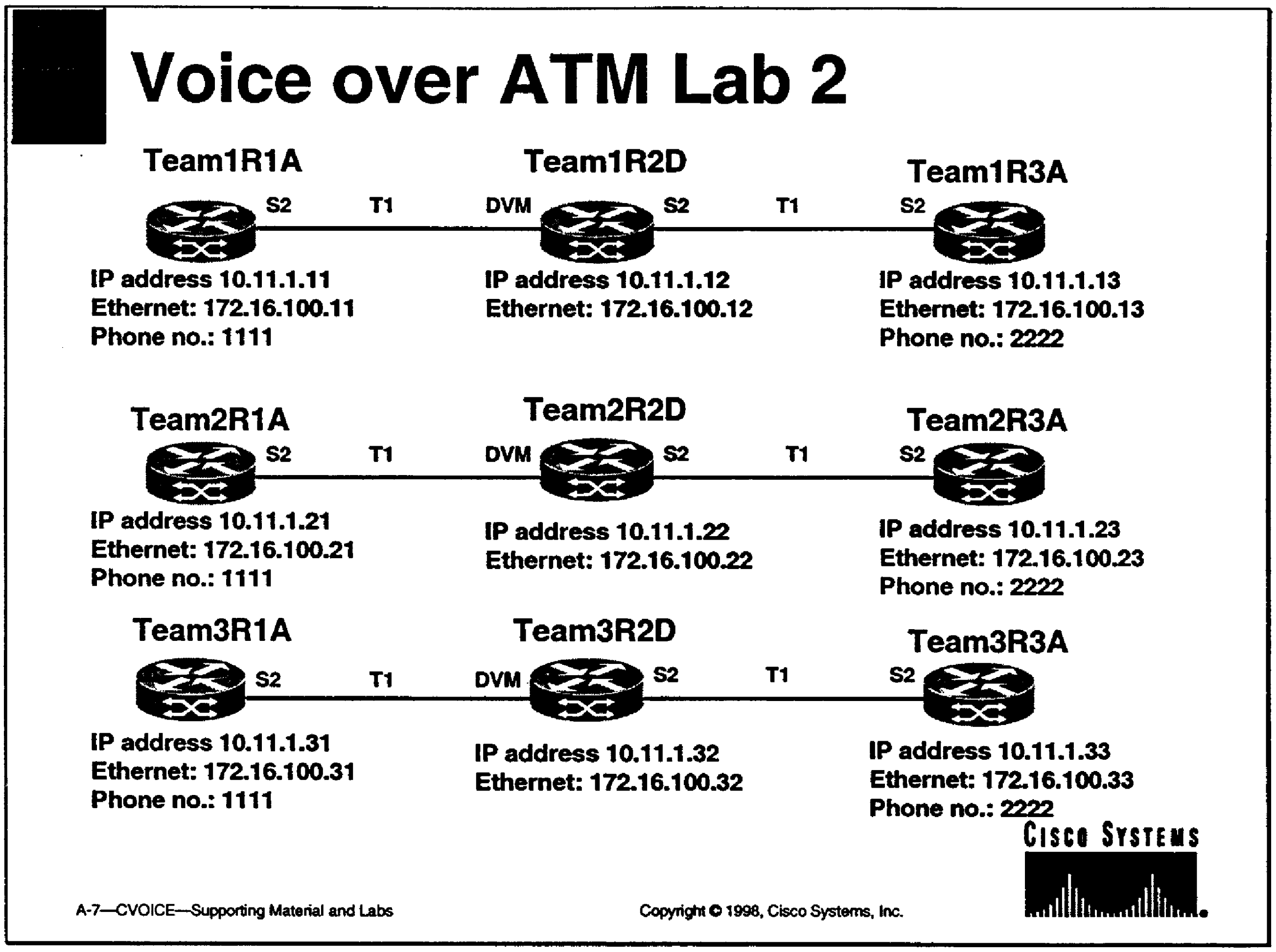
VoATM Lab 1: "Corporate" Station's Cisco IOS Configuration

```
! LAB 1    MC3810 CONFIGURATION ("Corporate" Site)
version 11.3
no service pad
no service password-encryption
!
hostname team1r3a
!
enable secret 5 $1$gCmM$s/16ooOJFIDkWuDpsyhn90
enable password cisco
!
!
!
controller T1 0
 framing esf
 clock source internal
 linecode b8zs
 description line
!
interface Ethernet0
 ip address 172.16.100.13 255.255.255.0
 no ip mroute-cache
!
interface Serial0
 no ip address
 shutdown
!
interface Serial1
 no ip address
 shutdown
!
interface Serial2
 ip address 10.11.1.13 255.255.255.0
 encapsulation atm
 no ip route-cache
 no ip mroute-cache
 map-group atm1
 atm pvc 1 1 100 aal5voice 384 192 48
 atm pvc 2 1 200 aal5snap
 serial restart-delay 0
!
interface FR-ATM20
 no ip address
 no ip mroute-cache
 shutdown
!
ip classless
!
map-list atm1
 ip 10.11.1.13 atm-vc 2 broadcast
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
!
!
voice-port 1/1
!
voice-port 1/2
!
```

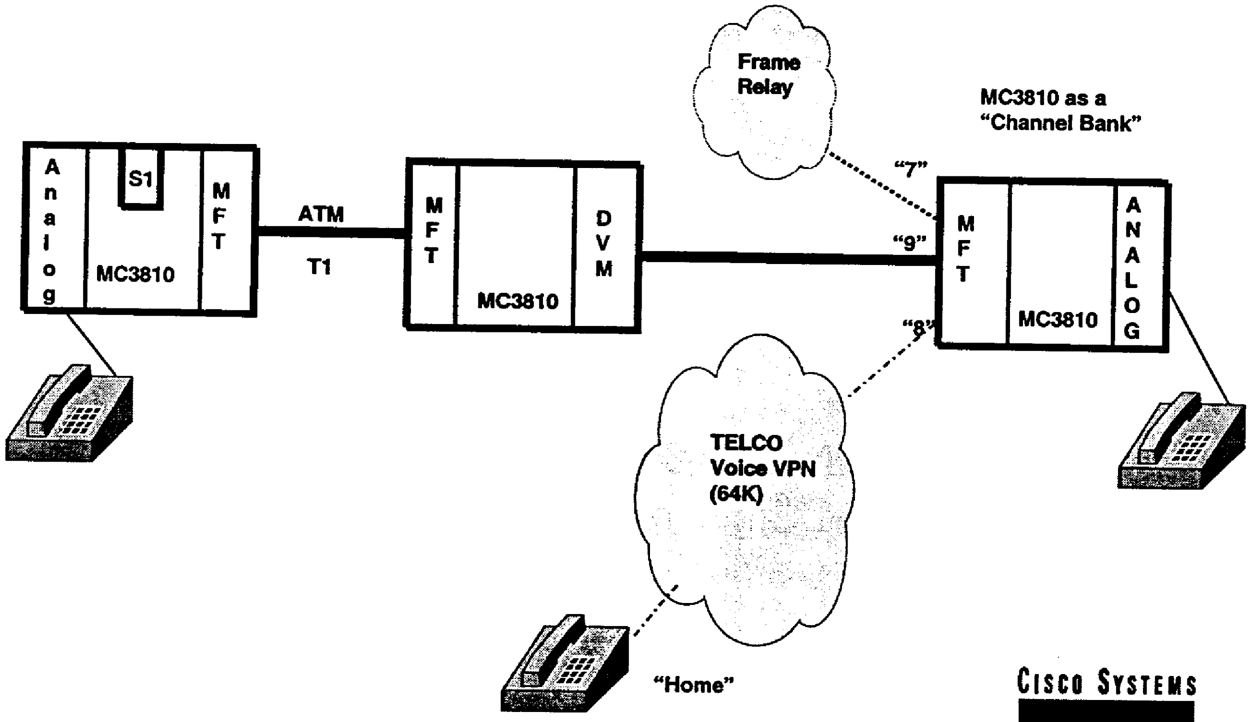


```
voice-port 1/5
!
voice-port 1/6
!
dial-peer voice 1 pots
 destination-pattern 2222
 port 1/1
!
dial-peer voice 2 voatm
 destination-pattern 1111
 session target Serial2 1
!
end
```

Voice over ATM Lab 2



Voice over ATM Lab 2



A-8—CVOICE—Supporting Material and Labs

VoATM Lab 2: "Corporate" Site MC3810 "Channel Bank" Cisco IOS Configuration

```
! LAB 2: MC3810  (CHANNEL BANK LAB - THIS IS THE "Corporate" Site
"CHANNEL BANK")
!
version 11.3
no service pad
no service password-encryption
!
hostname team2r1a
!
enable secret 5 $1$gCmM$s/16ooOJFIDkWuDpsyhn90
enable password cisco
!
!
!
controller T1 0
 mode cas
 voice-group 1 timeslots 1 type e&m-immediate
 framing esf
 linecode b8zs
 description line
!
interface Ethernet0
 ip address 172.16.100.21 255.255.255.0
 no ip mroute-cache
!
interface Serial0
 no ip address
 shutdown
!
interface Serial1
 no ip address
 shutdown
!
interface Serial2
 no ip address
 no ip route-cache
 no ip mroute-cache
 no keepalive
 no fair-queue
!
interface FR-ATM20
 no ip address
 no ip mroute-cache
!
ip classless
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
!
!
voice-port 0/1
!
voice-port 1/1
!
voice-port 1/2
!
voice-port 1/5
!
```

```
voice-port 1/6
!
dial-peer voice 1 pots
 destination-pattern 2222
 port 1/1
!
dial-peer voice 2 pots
 destination-pattern 9
 port 0/1
!
end
```

VOATM Lab 2: DVM MC3810 "Digital Cloud" Cisco IOS Configuration

```
! LAB 2  DVM MC3810 (DIGITAL CLOUD)
!
version 11.3
no service pad
no service password-encryption
!
hostname team2r2d
!
enable secret 5 $1$rZ8g$9Uwp.s6FifLYGv2nX.09k0
enable password cisco
!
!
!
controller T1 0
 framing esf
 clock source internal
 linecode b8zs
!
controller T1 1
 mode cas
 voice-group 1 timeslots 1 type e&m-immediate
 framing esf
 clock source internal
 linecode b8zs
!
interface Ethernet0
 ip address 172.16.100.22 255.255.255.0
!
interface Serial0
 no ip address
 shutdown
!
interface Serial1
 no ip address
 shutdown
!
interface Serial2
 ip address 10.11.1.22 255.255.255.0
 encapsulation atm
 no ip mroute-cache
 map-group atm1
 atm pvc 1 1 100 aal5voice 384 192 48
 atm pvc 2 1 200 aal5snap
 serial restart-delay 0
!
interface FR-ATM20
 no ip address
 no ip mroute-cache
 shutdown
!
router rip
 network 10.0.0.0
!
ip classless
!
map-list atm1
 ip 10.11.1.22 atm-vc 2 broadcast
!
line con 0
line aux 0
line vty 0 4
```

```
 password cisco
 login
!
!
voice-port 1/1
!
dial-peer voice 1 pots
 destination-pattern 9
 port 1/1
!
dial-peer voice 2 voatm
 destination-pattern 1111
 session target Serial2 1
!
end
```

VoATM Lab 2: MC3810 "Remote Site" Cisco IOS Configuration

```
!LAB 2 MC3810 (Remote Site)
version 11.3
no service pad
no service password-encryption
!
hostname team2r1a
!
enable secret 5 $1$gCmM$s/16ooOJFIDkWuDpsyhn90
enable password cisco
!
!
!
controller T1 0
 framing esf
 linecode b8zs
 description line
!
interface Ethernet0
 ip address 172.16.100.23 255.255.255.0
 no ip mroute-cache
!
interface Serial0
 no ip address
 shutdown
!
interface Serial1
 no ip address
 shutdown
!
interface Serial2
 ip address 10.1.1.2 255.255.255.0
 encapsulation atm
 no ip route-cache
 no ip mroute-cache
 map-group atm1
 atm pvc 1 1 100 aal5voice 384 192 48
 atm pvc 2 1 200 aal5snap
 serial restart-delay 0
!
interface FR-ATM20
 no ip address
 no ip mroute-cache
 shutdown
!
ip classless
!
map-list atm1
 ip 10.1.1.1 atm-vc 2 broadcast
!
line con 0
line aux 0
line vty 0 4
 password cisco
 login
!
!
voice-port 1/1
!
voice-port 1/2
!
voice-port 1/5
```



```
!
voice-port 1/6
!
dial-peer voice 1 pots
 destination-pattern 1111
 port 1/1
!
dial-peer voice 2 voatm
 destination-pattern 9
 session target Serial2 1
!
end
```