Catalyst 4224 Access Gateway Switch Software Configuration Guide

March 2003

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

Text Part Number: OL-2031-02

Catalyst 4224 Access Gateway Switch Software Configuration Guide

CONTENTS

Preface

This preface contains these sections:

Objectives

This guide explains how to configure basic commands and scenarios for Ethernet switching, IP WAN routing, Voice over IP (VoIP), and IP telephony on the Catalyst 4224 Access Gateway Switch. To use this document effectively, you need to be an experienced data networking professional with a background in telecommunications.

Audience

This guide is intended for network administrators, engineers, and managers who need to understand the Catalyst 4224 system or configure the software. It is also intended for Cisco customer service representatives and system engineers.

Organization

This guide contains the following chapters:

	Title	Description
Chapter 1	Product Overview	Provides an overview of the Catalyst 4224 Access Gateway Switch software features.
Chapter 2	Configuring for the First Time	Describes the initial steps of configuring the Catalyst 4224.
Chapter 3	Configuring Ethernet Switching	Describes how to configure the Ethernet ports and voice VLANs.
Chapter 4	Configuring the Data Interfaces	Describes how to configure the data interfaces for IP WAN routing.
Chapter 5	Configuring the Voice Interfaces	Describes how to configure key voice interfaces.
Chapter 6	Configuring VoIP	Provides comprehensive information on Cisco VoIP configuration.
Chapter 7	Configuring the Eight-Port FXS RJ-21 Module	Describes how to configure the 8-Port FXS Module.
Chapter 8	Configuring Survivable Remote Site Telephony	Describes how to configure survivable remote site telephony.
Chapter 9	Implementing Fax over IP on Cisco Voice Gateways	Provides an overview and configuration information for fax over IP technologies supported on Cisco voice gateways.

	Title	Description
Chapter 10	Traffic Shaping	Describes different types of traffic shaping and provides pointers to configuration and command information.
Chapter 11	Configuring Encryption Services	Describes how to configure encryption services.
Chapter 12	Configuring Other Routing Protocols	Describes how to configure the Novell Internetwork Packet Exchange (IPX) and IBM Systems Network Architecture (SNA) routing protocols.
Appendix A	Command Reference for Voice VLAN	Describes the key voice VLAN commands used on Catalyst 4224.
Appendix B	Synopsis of Basic VoIP Concepts	Describes some basic VoIP concepts.
Appendix C	VoIP Configuration Examples	Provides examples of VoIP configurations.

Related Documentation

The following publications are available for the Catalyst 4224 Access Gateway Switch:

- *Catalyst 4224 Access Gateway Switch Hardware Installation Guide*
- Cisco IOS configuration guides and command references—Use these publications to help you configure the Cisco IOS software.
- For information about MIBs, refer to the following URL:

  http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

Conventions

This guide uses the following conventions:

Convention	Description
boldface font	Commands and keywords are in **boldface**.
italic font	Arguments for which you supply values are in *italics*.
[ ]	Elements in square brackets are optional.
{ x | y | z }	Alternative keywords are grouped in braces and separated by vertical bars.
[x | y | z]	Optional alternative keywords are grouped in brackets and separated by vertical bars.
string	A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.
`screen` font	Terminal sessions and information the system displays are in `screen` font.
`boldface screen` font	Information you must enter is in **`boldface screen`** font.
italic screen font	Arguments for which you supply values are in *italic screen* font.
——→	`This pointer highlights an important line of text in an example.`
^	The symbol ^ represents the key labeled Control—for example, the key combination ^D in a screen display means hold down the Control key while you press the D key.
< >	Nonprinting characters, such as passwords are in angle brackets.

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Obtaining Documentation

Cisco provides several ways to obtain documentation, technical assistance, and other technical resources. These sections explain how to obtain technical information from Cisco Systems.

Cisco.com

You can access the most current Cisco documentation on the World Wide Web at this URL:

http://www.cisco.com/univercd/home/home.htm

You can access the Cisco website at this URL:

http://www.cisco.com

International Cisco websites can be accessed from this URL:

http://www.cisco.com/public/countries_languages.shtml

Documentation CD-ROM

Cisco documentation and additional literature are available in a Cisco Documentation CD-ROM package, which may have shipped with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or through an annual subscription.

Registered Cisco.com users can order the Documentation CD-ROM (product number DOC-CONDOCCD=) through the online Subscription Store:

http://www.cisco.com/go/subscription

Ordering Documentation

You can find instructions for ordering documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/es_inpck/pdi.htm

You can order Cisco documentation in these ways:

- Registered Cisco.com users (Cisco direct customers) can order Cisco product documentation from the Networking Products MarketPlace:

  http://www.cisco.com/en/US/partner/ordering/index.shtml

- Registered Cisco.com users can order the Documentation CD-ROM (Customer Order Number DOC-CONDOCCD=) through the online Subscription Store:

  http://www.cisco.com/go/subscription

- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco Systems Corporate Headquarters (California, U.S.A.) at 408 526-7208 or, elsewhere in North America, by calling 800 553-NETS (6387).

Documentation Feedback

You can submit comments electronically on Cisco.com. On the Cisco Documentation home page, click **Feedback** at the top of the page.

You can e-mail your comments to bug-doc@cisco.com.

You can submit your comments by mail by using the response card behind the front cover of your document or by writing to the following address:

Cisco Systems
Attn: Customer Document Ordering
170 West Tasman Drive
San Jose, CA 95134-9883

We appreciate your comments.

Obtaining Technical Assistance

Cisco provides Cisco.com, which includes the Cisco Technical Assistance Center (TAC) Website, as a starting point for all technical assistance. Customers and partners can obtain online documentation, troubleshooting tips, and sample configurations from the Cisco TAC website. Cisco.com registered users have complete access to the technical support resources on the Cisco TAC website, including TAC tools and utilities.

Cisco.com

Cisco.com offers a suite of interactive, networked services that let you access Cisco information, networking solutions, services, programs, and resources at any time, from anywhere in the world.

Cisco.com provides a broad range of features and services to help you with these tasks:

- Streamline business processes and improve productivity
- Resolve technical issues with online support
- Download and test software packages
- Order Cisco learning materials and merchandise
- Register for online skill assessment, training, and certification programs

To obtain customized information and service, you can self-register on Cisco.com at this URL:

http://www.cisco.com

Technical Assistance Center

The Cisco TAC is available to all customers who need technical assistance with a Cisco product, technology, or solution. Two levels of support are available: the Cisco TAC website and the Cisco TAC Escalation Center. The avenue of support that you choose depends on the priority of the problem and the conditions stated in service contracts, when applicable.

We categorize Cisco TAC inquiries according to urgency:

- Priority level 4 (P4)—You need information or assistance concerning Cisco product capabilities, product installation, or basic product configuration.
- Priority level 3 (P3)—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- Priority level 2 (P2)—Your production network is severely degraded, affecting significant aspects of business operations. No workaround is available.
- Priority level 1 (P1)—Your production network is down, and a critical impact to business operations will occur if service is not restored quickly. No workaround is available.

Cisco TAC Website

You can use the Cisco TAC website to resolve P3 and P4 issues yourself, saving both cost and time. The site provides around-the-clock access to online tools, knowledge bases, and software. To access the Cisco TAC website, go to this URL:

http://www.cisco.com/tac

All customers, partners, and resellers who have a valid Cisco service contract have complete access to the technical support resources on the Cisco TAC website. Some services on the Cisco TAC website require a Cisco.com login ID and password. If you have a valid service contract but do not have a login ID or password, go to this URL to register:

http://tools.cisco.com/RPF/register/register.do

If you are a Cisco.com registered user, and you cannot resolve your technical issues by using the Cisco TAC website, you can open a case online at this URL:

http://www.cisco.com/en/US/support/index.html

If you have Internet access, we recommend that you open P3 and P4 cases through the Cisco TAC website so that you can describe the situation in your own words and attach any necessary files.

Cisco TAC Escalation Center

The Cisco TAC Escalation Center addresses priority level 1 or priority level 2 issues. These classifications are assigned when severe network degradation significantly impacts business operations. When you contact the TAC Escalation Center with a P1 or P2 problem, a Cisco TAC engineer automatically opens a case.

To obtain a directory of toll-free Cisco TAC telephone numbers for your country, go to this URL:

http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

Before calling, please check with your network operations center to determine the level of Cisco support services to which your company is entitled: for example, SMARTnet, SMARTnet Onsite, or Network Supported Accounts (NSA). When you call the center, please have available your service agreement number and your product serial number.

Obtaining Additional Publications and Information

Information about Cisco products, technologies, and network solutions is available from various online and printed sources.

- The *Cisco Product Catalog* describes the networking products offered by Cisco Systems as well as ordering and customer support services. Access the *Cisco Product Catalog* at this URL:

  http://www.cisco.com/en/US/products/products_catalog_links_launch.html

- Cisco Press publishes a wide range of networking publications. Cisco suggests these titles for new and experienced users: *Internetworking Terms and Acronyms Dictionary, Internetworking Technology Handbook, Internetworking Troubleshooting Guide,* and the *Internetworking Design Guide.* For current Cisco Press titles and other information, go to Cisco Press online at this URL:

  http://www.ciscopress.com

- *Packet* magazine is the Cisco monthly periodical that provides industry professionals with the latest information about the field of networking. You can access *Packet* magazine at this URL:

  http://www.cisco.com/en/US/about/ac123/ac114/about_cisco_packet_magazine.html

- *iQ Magazine* is the Cisco monthly periodical that provides business leaders and decision makers with the latest information about the networking industry. You can access *iQ Magazine* at this URL:

  http://business.cisco.com/prod/tree.taf%3fasset_id=44699&public_view=true&kbns=1.html

- *Internet Protocol Journal* is a quarterly journal published by Cisco Systems for engineering professionals involved in the design, development, and operation of public and private internets and intranets. You can access the *Internet Protocol Journal* at this URL:

  http://www.cisco.com/en/US/about/ac123/ac147/about_cisco_the_internet_protocol_journal.html

- Training—Cisco offers world-class networking training, with current offerings in network training listed at this URL:

  http://www.cisco.com/en/US/learning/le31/learning_recommended_training_list.html

Product Overview

The Cisco Catalyst 4224 Access Gateway Switch (Catalyst 4224) is an integrated switch/router that provides Voice-over-IP (VoIP) gateway and IP telephony services to a small branch office. This section provides an overview of the Catalyst 4224.

This section contains the following topics:

For a synopsis of basic VoIP concepts, see the following section in this manual:

Appendix B, “Synopsis of Basic VoIP Concepts”

For Voice-over-IP (VoIP) configuration examples, see the following section in this manual:

Appendix C, “VoIP Configuration Examples”

Features

The Catalyst 4224 supports the following features:

- 24 10/100 Ethernet ports with inline power and quality of service (QoS) that connect IP telephony phones and PCs
- An integrated eight-port Foreign Exchange Station (FXS) module that connects analog phones, fax machines, modems, key telephone systems (KTS) or voicemail systems for VoIP
- Three modular slots that support up to six ports on a wide variety of cards and can provide connectivity to the public switched telephone network (PSTN) or wide-area network (WAN)
  - Multiflex Voice/WAN Interface Card (VWIC)
  - Voice interface card (VIC)
  - WAN interface card (WIC)
- Onboard hardware-based encryption accelerator

Note These cards can be shared with the Cisco 1600 series, Cisco 1700 series, Cisco 2600 series, and Cisco 3600 series platforms.

The Catalyst 4224 supports Cisco IOS software feature sets from the Cisco IOS Release 12.1.4T. The following Cisco IOS images are available:

- IP Plus (standard)
- IP Plus with Firewall
- IP Plus with IPsec 56
- IP Plus with 3DES
- IP Plus with Firewall and IPsec 56
- IP Plus with Firewall and 3DES
- Optional feature license required to use SRST

Solution

The Catalyst 4224 can be deployed as part of a *centralized call processing* network with a Cisco CallManager and Survivable Remote Site Telephony (SRST) software that provides Ethernet switching, IP routing, VoIP gateway, and IP telephony services for a small branch office.

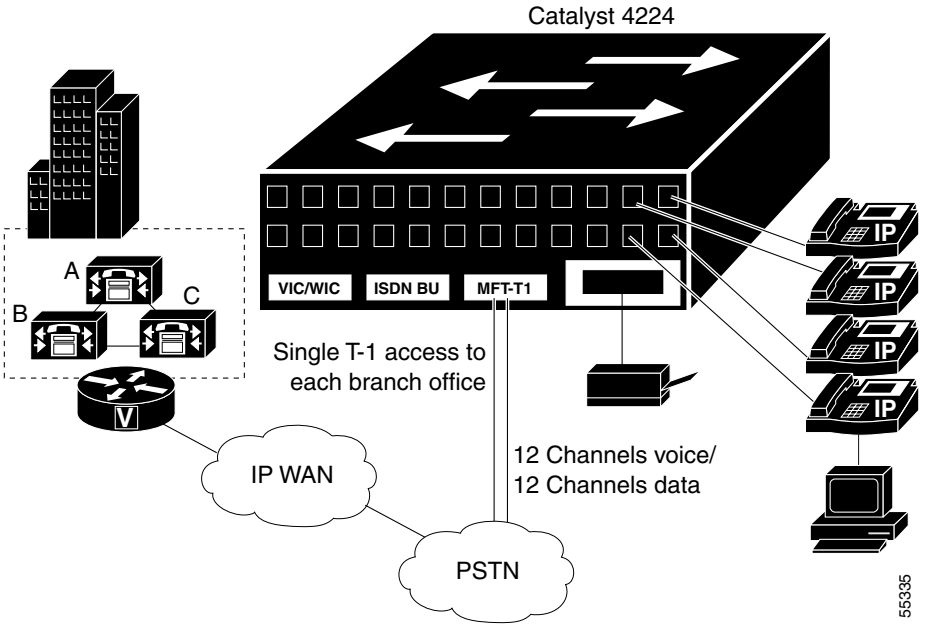
Centralized call processing allows network administrators to deploy and manage IP telephony applications at the corporate headquarters or the corporate data center. Deploying and managing key systems or PBXs in branch offices is no longer necessary. Centralized call processing provides remote branch office users with access to IP telephony applications at centralized locations over the IP WAN.

Centralized call processing has the following benefits:

- Centralized configuration and management
- Remote access at to all Cisco CallManager features
- IT staff not required at each remote site
- Ability to rapidly deploy applications for remote users
- Easy upgrades and maintenance
- Lower total cost of ownership (TCO)

Figure 1-1 shows the Catalyst 4224 at a remote site with a centrally deployed Cisco CallManager at corporate headquarters.

Figure 1-1 Centralized Call Processing Solution



In the diagram, a Cisco CallManager cluster at a central site uses Simple Client Control Protocol (SCCP) to control IP phones at two branch offices. In the branch VoIP network, a Catalyst 4224 acts as an H.323 gateway, interconnecting the analog devices, the PSTN, and IP WAN. This system uses ISDN Basic Rate Interface (BRI) as a lifeline to the PSTN.

If the WAN link (or the Cisco CallManager cluster) becomes unavailable, Survivable Remote Site Telephony (SRST) allows the Catalyst 4224 to keep the IP phones on the branch networks running. Under these circumstances, the Catalyst 4224 functions as an H.323 gateway, thereby ensuring uninterrupted connectivity to the PSTN.

IP Telephony

The term *IP telephony* identifies a networking solution that integrates a switched LAN, the Cisco CallManager, and IP phones.

The Catalyst 4224 is designed to work as part of a centralized Cisco CallManager network that supports up to 24 remote users. As part of an IP telephony solution, the Catalyst 4224 provides:

- 24 ports of switched 10/100 Ethernet connectivity to PCs and servers on a LAN
- Line-powered Ethernet for Cisco IP phones
- Limited backup capability when Cisco CallManager is unavailable

Note The Catalyst 4224 has digital signal processors (DSPs) installed on the motherboard but does not support a *DSP farm*. Transcoding and conferencing services are supplied to the branch office by a central Cisco CallManager.

Ethernet Switching

Using the auxiliary VLAN feature, you can segment phones into separate logical networks even though the data and voice infrastructure are physically the same. The auxiliary VLAN feature places the phones into their own VLANs without the need for end-user intervention. You can plug the phone into the switch, and the switch provides the phone with the necessary VLAN information.

Key Ethernet switch features include:

- Hardware-based Layer 2 switching
- Software-based Layer 3 switching
- Twenty-four 10BASE-T/100BASE-TX auto-sensing ports, each delivering up to 200 Mbps of bandwidth or 100 Mbps of full-duplex bandwidth
- Forwarding and filtering at full wire speed on each port
- Port security that restricts a port to a user-defined group of stations
- Support for up to 8000 unicast and more than 242 multicast addresses

- Protocol Independent Multicast (PIM) and Internet Group Management Protocol (IGMP) snooping
- Per-port broadcast, multicast, and unicast storm control that prevents faulty end stations from degrading overall system performance
- Inline 48-volt DC power
- MAC-based port-level security to prevent unauthorized stations from accessing the switch

Survivable Remote Site Telephony

As enterprises extend IP telephony from central sites to remote offices, it is important to provide backup redundancy at the remote branch office. The Survivable Remote Site Telephony (SRST) software feature on the Catalyst 4224 automatically detects a failure in the network and uses Cisco Simple Network Automated Provisioning (SNAP) to provide call-processing backup for the IP phones in the remote office.

The Catalyst 4224 provides call processing for the duration of the failure and ensures that the phones remain operational. Upon restoration of the WAN and connectivity to the network, the system automatically shifts call-processing functions to the primary Cisco CallManager cluster. Configuration for this capability is done only once in the Cisco CallManager at the central site.

VoIP Gateway

Voice Over IP (VoIP) receives voice traffic at one location, converts it to TCP/IP packets for the benefits of *toll-bypass*, and transports the packets across the WAN to their destination.

To facilitate the migration to VoIP, the Catalyst 4224 includes an integrated high-density eight-port FXS module. These FXS ports can connect analog phones, modems, and fax machines to the Catalyst 4224.

The Catalyst 4224 supports a wide range of voice interface cards with the most popular signaling protocols. Supported protocols and interface types include T1-PRI, E1-PRI, T1-CAS, E1-CAS R2, ISDN BRI, and FXO.

Table 1-1 describes the voice interface cards supported by the Catalyst 4224.

Table 1-1 Voice Interface Cards

Module	Description
VIC-2FXS	Two-port FXS voice/fax interface card
VIC-2FXO	Two-port FXO voice/fax interface card (North American version)
VIC-2FXO-EU	Two-port FXO voice/fax interface card (European version)
VIC-2BRI-S/T-TE	Two-port BRI S/T terminal equipment voice/fax interface card (also supports data)
VWIC-1MFT-T1	One-port T1/Fractional T1 Multiflex Trunk with CSU/DSU
VWIC-2MFT-T1	Dual-port T1/Fractional T1 Multiflex Trunk with CSU/DSU
VWIC-1MFT-E1	One-port E1/Fractional E1 Multiflex Trunk with DSU
VWIC-2MFT-E1	Dual-port E1/Fractional E1 Multiflex Trunk with DSU

Additional VoIP gateway benefits include:

- Private branch exchange (PBX) and PSTN connectivity
- H.323v2 VoIP gateway functions
- Onboard DSPs allocated to voice interfaces
- Fax pass-through and Fax relay
- Modem pass-through

IP Routing and WAN Features

The Catalyst 4224 supports the following WAN features:

- Multilink Point-to-Point Protocol (MLPPP)
- Frame relay

- Asynchronous start-stop on ASCII
- Synchronous PPP

The Catalyst 4224 supports the following IP routing features:

- A high-performance MPC8260 processor running at 200 MHz provides the processing power required for delivering voice, streaming video, and data to the branch office
- Packet-processing capabilities of 35,000 pps at 64-byte Layer 3
- Onboard hardware encryption provides up to 10 times the performance of software-only encryption by offloading the processing from the routing CPU
- Modular WIC interfaces that can be shared with the Cisco 1600 series, Cisco 1700 series, Cisco 2600 series, and Cisco 3600 series platforms

Table 1-2 describes the data interface cards supported by the Catalyst 4224.

Table 1-2 Data Interface Cards

Module	Description
VWIC-1MFT-T1	One-port T1/Fractional T1Multiflex Trunk with CSU/DSU
VWIC-2MFT-T1	Dual-port T1/Fractional T1 Multiflex Trunk with CSU/DSU
VWIC-1MFT-E1	One-port E1/Fractional E1 Multiflex Trunk with DSU
VWIC-2MFT-E1	Dual-port E1/Fractional E1 Multiflex Trunk with DSU
WIC-1DSU-T1	T1/Fractional T1 CSU/DSU
WIC-1DSU-56K4	One-port four-wire 56/64 Kbps CSU/DSU
WIC-1T	One-port high-speed serial
WIC-2T	Dual-port high-speed serial
WIC-2A/S	Dual-port async/sync serial

Quality of Service

The Catalyst 4224 provides the performance and intelligent services of Cisco IOS software for branch office applications. The Catalyst 4224 can identify user applications—such as voice or multicast video—and classify traffic with the appropriate priority levels. Quality of service (QoS) policies are enforced using Layer 2 and 3 information such as 802.1p and IP precedence. The Catalyst 4224 queues use weighted random early detection (WRED), weighted round-robin (WRR), and type-of-service/class-of-service (ToS/CoS) mapping to ensure that QoS is maintained as packets traverse the network.

To ease the deployment of QoS, the Catalyst 4224 supports Cisco QoS Policy Manager (QPM). QPM is a complete policy management tool that enables provisioning of end-to-end differentiated services across network infrastructures with converged voice, video, and data applications. The combination of QPM and CiscoWorks Service Management Solution enables network administrators to adjust service levels in accordance with defined QoS policies. The end result is network-wide intelligent and consistent QoS that enables performance protection for voice applications while reducing costs for growing networks.

VPN and Firewall Features

The Catalyst 4224 provides the same security to voice and video networks that is available for data networks. The Catalyst 4224 supports the optional Cisco IOS Software Firewall Feature Set, IP Security (IPsec) with data encryption standard (DES), and Triple DES (3DES). Hardware encryption using the onboard encryption accelerator provides higher performance than software-based encryption, and frees processor capacity for other services.

The Catalyst 4224 supports the following encryption features:

- 56-bit DES encryption using Cipher Block Chaining (CBC) mode
- 168-bit 3DES encryption using CBC mode
- MD5 and SHA-1 hashing, including support for the HMAC transform with IPsec AH and ESP
- Support for Dif-fie-Hellman key exchange
- RSA and DSA public key signature and verification (when implemented by IOS IPsec Crypto Engine)

Note DES and 3DES software is controlled by U.S. export regulations on encryption products. For additional details visit the following URL:

http://www.cisco.com/wwl/export/crypto/

Application Notes

This section contains the following topics:

Architecture

The Catalyst 4224 consists of the following physical subsystems:

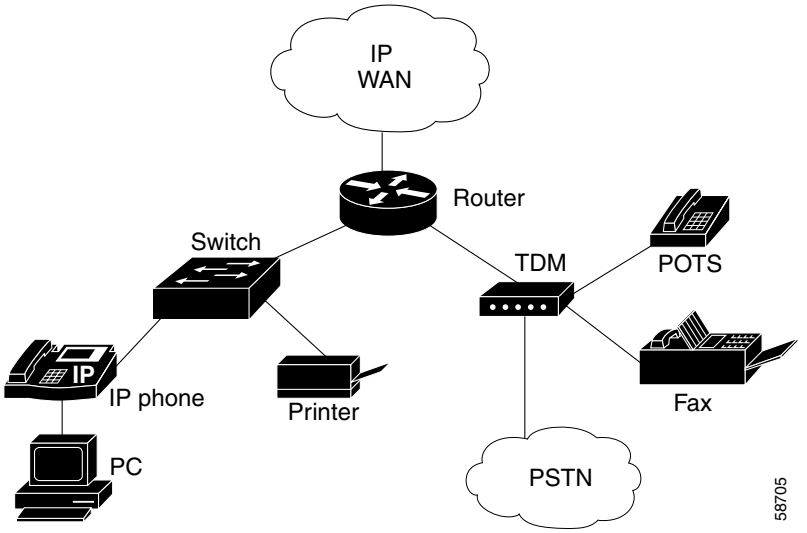
- Time-division multiplexing (TDM) subsystem—Two VIC/WIC/VWIC slots, one VIC slot, a built-in high-density FXS module, and a TDM switch
- CPU subsystem—Supports routing functions and interfaces for the TDM and switch subsystems
- Switch subsystem—24 10/100 Ethernet port switch with QoS and interfaces to the CPU and power subsystems

- DSP subsystem—Interfaces to the TDM subsystem and CPU subsystem for converting voice streams to IP packets
- Power subsystem—Provides power to the Catalyst 4224 and inline power to the IP phones that connect to the 10/100 Ethernet ports

From a logical view, the Catalyst 4224 looks like a router that connects an Ethernet switch and a TDM switch in one system.

Figure 1-2 shows a logical view of the Catalyst 4224.

Figure 1-2 Logical View of Catalyst 4224



DSP Allocation

The Catalyst 4224 has six digital signal processors (DSPs) installed on the motherboard. The DSPs convert voice signals into data packets and data packets into voice signals. The DSPs on the Catalyst 4224 do not perform transcoding and hardware conferencing. These services are performed at the central site by Cisco CallManager. The DSPs only support VoIP. They do not support Voice over Frame Relay (VoFR) and Voice over ATM (VoATM). The DSPs are not field upgradeable.

The DSPs compress and decompress packets based on codecs. The Catalyst 4224 supports the following codecs:

- G.711 a-law 64 Kbps
- G.711 mu-law 64 Kbps
- G.729 abr 8 Annex-A & B 8 Kbps
- G.729 ar8 G729 Annex-A 8 Kbps
- G 729 r8 G729 8 Kbps

The number of DSP channels that you can use depends upon the VIC configuration. The following rules apply when you allocate DSP channels for T-1/E-1 VWICs:

- The eight-port FXS Module uses two of the six DSPs by default, leaving four DSPs to configure for digital voice.
- The maximum number of T1/E1 DS0s that you can configure is 24 on the six available DSPs. The eight-port FXS module must be disabled using the CLI; otherwise, only 16 channels are supported by four available DSPs.
- Only use DSP channels if you configure voice DS0s.
- Each set of four DS0s uses one DSP.
- One entire DSP is used even if less than four DS0s are configured. Five DS0s use two DSPs.
- T1/E-1 DS0 channels cannot be used for analog channels.

Note You can disable the eight-Port FXS Module using the command line interface (CLI), and thereby free up eight DSP channels for additional digital voice channels.

The following rules apply when you allocate DSP channels for BRI, FXS, and FXO VICs:

- The two-Port BRI VIC uses two of the available four DS0 channels if you configure voice. You can use the other two DS0s for voice FXS or FXO, but not T-1/E-1.

- The two-Port FXS and FXO VIC uses two of the available four DSO channels. The DSP is used when the VIC is plugged in, even if the ports are not configured.
- The eight-Port FXS Module uses eight DS0 channels or two DSPs, even if it is not used (unless it is disabled using the CLI).

The following sample configuration shows DSP allocations:

```
C4224# sh voice dsp
                                      BOOT
TYPE DSP CH CODEC    VERS STATE STATE   RST AI PORT    TS
==== === == ======== ==== ===== ======= === == ======= ==

5409 001 00 {anlgHC}   .8 IDLE  idle      0  0 3/0       0
         01 {anlgHC}   .8 IDLE  idle      0  0 3/1       0
5409 002 00 {anlgMC}   .3 IDLE  idle      0  0 4/0       0
         01 {anlgMC}   .3 IDLE  idle      0  0 4/1       0
         02 {anlgMC}   .3 IDLE  idle      0  0 4/2       0
         03 {anlgMC}   .3 IDLE  idle      0  0 4/3       0
5409 003 04 {anlgMC}   .3 IDLE  idle      0  0 4/4       0
         05 {anlgMC}   .3 IDLE  idle      0  0 4/5       0
         06 {anlgMC}   .3 IDLE  idle      0  0 4/6       0
         07 {anlgMC}   .3 IDLE  idle      0  0 4/7       0
5409 004 00 {medium}  3.5 IDLE  idle      0  0 2/0:1     1
         01 {medium}   .8 IDLE  idle      0  0 2/0:1     2
         02 {medium}      IDLE  idle      0  0 2/0:1     3
         03 {medium}      IDLE  idle      0  0 2/0:1     4
5409 005 00 {medium}  3.5 IDLE  idle      0  0 2/0:1     5
         01 {medium}   .8 IDLE  idle      0  0 2/0:1     6
         02 {medium}      IDLE  idle      0  0 2/0:1     7
         03 {medium}      IDLE  idle      0  0 2/0:1     8
5409 006 00 {medium}  3.5 IDLE  idle      0  0 2/0:1     9
         01 {medium}   .8 IDLE  idle      0  0 2/0:1    10
         02 {medium}      IDLE  idle      0  0 2/0:1    11
         03 {medium}      IDLE  idle      0  0 2/0:1    12
```

In the sample configuration, port 3/0 is an analog VIC that uses two of the four channels in DSP 1. Slot 4 contains the eight-port FXS Module. This module takes up two DSPs. DSP four, five, and six are being used for the 12 voice channels on a MFT VWIC.

DSP resources are used for signaling and voice bearer channels. The signaling channel is used for detecting off-hook/on-hook transitions.

InterVLAN Routing

The forwarding performance for interVLAN routing on the Catalyst 4224 is 35 Kpps for 64-byte packets. Fast Switching is the default switching path. The Catalyst 4224 supports Cisco Express Forwarding (CEF).

Quality of Service

The Catalyst 4224 can function as a Layer 2 switch connected to a Layer 3 router. When a packet enters the Layer 2 engine directly from a switch port, it is placed into one of four queues in the dynamic, 32-Mbyte shared memory buffer. The queue assignment is based on the dot1p value in the packet. Any voice bearer packets that come in from the IP phones on the voice VLAN are automatically placed in the highest priority (Queue 3) based on the 802.1p value generated by the IP phone. The queues are then serviced on a WRR basis. The control traffic, which uses a CoS/ToS of 3, is placed in Queue 2.

Layer 2 QoS

Table 1-3 summarizes the queues, CoS values, and weights for Layer 2 QoS on the Catalyst 4224.

Table 1-3 Queues, CoS values, and Weights for Layer 2 QoS

Queue Number	CoS Value	Weight
3	5,6,7	255
2	3,4	64
1	2	16
0	0,1	1

The weights specify the number of packets that are serviced in the queue before moving on to the next queue. Voice Real-Time Transport Protocol (RTP) bearer traffic marked with a CoS /ToS of 5 and Voice Control plane traffic marked with

a CoS/ToS of 3 are placed into the highest priority Queues. If the queue has no packets to be serviced, it will be skipped. WRED is not supported on the Fast Ethernet ports.

The WRR default values cannot be changed. There are currently no CLI commands to determine QoS information for WRR weights and queue mappings. You cannot configure port-based QoS on the Layer 2 switch ports.

Separate Voice and Data VLANs

To be consistent with Cisco IP Telephony QoS design recommendations, you should configure separate voice and data VLANs. The following sample configuration shows how to configure separate voice and data VLANs.

```
interface FastEthernet5/22
 no ip address
 duplex auto
 speed auto
 switchport access vlan 60
 switchport voice vlan 160
 snmp trap link-status
```

Packets arriving on the specified voice VLAN will automatically have the 802.1p priority values read on ingress. Unlike the Catalyst 3500, trunking mode does not have to be used to distinguish between a voice and a data VLAN on a single port.

Single Voice and Data VLAN with dot1p

If the voice and data VLAN must be the same (a single subnet), using the dot1p extension will enable the Catalyst 4224 to recognize the dot 1p CoS value from the IP phone and place the packet in a queue based on the 802.1p value. The following sample configuration shows how to configure a single voice and data VLAN with dot1p.

```
interface FastEthernet5/23
 no ip address
 duplex auto
 speed auto
 switchport access vlan 160
 switchport voice vlan dot1p
```

Similar to other voice-enabled Catalyst platforms, the Catalyst 4224 learns that an IP phone is attached to the port via the CDP message exchange.

Layer 3 QoS

You can configure QoS on the Layer 3 CPU from the CLI, which is very similar to the interface on the Cisco 1750, Cisco 2600 series, and Cisco 3600 series routers.

WAN QoS Queuing and Scheduling

The Catalyst 4224 supports WAN QoS queuing and scheduling. Table 1-4 shows Catalyst 4224 WAN QoS queuing and scheduling features.

Table 1-4 Catalyst 4224 WAN QoS Queuing and Scheduling Features

Frame Relay	MLPPP	PPP	HDLC
No LLQ/CBWFQ	No LLQ/CBWFQ	LLQ/CBWFQ	LLQ/CBWFQ
IP RTP Priority	No IP RTP Priority	IP RTP Priority with CBWFQ	IP RTP Priority
FRF.12	LFI		

The Service Policy command is disabled for Frame Relay.

Summary of the Layer 3 WAN QoS Features

In summary, the Catalyst 4224 supports the following Cisco IOS Layer 3 WAN QoS features:

- Classification and Marking
  - Access control lists (ACL)
  - Class-based marking
  - Class-based matching
  - Committed Access Rate (CAR)

  - Differentiated services code point (DSCP) marking
  - IP Precedence
  - L2 Marking
  - L2 Matching
  - Match RTP
  - Network-based Application Recognition (NBAR)
  - Policy-based Routing (PBR)
  - QoS Preclassification for tunnels
- Congestion Avoidance:
  - Flow-based RED (FRED)
  - Weighted RED (WRED)
  - WRED with DSCP
- Policing and Traffic Shaping:
  - Class-based policing
  - Class-based shaping
  - IP Precedence
  - Frame Relay Traffic Shaping (FRTS)
  - General Traffic Shaping (GTS)
- Link Efficiency:
  - cRTP fast/CEF-switching
  - FR LFI (FRF.12)
  - MLPPP/LFI
  - RTP Header Compression (cRTP)

Configuration Guidelines

This section provides platform specific guidelines for configuring the Catalyst 4224.

This section contains the following topics:

For Voice over IP (VoIP) configuration examples, see the following section in this manual:

Appendix C, “VoIP Configuration Examples”

Default Port Configuration

The Catalyst 4224 boots up as a Layer 2 switch. The following sample configuration shows a default port configuration.

```
C4224_SF# sh run int fas5/7

Building configuration...
Current configuration : 96 bytes
interface FastEthernet5/7
 no ip address
 duplex auto
 speed auto
 snmp trap link-status
end
```

Separate VLAN for Voice and Data

Unlike the Catalyst 3500, you do not need to preconfigure VLANs with a VLAN database command. To be consistent with Cisco IP Telephony QoS design guidelines, you should configure a separate VLAN for voice and data. The following example shows a recommended configuration.

```
interface FastEthernet5/22
 no ip address
 duplex auto
 speed auto
 switchport access vlan 60
 switchport voice vlan 160
 snmp trap link-status
 spanning-tree portfast
```

This sample configuration instructs the IP phone to generate a packet with an 802.1q VLAN ID of 160 and an 802.1p value of 5 (default for voice bearer traffic).

Note Portfast is supported only on nontrunk ports.

Port Configuration for a Single Subnet

If you have only a single subnet available, use the same subnet for voice and data. The following sample configuration shows a port configuration for a single subnet.

```
interface FastEthernet5/23
 no ip address
 duplex auto
 speed auto
 switchport access vlan 160
 switchport voice vlan dot1p
 snmp trap link-status
 spanning-tree portfast
```

This sample configuration instructs the IP phone to generate an 802.1 Q frame with a null VLAN ID value and an 802.1p value (default is CoS of 5 for bearer traffic). The voice VLAN and data VLAN are both 160 in this example.

InterVLAN and WAN Routing Configuration

Configuring interVLAN routing on the Catalyst 4224 is identical to configuring interVLAN routing on the Catalyst 6000 with an MSFC. Configuring an interface for WAN routing is consistent with other Cisco IOS platforms. The following sample shows a configuration for interVLAN routing.

```
interface Vlan 160
 description Voice VLAN
 ip address 10.6.1.1 255.255.255.0

interface Vlan 60
 description Data VLAN
 ip address 10.60.1.1 255.255.255.0

interface Serial1/0
 ip address 160.3.1.2 255.255.255.0
```

The Catalyst 4224 supports standard IGP routing protocols such as RIP, Interior Gateway Routing Protocol (IGRP), Enhanced Interior Gateway Routing Protocol (EIGRP), and open shortest path first (OSPF). It also supports multicast routing for PIM dense mode, sparse mode, and sparse-dense mode.

Centralized Cisco CallManager and DHCP Server

In a centralized Cisco CallManager deployment model, the DHCP server would probably be located across the WAN link. You should include an **ip helper** command on the voice VLAN interface that points to the DHCP server so that the IP phone can obtain the IP address and the TFTP server address. The following sample configuration shows a configuration for **ip helper-address** on the voice VLAN:

```
interface Vlan 160
 description Voice VLAN
 ip address 10.6.1.1 255.255.255.0
 ip helper-address 172.20.73.14
```

As an alternative, you could use the Cisco IOS DHCP server capabilities on the Catalyst 4224. The following sample configuration shows a configuration for the DHCP configuration options.

```
C4224_SF(config)# ip dhcp pool SF
C4224_SF(dhcp-config)# ?
```

```
DHCP pool configuration commands:

  client-identifier    Client identifier
  client-name          Client name
  default-router       Default routers
  dns-server           DNS servers
  domain-name          Domain name
  hardware-address     Client hardware address
  host                 Client IP address and mask
option Raw DHCP options

C4224_SF(dhcp-config)# option 150 ip ?

  Hostname or A.B.C.D  Server's name or IP address
```

Note DHCP option 150 is supported locally. This local support provides the IP address of the TFTP server, which has the IP phones' configuration. An **ip helper-address** would not be required in this case because the IP phone has its IP address and the TFTP server address. The configuration request to the TFTP server is a unicast packet.

Voice Port Configuration

You configure voice ports on the Catalyst 4224 as you would in standard Cisco IOS software. The following sample configuration shows a configuration for the eight-port FXS Module:

```
dial-peer voice 41 voip
 destination-pattern 1...
 session target ipv4:172.20.73.13
 codec g711ulaw
!
dial-peer voice 1005 pots
 destination-pattern 1005
 port 4/0
```

Interface Range Command Support

You can use the range command. The following sample configuration shows how to configure the range command:

```
C4224_SF(config)# int range fas5/2 - 5
switchport access vlan 60
switchport voice vlan 160
```

Switched Port Analyzer (SPAN)

Switched Port Analyzer (SPAN), also known as port monitoring, is supported for up to two sessions. Spanning a VLAN is not supported. You can only span selected interfaces. The following sample configuration shows a configuration for setting a port monitor session with the range command:

```
C4224_SF(config)# monitor session 1 ?
  destination  SPAN destination interface or VLAN
  source       SPAN source interface or VLAN
```

Recommended Configurations

This section contains the following topics:

- Maximum Number of VLAN and Multicast Groups, page 1-29
- IP Multicast Support, page 1-29

No VTP or DTP Support

Using the interface **switchport access** or **switchport trunk** VLAN commands automatically creates a voice VLAN and data VLAN. If you require an additional VLAN beyond the voice and data VLAN when connecting to another switch, you must add it manually using the VLAN database command from the EXEC prompt.

Creating a VLAN

The following sample configuration shows how to define a VLAN manually:

```
C4224(vlan)# vlan 10 name external_switch_Vlan10

VLAN 10 added:
        Name: external_switch_Vlan10
C4224(vlan)#vlan 11 name external_switch_Vlan11
VLAN 11 added:
        Name: external_switch_Vlan11
C4224(vlan)#exit
APPLY completed.
Exiting...
```

Defining a VLAN on a Trunk Port

The following sample configuration shows how to define a VLAN on a trunk port.

```
C4224(config-if)# switchport trunk allowed vlan 10
C4224(config-if)# switchport trunk allowed vlan add 11
```

By default, the trunk interface accepts all VLANs created by the VLAN database. Therefore, you should use the **switchport trunk allowed** command to delete unwanted VLANs from the interface.

Trunking

The Catalyst 4224 supports only dot1Q trunking. Dynamic Trunking Protocol (DTP) is not supported. A Catalyst switch that is trunked to the Catalyst 4224 must have the trunking mode set to either **On** or **No negotiate** and type **dot1q**.

Fractional PRI Configuration

The maximum of 16 channels are available for trunk voice ports. This can cause a problem when a PSTN or PBX uses an unavailable channel to send a call to the Catalyst 4224. To prevent this type of problem, follow this procedure:

Step 1 Always configure the Primary Rate Interface (PRI) VIC last (after you configure all the VIC cards that require the DSP resources).

Step 2 Allocate all 24 time slots for the PRI group. The following sample configuration shows a configuration for the range command.

```
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
```

Note The DSP resources are not sufficient for the 24 time slots. Therefore you receive this message: *insufficient DSP resources*. Ignore this message.

Note You need to tell the switch or PBX to make the time slots out-of-service. If you do not allocate 24 time slots, a SERVICE message is not sent for the unallocated time slots. This is an important caveat.

Step 3 Use the **show voice dsp** command to see how many channels are allocated with the available DSP resources. In a test case, 16 time slots could be allocated DSP resources.

Step 4 Busy-out the time slots for which DSP resources could not be allocated. The following sample configuration shows how to busy-out the time slots:

```
isdn service dsl 0 b_channel 17-24 state 2
```

Step 5 Use the **show isdn service** command to ensure that the channel is out-of-service. The following sample configuration shows channels in service:

```
C4224# sh isdn ser
PRI Channel Statistics:
ISDN Se1/0:23, Channel [1-24]
  Configured Isdn Interface (dsl) 0
   Channel State (0=Idle 1=Proposed 2=Busy 3=Reserved 4=Restart
5=Maint_Pend)
    Channel :   1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4
    State   :     0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 3

   Service State (0=Inservice 1=Maint 2=Outofservice)
    Channel :  1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4
    State   :    0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 2 2 2 2 2 2 2 2
```

For incoming calls, only time slots 1-16 are used by the switch or PBX. North American CO ISDN switches should support SERVICE/SERVCE ACK messages for maintenance service of B-channels on the PRI span. For the CO switches that do not support this service, you must ask the provider to busy-out the channels.

No Ring Back Tone Generated

When receiving or placing a call from an ISDN terminal (T1/E1 PRI), there is no Progress IE in the setup. (Progress IE = 0.) The Catalyst 4224 does not generate ringback when it receives an alert from Cisco CallManager. You can avoid this situation and force the Catalyst 4224 to generate a ringback using the progress indicator commands on the VoIP and POTS dial-peer statements. The following sample configuration shows how to generate a ringback.

```
dial-peer voice 500 voip
 destination-pattern 5...
 progress_ind setup enable 3
 session target ipv4:10.200.73.15
 codec g711ulaw

dial-peer voice 300 pots
 destination-pattern 1...
```

```
progress_ind alert enable 8
port 3/1:23
forward-digits all
```

Note **alert enable 8** is a hidden command option, which you cannot find by using the ? at the CLI.

The following sample configuration shows what happens when you try to find this command option:

```
C4224-2(config-dial-peer)# progress_ind alert ?

% Unrecognized command
```

This ringback situation applies only to PRI. It does not apply to BRI.

MTP Required on Cisco CallManager

Prior to support for H.323 Version 2, you needed to enable the MTP Required checkbox. This checkbox is located in the Catalyst 4224 H.323 Gateway Configuration page in Cisco CallManager to define an H.323 gateway. All Cisco IOS H.323 gateways with Cisco IOS 12.07 or later now support H.323 Version 2. You should not ordinarily enable this checkbox box when defining the Catalyst 4224 as an H.323 Gateway. The only time you should check the box is if transcoding is used at the central site. Transcoding would be necessary in situations where the Catalyst 4224 uses G.729 for IP WAN calls and a voice mail system at the central site supports G.711 only.

If you enable MTP Required on the Catalyst 4224 H.323 Gateway, analog POTS calls to an IP phone locally connected will traverse the IP WAN. The call between the analog FXS POTS and the IP Phone is anchored at the central site transcoding device. This is normal behavior for a Cisco IOS H.323 gateway when MTP Required is enabled. This leads to performance that is not optimal. Therefore, unless transcoding is required, Media Termination Point (MTP) should *not* be enabled on the H.323 Gateway definition for Catalyst 4224. Another option would be to use G.711 across the IP WAN.

The following sample configurations show how the VoIP endpoints can be verified.

With the MTP Required checkbox enabled:

```
The Catalyst 4224 is 10.253.1.1
Transcoder is Catalyst 6000 at Central :  10.1.1.11

C4224_SF# sh ip sock

Proto    Remote      Port      Local       Port  In Out Stat TTY
OutputIF

 17 10.1.1.11       16541 10.253.1.1      18757   0   0    1   0
```

With the MTP Required checkbox not enabled:

```
IP Phone is 10.6.1.4

C4224_SF# sh ip sock

Proto    Remote      Port      Local       Port  In Out Stat TTY
OutputIF

 17 10.6.1.4        26287 10.253.1.1      17047   0   0    1   0
```

When the MTP Required checkbox is enabled, a call between a local IP phone and an FXS analog POTS connected to the Catalyst 4224 is anchored at the transcoder at the Central site, and local calls use WAN bandwidth. When the MTP Required checkbox is not enabled, the local FXS-to-IP phone call is directly connected between the Catalyst 4224 H.323 gateway and the IP phone.

H323-Gateway VOIP Bind SRCADDR Command

You must always use the following command to configure the IP address of the gateway. This command ensures that the IP address included in the H.323 packet is deterministic—it consistently indicates the same address for the source. If the end point is non-deterministic, the call is anchored at the serial interface port and silence or one-way audio occurs. The following sample configuration shows the H323-gateway **voip bind srcaddr** command:

```
interface Loopback1
 description h323 gateway address
 ip address 10.253.1.1 255.255.255.0
 h323-gateway voip bind srcaddr 10.253.1.1
```

You must place the **h323 bind interface** command on the interface with the IP address that Cisco CallManager uses to define the H.323 gateway.

Port Fast Not Enabled on Trunk Ports

You should configure ports as switched access ports. However, there may be implementations where your requirements dictate configuring ports as trunks and you want to standardize this configuration across all 24 ports. This is not the recommended configuration. Because of DHCP request timeouts on Windows 95/NT, portfast may be a desirable feature. However, portfast is not supported on ports in trunking mode. To reduce the forwarding delay time of a port, use the global configuration commands for the specific VLAN and reduce the forwarding timers to the minimum value of four seconds. See the following sample configuration:

```
spanning-tree portfast bpduguard
spanning-tree vlan 60 forward-time 4
spanning-tree vlan 160 forward-time 4
```

These commands configure VLAN ports to forward data in eight seconds.

If you connect Catalyst 4224 to another switch, make sure that the timers are the same value. Otherwise Spanning Tree issues may arise. Also note that only 802.1Q trunking is supported on the Catalyst 4224.

Priority Queuing on Frame Relay

Frame relay does not support LLQ/CBWFQ. The service policy output command is currently disabled. Therefore, only **ip rtp priority** is supported as a voice priority queuing scheme. FRF.12 is supported on frame relay links.

The recommended configuration for FRF.12 and **ip rtp priority** is shown in the following procedure. The parameter values are for illustrative purposes only. Your values may differ.

Step 1 Define the appropriate map class. The following sample configuration shows how to define the map class:

```
map-class frame-relay VOIP_256
 no frame-relay adaptive-shaping
 frame-relay cir 250000
 frame-relay bc 1000
 frame-relay be 0
 frame-relay mincir 250000
 frame-relay fair-queue
```

```
 frame-relay fragment 320
 frame-relay ip rtp priority 16384 16383 170
```

Step 2 Apply the map class to a frame relay sub interface. The following sample configuration shows how to apply the map class:

```
interface Serial0/0.300 point-to-point
 ip address 1.1.1.1
 frame-relay interface-dlci 300
 frame-relay class VOIP_256
```

Step 3 Apply frame relay shaping at the main interface. The following sample configuration shows how to apply shaping:

```
interface ser0/0
encapsulation frame-relay
frame-relay traffic-shaping
```

Maximum Number of VLAN and Multicast Groups

The maximum number of VLANs multiplied by the number of multicast groups must be less than or equal to 242. For example, the number for 10 VLANs and 20 groups would be 200, which is within the 242 limit.

IP Multicast Support

The maximum number of multicast groups is related to the maximum number of VLANs. The product of the number of multicast groups and the number of VLANs cannot exceed 242. Multicast support includes the following items:

- Support for sparse mode, dense mode, and sparse-dense mode
- IGMP snooping Versions 1, 2, and 3

Configuring for the First Time

This section describes the initial steps of configuring the Catalyst 4224 and outlines the features of the Cisco IOS command line interface (CLI). Use this tool when you configure Catalyst 4224 interfaces.

This section contains the following topics:

First-Time Configuration

This section contains the following topics:

Booting the Catalyst 4224

The factory configures the Catalyst 4224 to automatically load a Cisco IOS image. The software configuration register in the Catalyst 4224 determines where to find the image. The factory sets this register to load the Cisco IOS image into boot flash memory from configuration register 0x0101. This register enables autoboot at register 0x0103.

Table 2-1 shows the Catalyst 4224 default configuration.

Table 2-1 Catalyst 4224 Default Configuration

Feature	Default Value
Host name	Router
Interface configuration	None
VLAN configuration	None
Password encryption	Disabled
Break to console	Ignore

After booting the Catalyst 4224 for the first time, you can configure the interfaces and then save the configuration to a file in NVRAM.

Downloading an Image to Boot Flash Memory

Note Before you can download an image, you must first configure the management port. See "Configuring the Management Port" section on page 2-3.

If you have already configured the Catalyst 4224, you can download a run-time image from a TFTP server on the network. TFTP downloads can take place over the Ethernet management port.

To download an image to boot flash memory, use privileged mode to enter the following command:

copy tftp: [*/directory*] */filename* [*/directory*] */filename*

Connecting a Terminal

To connect a terminal to the console port using the cable and adapters provided with the Catalyst 4224, connect to the port using the RJ-45-to-RJ-45 cable and RJ-45-to-DB-25 DTE adapter or RJ-45-to-DB-9 DTE adapter (labeled Terminal).

Check the documentation that came with your terminal to determine the baud rate. The baud rate of the terminal must match the default baud rate (9600 baud) of the console port.

Set up the terminal as follows:

- 9600 baud
- Eight data bits
- No parity
- Two stop bits
- No flow control

Connecting a Modem

Connect the modem to the port using the RJ-45-to-RJ-45 cable and the RJ-45-to-DB-25 DCE adapter (labeled Modem).

Configuring the Management Port

You can manage the Catalyst 4224 through the 10/100 management port by assigning it an IP address.

Caution By default, the Fast Ethernet interface does not route data traffic. Cisco recommends that you do not override this default configuration.

If the Ethernet 10/100 management port is up and an IP address has been configured, the Catalyst 4224 selects the IP address assigned to the 10/100 Ethernet management port.

If the selected network management IP address is removed or the interface or subinterface associated with this IP address is shut down, the Catalyst 4224 selects another IP address as a replacement.

If all the interfaces are down or no IP address has been assigned to any interface or subinterface that is running, the IP address for network management is 0.0.0.0.

Interface Numbering

The Catalyst 4224 has three slots in which you can install interface cards:

- Slot 1 supports voice interface cards (VICs), WAN interface cards (WICs), and voice and WAN interface cards (VWICs).
- Slot 2 supports VICs, WICs, and VWICs.
- Slot 3 supports VICs and VWICs but does not support WICs.
- Slot 4 supports an eight-Port FXS RJ21 Module.
- Slot 5 supports 10/100 Ethernet switching ports.

Each individual interface is identified by a slot number and a port number. The slots are numbered as follows:

- Slot 0 supports the following interfaces embedded in the mainboard:
  - Console port (con 0)
  - Ethernet Management port (Fast Ethernet 0/0)
- Slot 1 ports are numbered from right to left (1/1 and 1/0).

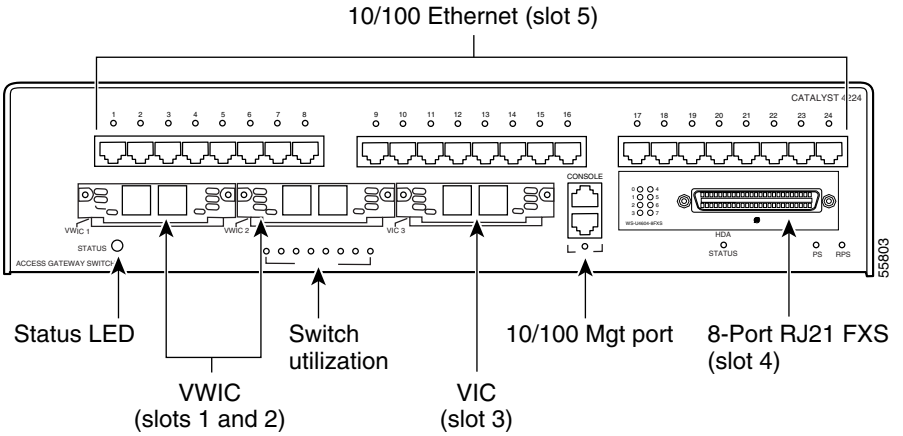
Note On the WIC-2A/S, the top slot is 1 and the bottom slot is 0.

- Slot 2 ports are numbered from right to left (2/1 and 2/0).
- Slot 3 ports are numbered from right to left (3/0 and 3/1).

When you configure an interface, identify the interface name before the slot and port numbers. For example, if you install a serial T1 VWIC interface in Slot 2, port 0 would be labeled as serial 2/0.

Figure 2-1 shows the Catalyst 4224 front panel.

Figure 2-1 Catalyst 4224 Front Panel



Using the Cisco IOS CLI

Cisco voice gateways run versions of Cisco IOS software that includes specialized adaptations for Voice over IP (VoIP) and Media Gateway Control Protocol (MGCP). If you are familiar with other versions of Cisco IOS, you will find configuring Cisco voice gateways straightforward because you will use the Cisco IOS CLI, with which you are familiar.

If you have never used the Cisco IOS CLI, you should still be able to perform the configuration required using the instructions and examples provided in this guide. To help get you started, this section provides a brief overview of some of the main features of the CLI. For further information, refer to the Cisco IOS configuration guides and command references for details about specific commands.

This section contains the following topics:

Getting Help

Use the question mark (?) and arrow keys to help you enter commands, as follows:

- For a list of available commands, enter a question mark, for example:

  ```
  Gateway> ?
  ```

- To complete a command, enter a few known characters followed by a question mark (with no space), for example:

  ```
  Gateway> s?
  ```

- For a list of command variables, enter the command followed by a space and a question mark, for example:

  ```
  Gateway> show ?
  ```

- To redisplay a command you previously entered, press the Up Arrow key. You can continue to press the Up Arrow key for more commands.

Command Modes

The Cisco IOS interface is divided into different modes. Each command mode permits you to configure different components on your gateway. The commands available at any given time depend on which mode you are currently using. Entering a question mark (**?**) at the prompt displays a list of commands available for each command mode. Table 2-2 lists the most common command modes.

Table 2-2 Common Command Modes

Command Mode	Access Method	Gateway Prompt Displayed	Exit Method
User EXEC	Log in.	`hostname>` The default is `router>`	Use the **logout** command.
Privileged EXEC	From user EXEC mode, enter the **enable** command.	`hostname#` The default is `router#`	To exit to user EXEC mode, use the **disable**, **exit**, or **logout** command.

Table 2-2 Common Command Modes (continued)

Command Mode	Access Method	Gateway Prompt Displayed	Exit Method

Global configuration	From the privileged EXEC mode, enter the **configure terminal** command.	`hostname (config)#`
		The default is
		`router (config)#`
			To exit to privileged EXEC mode, use the **exit** or **end** command, or press **Ctrl-Z**.

Interface configuration	From the global configuration mode, enter the **interface** *type number* command, such as **FastEthernet int 0/0**.	`hostname (config-if)#`
		The default is
		`router (config-if)#`
			To exit to global configuration mode, use the **exit** command.
			To exit directly to privileged EXEC mode, press **Ctrl-Z**.

Dial-peer configuration	From the global configuration mode, enter the dial-peer voice command, such as **dial-peer voice 1 pots/voip.**	`hostname(config-dial-peer)`
		The default is
		`router (config-dial-peer)#`
			To exit to global configuration mode, use the **exit** command.
			To exit directly to privileged EXEC mode, press **Ctrl-Z**.

Timesaver Each command mode restricts you to a subset of commands. If you are having trouble entering a command, check the prompt and enter the question mark (**?**) for a list of available commands. You might be in the wrong command mode or using the wrong syntax.

In the following example, which uses the default prompt (router>), notice how the prompt changes after each command to indicate a new command mode:

```
router> enable
Password: <enable password>
router# configure terminal
router(config-if)# line 0
router(config-line)# controller t1 1/0
router(config-controller)# exit
router(config)# exit
router#
%SYS-5-CONFIG_I: Configured from console by console
```

The last message is normal and does not indicate an error. Press **Return** to return to the prompt.

Note You can press **Ctrl-Z** in any mode to return immediately to privileged EXEC mode (router#), instead of entering **exit**, which returns you to the previous mode.

Disabling a Command or Feature

If you want to undo a command you entered or disable a feature, enter the keyword **no** before most commands; for example, **no mgcp**.

Saving Configuration Changes

You need to enter the **copy running-config startup-config** command to save your configuration changes to nonvolatile random-access memory (NVRAM), so the changes are not lost if there is a system reload or power outage; for example:

```
router# copy running-config startup-config

Building configuration...
```

Note It might take a minute or two to save the configuration to NVRAM. After the configuration has been saved, the privileged EXEC mode prompt (`router#`) reappears.

Configuring Ethernet Switching

This section describes the Ethernet switching capabilities of the Catalyst 4224. These capabilities are designed to work as part of the Cisco IP Telephony solution.

This section also outlines how to configure Ethernet ports on the Catalyst 4224 to support IP phones in a branch office on your network.

This section contains the following topics:

Configuring the Catalyst 4224 for Cisco IP Telephony

The Catalyst 4224 has 24 10/100 switched Ethernet ports with integrated inline power and Quality of Service (QoS) features. These features allow you to extend Voice-over-IP (VoIP) networks to small branch offices.

As an access gateway switch, the Catalyst 4224 can be deployed as a component of a centralized call processing network using a centrally deployed Cisco CallManager. Instead of deploying and managing key systems or PBXs in small branch offices, applications are centrally located at the corporate headquarters or data center and are accessed via the IP WAN.

Default Switch Configuration

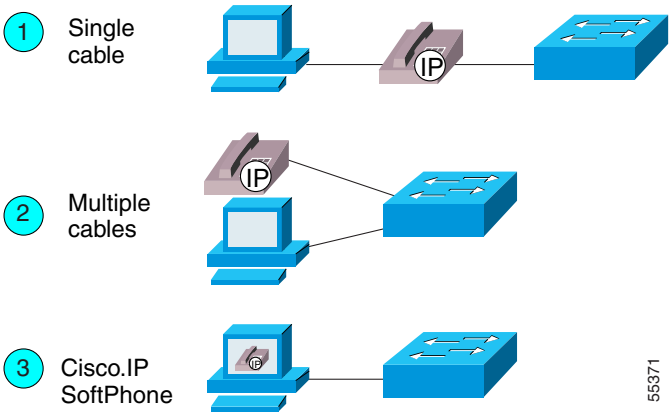
By default, the Catalyst 4224 provides the following settings with respect to Cisco IP Telephony:

- All switch ports are in access VLAN 1.
- All switch ports are static access ports, not 802.1Q trunk ports.
- Default voice VLAN is not configured on the switch.
- Inline power is automatically supplied on the 10/100 ports.

Connecting IP Phones to Your Campus Network

There are three ways to connect an IP phone to a campus network. You can use a single cable, multiple cables, or the Cisco IP SoftPhone application running on a PC. (See Figure 3-1.)

Figure 3-1 Ways to Connect IP Phones to the Network



For more information about Option 1, see the "Configuring Ethernet Ports to Support IP Phones and a Daisy-Chained Workstation" section on page 3-3.

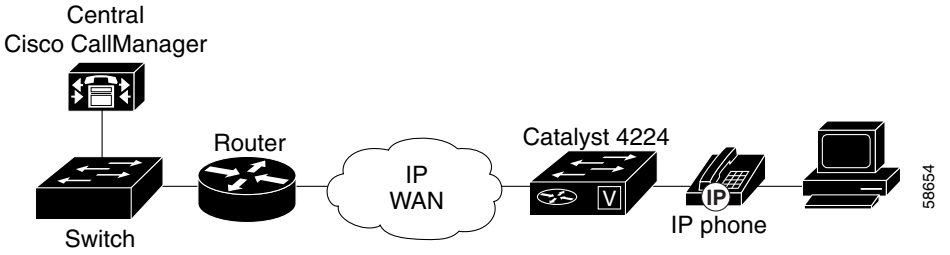
For more information about Option 2, see the "Configuring Ethernet Ports to Support IP Phones with Multiple Ports" section on page 3-9.

For more information about Option 3, which entails the Cisco IP SoftPhone application, see the Cisco IP SoftPhone documentation library. The Cisco IP SoftPhone application was developed to provide clients with a phone that runs on software. This application can be installed on any PC that connects to an IP telephony network.

Configuring Ethernet Ports to Support IP Phones and a Daisy-Chained Workstation

Figure 3-2 shows the topology of a centralized Cisco CallManager deployment model used to enable converged networks.

Figure 3-2 Catalyst 4224 with IP Phone and Workstation



The configurations described in this section use the model shown in Figure 3-2. In this model, voice traffic is given a higher priority (CoS=5) than data traffic (CoS=0). Hence, voice traffic is placed in a high-priority queue that gets serviced first, and data traffic is placed in a low-priority queue that gets serviced later.

This section describes the following configuration schemes:

- Configuring Separate Voice and Data Subnets, page 3-4
- Configuring a Single Subnet for Voice and Data, page 3-7

For details on the commands used in the following configuration examples, refer to Appendix A, "Command Reference for Voice VLAN."

Note In the following configurations, the **powerinline** command is set to **auto** by default.

Configuring Separate Voice and Data Subnets

For ease of network administration and increased scalability, network managers can configure the Catalyst 4224 to support Cisco IP phones such that the voice and data traffic reside on separate subnets. You should always use separate VLANs when you are able to segment the existing IP address space of your branch office.

User priority bits in the 802.1p portion of the 802.1Q standard header are used to provide prioritization in Ethernet switches. This is a vital component in designing Cisco IP Telephony networks.

The Catalyst 4224 provides the performance and intelligent services of Cisco IOS software for branch office applications. The Catalyst 4224 can identify user applications—such as voice or multicast video—and classify traffic with the appropriate priority levels. QoS policies are enforced using Layer 2 and 3 information such as 802.1p, IP precedence, and DSCP.

Note Refer to the *Cisco AVVID QoS Design Guide* for more information on how to implement end-to-end QoS as you deploy Cisco IP Telephony solutions.

The following exit procedure shows how to automatically configure Cisco IP phones to send voice traffic on the voice VLAN ID (VVID). (See the "Voice Traffic and VVID" section on page 3-5.)

	Task	Command

Step 1	Enable VLAN database.
	
	ID range is 1 to 1005.
		`enable`
		`vlan database`
		`vlan id`
		`exit`

Step 2	Set up switch port to configure IP phone on voice VLAN (on per-port basis).	

	Task	Command
	Enter the privileged EXEC mode. A preset password may be required to enter this mode.	**enable**
	Enter global configuration mode.	**configure terminal**
	Enter the interface configuration mode and the port to be configured (for example, interface fa5/1).	**interface** *interface*
	Configure the port as **access** and assign a data VLAN.	**switchport access vlan** *vlan-id*
	Configure the voice port with a VVID that will be used exclusively for voice traffic.	**switchport voice vlan** *vlan-id*
Step 3	Verify the switch port configuration and save it.	
	Verify the port configuration you just entered.	**show run interface** *interface*
	Save the current configuration in Flash memory.	**write memory**

Voice Traffic and VVID

The Catalyst 4224 can automatically configure voice VLAN. With the automatically configured voice VLAN feature, network administrators can segment phones into separate logical networks even though the data and voice infrastructure is physically the same. The voice VLAN feature places the phones into their own VLANs without the need for end-user intervention. A user can plug the phone into the switch, and the switch provides the phone with the necessary VLAN information.

Sample Configuration 1

The following example shows how to configure separate subnets for voice and data on the Catalyst 4224:

```
interface FastEthernet5/1
     description DOT1Q port to IP Phone
     switchport access vlan 50
     switchport voice vlan 150
     spanning-tree portfast (See Note below)

interface Vlan 150
     description voice vlan
     ip address 10.150.1.1 255.255.255.0
     ip helper-address 172.20.73.14 (See Note below)

interface Vlan 50
     description data vlan
     ip address 10.50.1.1 255.255.255.0
```

This configuration instructs the IP phone to generate a packet with an 802.1Q VLAN ID of 150 with an 802.1p value of 5 (default for voice bearer traffic).

Note The portfast command is only supported on nontrunk ports.

Note In a centralized Cisco CallManager deployment model, the DHCP server might be located across the WAN link. If so, an **ip helper-address** command pointing to the DHCP server should be included on the voice VLAN interface for the IP phone. This is done to obtain its IP address as well as the address of the TFTP server required for its configuration.

Be aware that Cisco IOS supports a DHCP server function. If this function is used, the Catalyst 4224 serves as a local DHCP server and a helper address would not be required.

Sample Configuration 2

Configuring inter-VLAN routing is identical to the configuration on a Catalyst 6000 with an MSFC. Configuring an interface for WAN routing is consistent with other Cisco IOS platforms.

The following example provides a sample configuration:

```
interface Vlan 160
     description voice vlan
     ip address 10.6.1.1 255.255.255.0

interface Vlan 60
     description data vlan
     ip address 10.60.1.1 255.255.255.0

interface Serial1/0
     ip address 160.3.1.2 255.255.255.0
```

Note Standard IGP routing protocols such as RIP, Interior Gateway Routing Protocol (IGRP), Enhanced Interior Gateway Routing Protocol (EIGRP), and open shortest path first (OSPF) are supported on the Catalyst 4224. Multicast routing is also supported for PIM dense mode, sparse mode, and sparse-dense mode.

Configuring a Single Subnet for Voice and Data

For network designs with incremental IP telephony deployment, network managers can configure the Catalyst 4224 so that the voice and data traffic coexist on the same subnet. This might be necessary when it is impractical to allocate an additional IP subnet for IP phones. You must still prioritize voice above data at both Layer 2 and Layer 3.

Layer 3 classification is already handled because the phone sets the type of service (ToS) bits in all media streams to an IP Precedence value of 5. (With Cisco CallManager Release 3.0(5), this marking changed to a Differentiated Services Code Point [DSCP] value of EF.) However, to ensure that there is Layer 2 classification for admission to the multiple queues in the branch office switches, the phone must also use the User Priority bits in the Layer 2 802.1p header to provide class of service (CoS) marking. Setting the bits to provide marking can be done by having the switch look for 802.1p headers on the native VLAN.

This configuration approach must address two key considerations:

- Network managers should ensure that existing subnets have enough available IP addresses for the new Cisco IP phones, each of which requires a unique IP address.
- Administering a network with a mix of IP phones and workstations on the same subnet might pose a challenge.

The following procedure shows how to automatically configure Cisco IP phones to send voice and data traffic on the same VLAN.

	Task	Command
Step 1	Set up switch port to configure IP phone on the same VLAN as the access VLAN.	
	Enter global configuration mode.	**configure terminal**
	Enter the interface configuration mode and the port to be configured (for example, **interface fa5/1**)	**interface** *interface*
	Set the native VLAN for untagged traffic. *vlan-id* represents the ID of the VLAN that is sending and receiving untagged traffic on the port. Valid IDs are from 1 to 1001. Leading zeroes are not accepted.	**switchport access vlan** *vlan-id*
	Configure the Cisco IP Phone to send voice traffic with higher priority (CoS=5 on 802.1Q tag) on the access VLAN. Data traffic (from an attached PC) is sent untagged for lower priority (port default=0).	**switchport voice vlan dot1p**
	Return to the privileged EXEC mode.	**end**
Step 2	Verify the switch port configuration and save.	
	Verify the port configuration you just entered.	**show run interface** *interface*
	Save the current configuration in Flash memory.	**write memory**

Sample Configuration

The Catalyst 4224 supports the use of an 802.1p-only option when configuring the voice VLAN. Use this option to allow the IP phone to tag VoIP packets with a CoS of 5 on the native VLAN, while all PC data traffic is sent untagged.

The following example shows a single subnet configuration for the Catalyst 4224 switch:

```
interface FastEthernet5/2
description Port to IP Phone in single subnet
     switchport access vlan 40
     switchport voice vlan dot1p
     spanning-tree portfast
```

The Catalyst 4224 instructs the IP phone to generate an 802.1Q frame with a null VLAN ID value but with an 802.1p value (default is CoS of 5 for bearer traffic). The voice and data vlans are both 40 in this example.

Configuring Ethernet Ports to Support IP Phones with Multiple Ports

You might want to use multiple ports to connect the IP phones (option 2 in Figure 3-1) if any of the following conditions apply to your Cisco IP telephony network:

- You are connecting IP phones that do not have a second Ethernet port for attaching a PC.
- You want to create a physical separation between the voice and data networks.
- You want to provide in-line power easily to the IP phones without having to upgrade the data infrastructure.
- You want to limit the number of switches that need UPS power.

IP Addressing

The recommended configuration for using multiple cables to connect IP phones to the network is to use a separate IP subnet and separate VLANs for IP telephony.

Sample Configuration

The following example illustrates the configuration on the IP phone:

```
interface FastEthernetx/x
     switchport voice vlan x
```

The following example illustrates the configuration on the PC:

```
interface FastEthernetx/y
     switchport access vlan y
```

Note Using a separate subnet, and possibly a separate IP address space, may not be an option for some small branch offices due to the IP routing configuration. If the IP routing can handle an additional subnet at the remote branch, you can use Cisco Network Registrar and secondary addressing.

Managing the Catalyst 4224 Access Gateway Switch

This section illustrates how to perform basic management tasks on the Catalyst 4224 with the Cisco IOS command-line interface (CLI). You might find this information useful when you configure the switch for the previous scenarios.

Note For reference information on the voice commands used in this section, refer to the Appendix A, "Command Reference for Voice VLAN."

This section contains the following topics:

Adding Trap Managers

A trap manager is a management station that receives and processes traps. When you configure a trap manager, community strings for each member switch must be unique. If a member switch has an assigned IP address, the management station accesses the switch by using its assigned IP address.

By default, no trap manager is defined, and no traps are issued.

Beginning in privileged EXEC mode, follow these steps to add a trap manager and community string:

	Task	Command
Step 1	Enter global configuration mode.	`config terminal`
Step 2	Enter the trap manager IP address, community string, and the traps to generate.	`snmp-server host 172.2.128.263 traps1 snmp vlan-membership`
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify that the information was entered correctly by displaying the running configuration.	`show running-config`

Configuring IP Information

This section describes how to assign IP information on the Catalyst 4224, and contains the following topics:

Assigning IP Information to the Switch—Overview

You can use a BOOTP server to automatically assign IP information to the switch; however, the BOOTP server must be set up in advance with a database of physical MAC addresses and corresponding IP addresses, subnet masks, and default

gateway addresses. In addition, the switch must be able to access the BOOTP server through one of its ports. At startup, a switch without an IP address requests the information from the BOOTP server; the requested information is saved in the switch running the configuration file. To ensure that the IP information is saved when the switch is restarted, save the configuration by entering the **write memory** command in privileged EXEC mode.

You can change the information in these fields. The mask identifies the bits that denote the network number in the IP address. When you use the mask to create a subnet on a network, the mask is then referred to as a subnet mask. The broadcast address is reserved for sending messages to all hosts. The CPU sends traffic to an unknown IP address through the default gateway.

Assigning IP Information to the Switch—Procedure

Beginning in privileged EXEC mode, follow these steps to enter the IP information:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter interface configuration mode, and enter the VLAN to which the IP information is assigned. VLAN 1 is the management VLAN, but you can configure any VLAN from IDs 1 to 1001.	`interface vlan 1`
Step 3	Enter the IP address and subnet mask.	`ip address` *ip_address subnet_mask*
Step 4	Return to global configuration mode.	`exit`
Step 5	Enter the IP address of the default router.	`ip default-gateway` *ip_address*
Step 6	Return to privileged EXEC mode.	`end`
Step 7	Verify that the information was entered correctly by displaying the running configuration. If the information is incorrect, repeat the procedure.	`show running-config`

Removing an IP Address

Use the following procedure to remove IP information from a switch.

Note Using the **no ip address** command in configuration mode disables the IP protocol stack and removes the IP information. Cluster members without IP addresses rely on the IP protocol stack being enabled.

Beginning in privileged EXEC mode, follow these steps to remove an IP address:

	Task	Command
Step 1	Enter interface configuration mode, and enter the VLAN to which the IP information is assigned. VLAN 1 is the management VLAN, but you can configure any VLAN from IDs 1 to 1001.	`interface vlan 1`
Step 2	Remove the IP address and subnet mask.	`no ip address`
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify that the information was removed by displaying the running configuration.	`show running-config`

Caution If you are removing the IP address through a Telnet session, your connection to the switch will be lost.

Specifying a Domain Name and Configuring the DNS

Each unique IP address can have an associated host name. Cisco IOS software maintains a cache of host name-to-address mappings for use by the EXEC mode commands **connect**, **telnet**, **ping**, and related Telnet support operations. This cache speeds the process of converting names to addresses.

IP defines a hierarchical naming scheme that allows a device to be identified by its location or domain. Domain names are pieced together with periods (.) as the delimiting characters. For example, Cisco Systems is a commercial organization that IP identifies by a *com* domain name, so its domain name is *cisco.com*. A specific device in this domain, such as the File Transfer Protocol (FTP) system, is identified as *ftp.cisco.com*.

To track domain names, IP has defined the concept of a domain name server (DNS), whose purpose is to hold a cache (or database) of names mapped to IP addresses. To map domain names to IP addresses, you must first identify the host names and then specify a name server and enable the DNS, the Internet's global naming scheme that uniquely identifies network devices.

Specifying the Domain Name

You can specify a default domain name that the software uses to complete domain name requests. You can specify either a single domain name or a list of domain names. When you specify a domain name, any IP host name without a domain name will have that domain name appended to it before being added to the host table.

Specifying a Name Server

You can specify up to six hosts that can function as a name server to supply name information for the DNS.

Enabling the DNS

If your network devices require connectivity with devices in networks for which you do not control name assignment, you can assign device names that uniquely identify your devices within the entire internetwork. The Internet's global naming scheme, the DNS, accomplishes this task. This service is enabled by default.

Configuring Voice Ports

The Catalyst 4224 can connect to a Cisco 7960 IP Phone and carry IP voice traffic. If necessary, the Catalyst 4224 can supply electrical power to the circuit connecting it to the Cisco 7960 IP Phone.

Because the sound quality of an IP telephone call can deteriorate if the data is unevenly transmitted, the current release of the Cisco IOS software supports Quality of Service (QoS) based on IEEE 802.1p Class of Service (CoS). QoS uses classification and scheduling to transmit network traffic from the switch in a predictable manner.

The Cisco 7960 IP Phone contains an integrated three-port 10/100 switch. These dedicated ports connect to the following devices:

- Port 1 connects to the Catalyst 4224 switch or other Voice-over-IP device.
- Port 2 is an internal 10/100 interface that carries the phone traffic.
- Port 3 connects to a PC or other device.

Figure 3-2 on page 3-3 shows a sample configuration for a Cisco 7960 IP Phone.

Configuring a Port to Connect to a Cisco 7960 IP Phone

Because a Cisco 7960 IP Phone also supports connection to a PC or other device, a port connecting a Catalyst 4224 to a Cisco 7960 IP Phone can carry a mix of traffic. There are three ways to configure a port connected to a Cisco 7960 IP Phone:

- All traffic is transmitted according to the default CoS priority (0) of the port. This is the default.
- Voice traffic is given a higher priority by the phone, and all traffic is in the same VLAN.
- Voice and data traffic are carried on separate VLANs, and voice traffic always has a CoS priority of five.

Disabling Inline Power on a Catalyst 4224

The Catalyst 4224 can supply inline power to the Cisco 7960 IP Phone if necessary. The Cisco 7960 IP Phone can also be connected to an AC power source and supply its own power to the voice circuit. When the Cisco 7960 IP Phone is supplying its own power, a Catalyst 4224 can forward IP voice traffic to and from the phone.

A detection mechanism on the Catalyst 4224 determines whether it is connected to a Cisco 7960 IP Phone. If the switch senses that there is no power on the circuit, the switch supplies the power. If there is power on the circuit, the switch does not supply it.

You can configure the switch to never supply power to the Cisco 7960 IP Phone and to disable the detection mechanism.

Beginning in privileged EXEC mode, follow these steps to configure a port to never supply power to Cisco 7960 IP Phones:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter interface configuration mode, and enter the port to be configured.	`interface` *interface*
Step 3	Permanently disable inline power on the port.	`power inline never`
Step 4	Return to privileged EXEC mode.	`end`
Step 5	Verify the change by displaying the setting as configured.	`show power inline` *interface* `configured`

Note Entering the **show power inline** [*interface-type number*] command in privileged EXEC mode displays the power allocated to the IP phone by the Catalyst 4224. To display the maximum power requested by the IP phone, enter the **show cdp neighbors** [*interface-type number*] **detail** command in privileged EXEC mode.

Enabling and Disabling Switch Port Analyzer

You can monitor traffic on a given port by forwarding incoming and outgoing traffic on the port to another port in the same VLAN. A Switch Port Analyzer (SPAN) port cannot monitor ports in a different VLAN, and a SPAN port must be a static-access port. Any number of ports can be defined as SPAN ports, and any combination of ports can be monitored. SPAN is supported for up to two sessions.

Enabling the Switch Port Analyzer

Beginning in privileged EXEC mode, follow these steps to enable SPAN:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enable port monitoring for a specific session (“*number”).* Optionally, supply a SPAN *destination* interface, and a *source* interface	`monitor session` *number destination source*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entries.	`show running-config`

Disabling Switch Port Analyzer

Beginning in privileged EXEC mode, follow these steps to disable SPAN:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Disable port monitoring for a specific session.	`no monitor session` *number*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entries.	`show running-config`

Managing the ARP Table

To communicate with a device (on Ethernet, for example), the software first must determine the 48-bit MAC or local data link address of that device. The process of determining the local data link address from an IP address is called *address resolution*.

The Address Resolution Protocol (ARP) associates a host IP address with corresponding media or MAC addresses and VLAN ID. Taking an IP address as input, ARP determines the associated MAC address. Once a MAC address is determined, the IP-MAC address association is stored in an ARP cache for rapid retrieval. Then, the IP datagram is encapsulated in a link-layer frame and sent over the network.

Encapsulation of IP datagrams and ARP requests and replies on IEEE 802 networks other than Ethernet is specified by the Subnetwork Access Protocol (SNAP). By default, standard Ethernet-style ARP encapsulation (represented by the **arpa** keyword) is enabled on the IP interface.

When you manually add entries to the ARP Table by using the CLI, you must be aware that these entries do not age and must be manually removed.

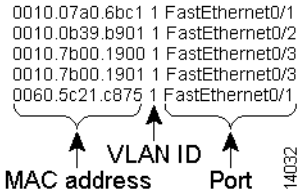
Managing the MAC Address Tables

The switch uses the MAC address tables to forward traffic between ports. All MAC addresses in the address tables are associated with one or more ports. These MAC tables include the following types of addresses:

- Dynamic address—A source MAC address that the switch learns and then drops when it is not in use.
- Secure address—A manually entered unicast address that is usually associated with a secured port. Secure addresses do not age.
- Static address—A manually entered unicast or multicast address that does not age and that is not lost when the switch resets.

The address tables list the destination MAC address and the associated VLAN ID, module, and port number associated with the address. Figure 3-3 shows an example of a list of addresses as they would appear in the dynamic, secure, or static address table.

Figure 3-3 Contents of the Address Table



MAC Addresses and VLANs

All MAC addresses are associated with one or more VLANs. An address can exist in more than one VLAN and have different destinations in each. Multicast addresses, for example, could be forwarded to port 1 in VLAN 1 and ports 9, 10, and 11 in VLAN 5.

Each VLAN maintains its own logical address table. A known address in one VLAN is unknown in another until it is learned or statically associated with a port in the other VLAN. An address can be secure in one VLAN and dynamic in another. Addresses that are statically entered in one VLAN must be static addresses in all other VLANs.

Changing the Address Aging Time

Dynamic addresses are source MAC addresses that the switch learns and then drops when they are not in use. Use the Aging Time field to define how long the switch retains unseen addresses in the table. This parameter applies to all VLANs.

Configuring the Aging Time

Setting too short an aging time can cause addresses to be prematurely removed from the table. When the switch receives a packet for an unknown destination, the switch floods the packet to all ports in the same VLAN as the receiving port. This unnecessary flooding can impact performance. Setting too long an aging time can cause the address table to be filled with unused addresses; it can cause delays in establishing connectivity when a workstation is moved to a new port.

Beginning in privileged EXEC mode, follow these steps to configure the dynamic address table aging time.

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter the number of seconds that dynamic addresses are to be retained in the address table. You can enter a number from 10 to 1000000.	`mac-address-table aging-time` *seconds*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entry.	`show mac-address-table aging-time`

Removing Dynamic Address Entries

Beginning in privileged EXEC mode, follow these steps to remove a dynamic address entry:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter the MAC address to be removed from dynamic MAC address table.	`no mac-address-table dynamic` *hw-addr*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entry.	`show mac-address-table`

You can remove all dynamic entries by using the **clear mac-address-table dynamic** command in privileged EXEC mode.

Adding Secure Addresses

The secure address table contains secure MAC addresses and their associated ports and VLANs. A secure address is a manually entered unicast address that is forwarded to only one port per VLAN. If you enter an address that is already assigned to another port, the switch reassigns the secure address to the new port.

You can enter a secure port address even when the port does not yet belong to a VLAN. When the port is later assigned to a VLAN, packets destined for that address are forwarded to the port.

Adding Secure Addresses

Beginning in privileged EXEC mode, follow these steps to add a secure address:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter the MAC address, its associated port, and the VLAN ID.	**mac-address-table secure** *hw-addr interface* **vlan** *vlan-id*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entry.	`show mac-address-table secure`

Removing Secure Addresses

Beginning in privileged EXEC mode, follow these steps to remove a secure address:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter the secure MAC address, its associated port, and the VLAN ID to be removed.	**no mac-address-table secure** *hw-addr* **vlan** *vlan-id*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entry.	`show mac-address-table secure`

You can remove all secure addresses by using the **clear mac-address-table secure** command in privileged EXEC mode.

Adding and Removing Static Addresses

A static address has the following characteristics:

- It is manually entered in the address table and must be manually removed.
- It can be a unicast or multicast address.
- It does not age and is retained when the switch restarts.

Because all ports are associated with at least one VLAN, the switch acquires the VLAN ID for the address from the ports that you select on the forwarding map. A static address in one VLAN must be a static address in other VLANs. A packet with a static address that arrives on a VLAN where it has not been statically entered is flooded to all ports and not learned.

Adding Static Addresses

Beginning in privileged EXEC mode, follow these steps to add a static address:

	Task	Command
Step 1	Enter global configuration mode.	**configure terminal**
Step 2	Enter the static MAC address, the interface, and the VLAN ID of those ports.	**mac-address-table static** *hw-addr* **[interface]** *interface* **[vlan]** *vlan-id*
Step 3	Return to privileged EXEC mode.	**end**
Step 4	Verify your entry.	**show mac-address-table static**

Removing Static Addresses

Beginning in privileged EXEC mode, follow these steps to remove a static address:

	Task	Command
Step 1	Enter global configuration mode.	`configure terminal`
Step 2	Enter the static MAC address, the interface, and the VLAN ID of the port to be removed	`no mac-address-table static` *hw-addr* `[interface]` *interface* `[vlan]` *vlan-id*
Step 3	Return to privileged EXEC mode.	`end`
Step 4	Verify your entry.	`show mac-address-table static`

You can remove all secure addresses by using the **clear mac-address-table static** command in privileged EXEC mode.

Configuring the Data Interfaces

This section describes how to configure the data interfaces on the Catalyst 4224 To configure a data interface, you must use configuration mode. In this mode, you enter Cisco IOS command-line interface (CLI) commands at the gateway prompt.

This section contains the following topics:

This section describes some of the most commonly used configuration procedures. For advanced configuration topics, refer to the Cisco IOS configuration guide and command reference publications. These publications are available on the Documentation CD-ROM or on Cisco.com. You can also order printed copies separately.

Configuring the Host Name and Password

One of your first configuration tasks is to configure the host name and set an encrypted password. Configuring a host name allows you to distinguish multiple Catalyst 4224s. Setting an encrypted password allows you to prevent unauthorized configuration changes.

To configure the host name and password, perform these tasks:

	Task	Command

Step 1	Enter enable mode.
	Enter the password.
	You know you have entered enable mode when the prompt changes to `gateway#`.
		`gateway> enable`
		`Password: <password>`
		`gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway# configure terminal`
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Provide the Catalyst 4224 a meaningful name. Substitute your host name for gwy1.	`gateway(config)# hostname gwy1` `gwy1(config)#`
Step 4	Substitute your enable secret password for `guessme`. This password gives you access to privileged EXEC mode. When you type **enable** at the EXEC prompt (`gateway>`), you must enter the enable secret password to gain access to configuration mode.	`gwy1(config)# enable secret guessme`
Step 5	Enter line configuration mode to configure the console port. When you enter line configuration mode, the prompt changes to `gwy1(config-line)#`.	`gwy1(config)# line con 0` `gwy1(config-line)#`

	Task	Command
Step 6	Enter **exec-timeout 0 0** to prevent the Catalyst 4224's EXEC facility from timing out if you do not type any information on the console screen for an extended period.	`gwy1(config-line)# exec-timeout 0 0`
Step 7	Exit to global configuration mode.	`gwy1(config-line)# exit` `gwy1(config)#`

To verify that you configured the correct host name and password, follow these steps:

Step 1 Enter the **show config** command:

```
gwy1# show config

Using 1888 out of 126968 bytes
!
version XX.X
.
.
.
!
hostname gwy1
!
enable secret 5 $1$60L4$X2JYOwoDc0.kqa1loO/w8/
.
.
.
```

Check the host name and encrypted password displayed near the top of the command output.

Step 2 Exit global configuration mode and attempt to reenter it using the new enable password:

```
gwy1# exit
.
.
.
gwy1 con0 is now available
Press RETURN to get started.

gwy1> enable
Password: guessme
gwy1#
```

Tip If you are having trouble, ensure that the Caps Lock function is off; passwords are case sensitive.

Configuring the Fast Ethernet Interface

This section describes how to configure the Fast Ethernet interface on the Catalyst 4224.

Timesaver Before you begin, disconnect all WAN cables from the Catalyst 4224 to prevent it from running the AutoInstall process. The Catalyst 4224 attempts to run AutoInstall if there is a WAN connection on both ends and the Catalyst 4224 does not have a valid configuration file stored in nonvolatile random-access memory (NVRAM). The Catalyst 4224 can take several minutes to determine that AutoInstall is not connected to a remote TCP/IP host.

To configure the Fast Ethernet interface, follow these steps:

	Task	Command

Step 1	Enter enable mode.
	Enter the password.
	You know you have entered enable mode when the prompt changes to `gateway#`.
		`gateway>` **`enable`**
		`Password: <password>`
		`gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway#` **`configure terminal`**
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Enable routing protocols as required for your global configuration.	`gateway(config)#` **`ip routing`**
Step 4	Enter interface configuration mode. You know you have entered interface configuration mode when the prompt changes to `gateway(config-if)#`.	`gateway(config)#` **`interface fastethernet 0/0`** `gateway(config-if)#`
Step 5	Assign an IP address and subnet mask to the interface.	`gateway(config-if)#` **`ip address 172.16.74.3 255.255.255.0`**
Step 6	Exit to global configuration mode. If your Catalyst 4224 has more than one Fast Ethernet interface that you need to configure, repeat Step 4 through Step 6.	`gateway(config-if)#` **`exit`**
Step 7	When you finish configuring interfaces, return to enable mode. You know you have entered enable mode when the prompt changes to `gateway#`.	`gateway(config)#` **`Ctrl-Z`** `gateway#`

Configuring Asynchronous/Synchronous Serial Interfaces

This section describes how to configure the serial interfaces on your asynchronous/synchronous serial WIC.

Note The asynchronous/synchronous serial WIC supports synchronous mode only. At this time, asynchronous mode is not supported.

Timesaver Before you begin, disconnect all WAN cables from the Catalyst 4224 to keep it from running the AutoInstall process. The Catalyst 4224 attempts to run AutoInstall if there is a WAN connection on both ends and the Catalyst 4224 does not have a valid configuration file stored in NVRAM. The Catalyst 4224 can take several minutes to determine that AutoInstall is not connected to a remote TCP/IP host.

To configure the serial interfaces, perform these steps:

	Task	Command

Step 1	Enter enable mode.
	Enter the password.
	You know you have entered enable mode when the prompt changes to `gateway#`.
		`gateway> enable`
		`Password: <password>`
		`gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway# configure terminal`
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Enable routing protocols as required for your global configuration.	`gateway(config)# ip routing`

	Task	Command
Step 4	Enter the interface configuration mode. You know you have entered interface configuration mode when the prompt changes to `gateway(config-if)#`.	`gateway(config)# interface serial 1/0` `gateway(config-if)#`
Step 5	Assign the IP address and subnet mask to the interface. **Note** All serial ports are initially configured as synchronous.	`gateway(config-if)# ip address 172.16.74.1 255.255.255.0`
Step 6	To use a port in Data Communication Equipment (DCE) mode, connect a DCE cable and set the internal transmit clock signal (TXC) speed in bits per second. (For ports used in Data Terminal Equipment (DTE) mode, the Catalyst 4224 automatically uses the external timing signal.)	`gateway(config-if)# clock rate 7200`
Step 7	When a port is operating in DCE mode, the default operation is for the DCE to send serial clock transmit (SCT) and serial clock receive (SCR) clock signals to the DTE, and for the DTE to return an serial clock transmit external (SCTE) signal to the DCE. If the DTE does not return an SCTE signal, enter this command to configure the DCE port to use its own clock signal.	`gateway(config-if)# dce-terminal-timing-enable`
Step 8	A Catalyst 4224 that uses long cables might experience high error rates when operating at higher transmission speeds, because the clock and data signals can shift out of phase. If a DCE port is reporting a high number of error packets, you can often correct the shift by inverting the clock using this command.	`gateway(config-if)# invert-txcclock`
Step 9	All serial interfaces support both nonreturn to zero (NRZ) and nonreturn to zero inverted (NRZI) formats. NRZ is the default; NRZI is commonly used with EIA/TIA-232 connections in IBM environments. To enable NRZI encoding on an interface, enter this command.	`gateway(config-if)# nrzi-encoding`

	Task	Command
Step 10	Exit back to global configuration mode. If your Catalyst 4224 has more that one serial interface that you need to configure, repeat Step 4 through Step 9.	`gateway(config-if)# exit`
Step 11	When you finish configuring the interface, return to enable mode.	`gateway(config)# Ctrl-z` `gateway#`

Table 4-1 lists the half-duplex timer commands.

Table 4-1 Half-duplex timer commands

Timer	Syntax	Default Setting (Milliseconds)
CTS delay[1]	`half-duplex timer cts-delay`	100
CTS drop timeout	`half-duplex timer cts-drop-timeout`	5000
DCD[2] drop delay	`half-duplex timer dcd-drop-delay`	100
DCD transmission start delay	`half-duplex timer dcd-txstart-delay`	100
RTS[3] drop delay	`half-duplex timer rts-drop-delay`	100
RTS timeout	`half-duplex timer rts-timeout`	2000
Transmit delay	`half-duplex timer transmit-delay`	0

1. CTS = Clear To Send.
2. DCD = Data Carrier Detect
3. RTS = Request To Send.

The following clock rate settings are for two-port asynchronous/synchronous serial WICs:

- 1200 bps
- 2400 bps
- 4800 bps
- 9600 bps

- 14400 bps
- 19200 bps
- 28800 bps
- 32000 bps
- 38400 bps
- 56000 bps
- 57600 bps
- 64000 bps
- 72000 bps
- 115200 bps
- 125000 bps
- 128000 bps

Configuring ISDN BRI Interfaces

This section describes how to configure the interfaces on the basic rate interface (BRI) card of your Catalyst 4224.

Note Before using a Catalyst 4224 with an ISDN BRI interface, you must order a correctly configured ISDN BRI line from your local telecommunications service provider. ISDN BRI provisioning refers to the types of services provided by the ISDN BRI line. Although provisioning is performed by your ISDN BRI service provider, you must tell the provider what you want.

Timesaver Before you begin, disconnect all WAN cables from the Catalyst 4224 to keep it from running the AutoInstall process. The Catalyst 4224 attempts to run AutoInstall if there is a WAN connection on both ends and the Catalyst 4224 does not have a valid configuration file stored in NVRAM. The Catalyst 4224 can take several minutes to determine that AutoInstall is not connected to a remote TCP/IP host.

To configure ISDN BRI interfaces, perform the following steps:

	Task	Command

Step 1	Enter enable mode.
	Enter the password.
	You know you have entered enable mode when the prompt changes to `gateway#`.
		`gateway> enable`
		`Password: <password>`
		`gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway# configure terminal`
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Enter an ISDN switch type. See Table 4-2 for a list of ISDN switch types. **Note** Switch types configured in interface configuration mode override this setting for the configured interface.	`gateway(config)# isdn switch-type switch-type`
Step 4	Enable routing protocols as required for your global configuration.	`gateway(config)# ip routing`
Step 5	Enter the interface configuration mode. You know you have entered interface configuration mode when the prompt changes to `gateway(config-if)#`.	`gateway(config)# interface bri 2/0` `gateway(config-if)#`
Step 6	Enable encapsulation. For data, you need to change the encapsulation to *hdlc*, *ppp*, or *frame-relay*. The default is *voice*.	`gateway(config)# encaps encaps-type`
Step 7	Assign the IP address and subnet mask to the interface. If you are configuring this interface for voice, enter the switch type instead of an IP address.	`gateway(config-if)# ip address 172.16.74.2 255.255.255.0` `gateway(config-if)# isdn switch-type basic-5ess`
Step 8	Exit back to global configuration mode. If your Catalyst 4224 has more than one BRI interface that you need to configure, repeat Step 5 through Step 7.	`gateway(config-if)# exit`

	Task	Command
Step 9	By default, the Catalyst 4224 allocates 25 percent of DRAM to shared memory (used for data transmitted (or received) by WAN interface cards). Specifying **memory-size iomem 40** increases shared memory from 25 percent to 40 percent.	gateway(config)# **memory-size iomem 40**
Step 10	When you finish configuring the interface, return to enable mode.	gateway(config)# **Ctrl-z** gateway#

Table 4-2 lists the supported ISDN switch types by country.

Table 4-2 ISDN Switch Types

Region	ISDN Switch Type	Description
Australia	basic-ts013	Australian TS013 switches
Europe	basic-1tr6	German 1TR6 ISDN switches
	basic-nwnet3	Norwegian NET3 ISDN switches (phase 1)
	basic-net3	NET3 ISDN switches (UK and others)
	vn2	French VN2 ISDN switches
	vn3	French VN3 ISDN switches
Japan	ntt	Japanese NTT ISDN switches
New Zealand	basic-nznet3	New Zealand NET3 switches
North America	basic-5ess	AT&T basic rate switches
	basic-dms100	NT DMS-100 basic rate switches
	basic-nil1	National ISDN-1 switches

Configuring T1 and E1 Interfaces

This section describes how to configure a T1/E1 multiflex trunk interface on your Catalyst 4224. It describes a basic configuration, including how to enable the interface and to specify IP routing. Depending on your own requirements and the protocols you plan to route, you might also need to enter other configuration commands.

Timesaver Before you begin, disconnect all WAN cables from the Catalyst 4224 to keep it from running the AutoInstall process. The Catalyst 4224 attempts to run AutoInstall if there is a WAN connection on both ends and the Catalyst 4224 does not have a valid configuration file stored in NVRAM. The Catalyst 4224 can take several minutes to determine that AutoInstall is not connected to a remote TCP/IP host.

Configuring T1 Interfaces

To configure a new T1, Channelized T1 (CT1)/PRI, or CT1/PRI-channel status unit (CSU) interface, or to change the configuration of an existing interface, perform these steps:

	Task	Command
Step 1	Enter enable mode. Enter the password. You know you have entered enable mode when the prompt changes to `gateway#`.	`gateway>` **`enable`** `Password:` **`<`***`password`***`>`** `gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway#` **`configure terminal`**
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Enable routing protocols as required for your global configuration.	`gateway(config)#` **`ip routing`**
Step 4	Select frame clock.	`gateway(config)#` **`frame-clock -select`**

	Task	Command
Step 5	Enter controller configuration mode for the CT1/PRI interface at the specified slot/port location. This example configures a T1 interface in slot 1 and unit 0.	`gateway(config)# controller t1 1/0`
Step 6	Specify which end of the circuit provides clocking. The clock source should be set to use internal clocking only for testing the network or if the full T1 line is used as the channel group. Only one end of the T1 line should be set to internal.	`gateway(config-controller)# clock source line`
Step 7	Specify the T1 framing type. The framing type defines the control bits and data bits. Cisco supports super frame (SF) and extended super frame (ESF) for T1s. SF is used in channel-bank robbed bit signalling (RBS) configurations. SF uses the framing bit to identify the channel and voice-related signaling within the frame. SF is not recommended for PRI configurations. ESF is required for 64 kb operation on DS0s. ESF requires 2k-framing bits for synchronization. The remaining 6k is used for error detection, CRC, and data link monitoring. ESF is recommended for PRI configurations. This example uses ESF.	`gateway(config-controller)# framing esf`
Step 8	Specify the line code format. This is an encoding method used to allow synchronous data to be transmitted in a compatible format for T1 transmission. Common line codes are RZ (return to zero), NRZ (non-return to zero), binary zero 0 substitution (B8ZS), alternate mark inversion (AMI), and HDB3 (high density bipolar order 3). The most popular line-code scheme used in North America is B8ZS. To maintain clock synchronization, B8ZS replaces a string of eight binary 0s with variations. B8ZS is more reliable than AMI, and it should be used with PRI configurations.	`gateway(config-controller)# linecode b8zs`

	Task	Command
Step 5	Enter controller configuration mode for the CT1/PRI interface at the specified slot/port location. This example configures a T1 interface in slot 1 and unit 0.	`gateway(config)# controller t1 1/0`
Step 6	Specify which end of the circuit provides clocking. The clock source should be set to use internal clocking only for testing the network or if the full T1 line is used as the channel group. Only one end of the T1 line should be set to internal.	`gateway(config-controller)# clock source line`
Step 7	Specify the T1 framing type. The framing type defines the control bits and data bits. Cisco supports super frame (SF) and extended super frame (ESF) for T1s. SF is used in channel-bank robbed bit signalling (RBS) configurations. SF uses the framing bit to identify the channel and voice-related signaling within the frame. SF is not recommended for PRI configurations. ESF is required for 64 kb operation on DS0s. ESF requires 2k-framing bits for synchronization. The remaining 6k is used for error detection, CRC, and data link monitoring. ESF is recommended for PRI configurations. This example uses ESF.	`gateway(config-controller)# framing esf`
Step 8	Specify the line code format. This is an encoding method used to allow synchronous data to be transmitted in a compatible format for T1 transmission. Common line codes are RZ (return to zero), NRZ (non-return to zero), binary zero 0 substitution (B8ZS), alternate mark inversion (AMI), and HDB3 (high density bipolar order 3). The most popular line-code scheme used in North America is B8ZS. To maintain clock synchronization, B8ZS replaces a string of eight binary 0s with variations. B8ZS is more reliable than AMI, and it should be used with PRI configurations.	`gateway(config-controller)# linecode b8zs`

	Task	Command
Step 9	Specify the channel group and time slots to be mapped. When configuring a T1 data line, channel-group numbers can be values from 0 to 23. Time slots are assigned to channels. One or more time slots or ranges of time slots belong to the channel group. The first time slot is numbered 1. For a T1 controller, the time slot range is from 1 to 24. For T1 PRI scenarios, all 24 T1 time slots are assigned as ISDN/PRI channels. The default line speed when configuring a T1 controller is 56 kbps. In this example, channel-group 0 consists of five time slots and runs at a speed of 56 kbps per time slot.	`gateway(config-controller)#` **`channel-group 0 timeslots 1,3-5,7`**
Step 10	Configure each channel group as a virtual serial interface. Specify the T1 interface (1), unit number (0), and channel group (0) to modify and enter the interface configuration mode. You know you have entered interface configuration mode when the prompt changes to `gateway(config-if)#`.	`gateway(config-controller)#` **`interface serial 1/0:0`** `gateway(config-if)#`
Step 11	Assign an IP address and subnet mask to the interface.	`gateway(config-if)#` **`ip address 10.1.15.1 255.255.255.0`**
Step 12	Exit back to global configuration mode. If your Catalyst 4224 has more than one CT1/PRI interface that you need to configure, repeat Steps 4 through 10.	`gateway(config-if)#` **`exit`**
Step 13	When you finish configuring interfaces, return to enable mode.	`gateway(config)#` **`Ctrl-z`** `gateway#`

Configuring E1 Interfaces

To configure a new E1 interface (balanced or unbalanced) or to change the configuration of an existing interface, perform these steps:

	Task	Command

Step 1	Enter enable mode.
	Enter the password.
	You know you have entered enable mode when the prompt changes to `gateway#`.
		`gateway> enable`
		`Password: <password>`
		`gateway#`

Step 2	Enter global configuration mode.
	You know you have entered global configuration mode when the prompt changes to `gateway(config)#`.
		`gateway# configure terminal`
		`Enter configuration commands, one per line. End with Ctrl-Z.`
		`gateway(config)#`

Step 3	Enable routing protocols as required for your global configuration.	`gateway(config)# ip routing`
Step 4	Select frame clock.	`gateway(config)# frame-clock-select`
Step 5	Enter controller configuration mode for the CE1/PRI interface at the specified slot/port location. This example configures a E1 interface in slot 1 and unit 0.	`gateway(config)# controller e1 1/0`
Step 6	Specify the framing type as cyclic redundancy check 4 (CRC4).	`gateway(config-controller)# framing crc4`
Step 7	Specify the line code format as high-density bipolar 3 (HDB3).	`gateway(config-controller)# linecode hdb3`

	Task	Command
Step 8	Specify the channel group and time slots to be mapped. When configuring a E1 data line, channel-group numbers can be values from 0 to 30. Time slots are assigned to channels. One or more time slots or ranges of time slots belong to the channel group. The first time slot is numbered 1. For an E1 controller, the time slot range is from 1 to 31. For E1 PRI scenarios, all 31 T1 time slots are assigned as ISDNPRI channels. The default line speed when configuring an E1 controller is 64 kbps. In this example, channel-group 0 consists of five time slots and runs at a speed of 64 kbps per time slot.	`gateway(config-controller)#` **`channel-group 0 timeslots 1,3-5,7`**
Step 9	Configure each channel group as a virtual serial interface. Specify the E1 interface, unit number, and channel group to modify and enter the interface configuration mode. You know you have entered interface configuration mode when the prompt changes to `gateway(config-if)#`.	`gateway(config-controller)#` **`interface serial 1/0:0`** `gateway(config-if)#`
Step 10	Assign an IP address and subnet mask to the interface.	`gateway(config-if)#` **`ip address 10.1.15.1 255.255.255.0`**
Step 11	Exit back to global configuration mode. If your Catalyst 4224 has more than one CE1/PRI interface that you need to configure, return to Step 4.	`gateway(config-if)#` **`exit`**
Step 12	When you finish configuring interfaces, return to enable mode.	`gateway(config)#` **`Ctrl-z`** `gateway#`

Checking the Interface Configuration

After configuring the new interface, you can perform the following tests to verify that the new interface is operating correctly:

- Display the Catalyst 4224 hardware configuration with the **show version** command. Check that the list includes the new interface.
- Specify an interface with the **show interfaces** [*type slot/port*] command and verify that the first line of the display shows the interface with the correct slot and port number, and that the interface and line protocol are in the correct state, up or down.
- Display the protocols configured for the entire Catalyst 4224 and for individual interfaces with the **show protocols** command. If necessary, return to configuration mode to add or remove protocol routing on the Catalyst 4224 or its interfaces.
- Display the running configuration with the **show running-config** command, and the configuration stored in NVRAM using the **show startup-config** command.
- Use the **ping** command to send an echo request to a specified IP address. Each returned signal is displayed as an exclamation point (!) on the console; each signal that is not returned before the timeout is displayed as a period (.). A series of exclamation points (!!!!!) indicates a good connection; a series of periods (.....) or the messages “timed out” or “failed” indicate that the connection failed.

If an interface is down and you configured it as up, or if the displays indicate that the hardware is not functioning properly, make sure that the new interface is properly connected and configured.

Saving Configuration Changes

To prevent the loss of the Catalyst 4224 configuration, you need to save it to NVRAM.

To save configuration changes, perform these steps:

	Task	Command
Step 1	Enter enable mode. Enter the password. You know you have entered enable mode when the prompt changes to `gateway#`.	`gateway>` **`enable`** `Password:` *`password`* `gateway#`
Step 2	Save the configuration changes to NVRAM so that they are not lost during resets, power cycles, or power outages.	`gateway#` **`copy running-config startup-config`** [or] `gateway#` **`write memory`**

Step 3	Return to enable mode.
	This message is normal and does not indicate an error.
		`gateway(config-if)#` **`Ctrl-z`**
		`gateway#`
		`%SYS-5-CONFIG_I: Configured from console by console`

Configuring the Voice Interfaces

This section explains how to configure Voice-over-IP (VoIP) routing on the supported interface cards of your Catalyst 4224 Access Gateway Switch. (You need to perform the software configuration only for the cards that you have installed.) To configure a voice interface, you must use configuration mode. In this mode, you enter Cisco IOS command-line interface (CLI) commands at the Gateway prompt.

This section includes the following topics:

Configuring Voice Interfaces

Use a voice interface card (VIC) for a voice connection. For information about installing these components in a Catalyst 4224, refer to the *Cisco Catalyst 4224 Access Gateway Switch Hardware Installation Guide*. For an explanation about how these components work and how they are identified, see the "First-Time Configuration" section on page 2-1.

When you start a Catalyst 4224, it automatically detects the voice network modules and VICs that have been installed. The first time you use a Catalyst 4224, you need to configure each voice port that you want to enable. If you replace a card after you configure each port, the gateway will recognize the new hardware component and use the previous configuration settings.

If you replace a module that was already configured, the gateway recognizes it and brings up the interface in the existing configuration.

To configure the Catalyst 4224 to boot in a new configuration, perform the following steps:

Step 1 Connect a terminal or a PC running terminal emulation software to the console port on the Catalyst 4224. Refer to the *Catalyst 4224 Access Gateway Switch Hardware Installation Guide* for instructions.

Step 2 Power on the Catalyst 4224. If the current configuration is no longer valid, you see the following prompt within about one minute:

```
Would you like to enter the initial dialog? [yes/no]:
```

Step 3 Enter **no**. You now enter the normal operating mode of the gateway.

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

Step 4 After a few seconds, you see the user EXEC prompt (`c4224>`). Enter **enable** and the password to enter privileged EXEC mode, as follows:

```
c4224> enable
Password: <password>
c4224# enable
```

Configuration changes can be made only in privileged EXEC mode. When you enter privileged EXEC mode, the prompt changes to the host name followed by a pound sign (#), such as `c4224#`.

Step 5 Enter the commands required to configure the VICs installed in the Catalyst 4224.

Step 6 When you finish configuring the voice interfaces, return to global configuration mode using the **exit** command and return to enable mode by pressing **Ctrl-Z**.

Step 7 To see the current operating configuration, including any changes you just made, enter the **show running-config** command:

```
c4224# show running-config
```

To see the configuration currently stored in non-volatile random access memory (NVRAM), enter the **show startup-config** command at the enable prompt:

```
c4224# show startup-config
```

Step 8 The results of the **show running-config** and **show startup-config** commands differ from each other if you have changed the configuration but have not yet written the changes to NVRAM. To write your changes to NVRAM and make them permanent, enter the **copy running-config startup-config** command at the enable prompt:

```
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

MGCP Configuration

If you want to use Media Gateway Control Protocol (MGCP), configuration of the Catalyst 4224 differs depending on whether you are using it with Cisco CallManager 3.0 or 3.1.

With Cisco CallManager 3.1 and later, you can create the MGCP gateway configuration on the Cisco CallManager server and download the configuration to the Catalyst 4224. For the details of this configuration procedure, refer to the Cisco CallManager 3.1 online help and to *Configuring Cisco IP Telephony Gateways*, available online at Cisco.com.

With Cisco CallManager 3.0, you must configure each voice port for MGCP on the Catalyst 4224 and then duplicate this configuration in Cisco CallManager Administration.

This section contains the following Catalyst 4224 configuration topics:

- Configuring T1-CAS E&M Emulation, page 5-8
- T1/E1 Configuration (H.323), page 5-31
- Where to Go Next, page 5-16

For more information on using MGCP with Cisco CallManager 3.0, refer to the Cisco CallManager 3.0 online help and to *Configuring Cisco IP Telephony Gateways*.

Enabling MGCP

To configure the Catalyst 4224 so that it can be controlled by Cisco CallManager Release 3.0 using MGCP, you must identify the primary server and any backup Cisco CallManager servers in case the primary server becomes unavailable. You must also configure each voice gateway as an MGCP gateway in Cisco CallManager, as described in the *Cisco CallManager Administration Guide*. Finally, you must configure the voice ports installed on your gateway.

To enable generic MGCP support on a Cisco voice gateway, enter the following commands from the global configuration mode prompt:

```
Gateway(config)#mgcp
Gateway(config)#mgcp call-agent hostname
```

where *hostname* identifies the Cisco CallManager server (or possibly a generic MGCP call agent).

To enable support for Cisco CallManager within MGCP, enter the following command:

```
Gateway(config)#ccm-manager MGCP
```

Cisco CallManager controls dial-plan-related configuration elements, and they should not be configured in the Cisco voice gateway for MGCP-managed endpoints (those with **application MGCPAPP** in the dial-peer statement). You should *not* configure any of the following elements when using MGCP:

- Destination pattern
- Session target
- Expansion numbers
- Connection PLAR/tie-line/trunk (voice port)
- codec

Note H.323 and MGCP configurations will coexist when you enable MGCP gateway fallback or Survivable Remote Site Telephony (SRST).

Enabling Switchover and Switchback

To identify up to two backup Cisco CallManager servers, enter the following command:

```
Gateway(config)#ccm-manager redundant-host hostname1  hostname2
```

where *hostname1* identifies the first backup Cisco CallManager server using the DNS host name or dotted decimal format, and *hostname2* identifies the second backup Cisco CallManager server.

If you configure one or two backup Cisco CallManager servers, you can control how the gateway behaves if the primary server becomes unavailable at some point and then later becomes available again. This is called *switchback*.

To configure gateway switchback, enter the following command:

```
Gateway(config)#ccm-manager switchback
 {graceful|imm[ediate]|sch[edule-time] hh:mmm|uptime[-delay] minutes}
```

During switchover and switchback, Cisco CallManager maintains active connected calls. Transient calls (calls in progress or on hold without an active voice connection) are torn down. An exception applies for PRI interfaces that MGCP controls, in which case both active and transient calls are torn down during switchover and switchback. Table 5-1 describes each switchback option.

Table 5-1 Switchback Command Options

Switchback Command Option	Function
graceful	The default value. Completes all outstanding calls before returning the gateway to the control of the primary Cisco CallManager server.
immediate	Returns the gateway to the control of the primary Cisco CallManager server without delay, as soon as the network connection to the server is reestablished.
schedule-time *hh:mm*	Returns the gateway to the control of the primary Cisco CallManager server at the specified time, where *hh:mm* is the time according to a 24-hour clock. If the configured schedule time is earlier than the time at which the gateway reestablishes a network connection to the primary server, the switchback will occur at the specified time on the following day.
uptime-delay *minutes*	Returns the gateway to the control of the primary Cisco CallManager server. This occurs when the primary server runs for a specified number of minutes after a network connection is reestablished to the primary server. Permitted values range from 1 to 1440 (1 minute to 24 hours).

You can also manually redirect a Cisco voice gateway to the backup Cisco CallManager server by entering the following command:

```
Gateway(config)#ccm-manager switchover-to-backup
```

The switchover occurs immediately with a manual redirect. This command does not switch the gateway to the backup Cisco CallManager server if you have the switchback option set to **immediate** and the primary Cisco CallManager server is still running.

To view the current configuration of a Cisco voice gateway, enter the **show ccm-manager** command from privileged EXEC mode. Example 5-1 illustrates a typical display that appears in response to this command.

Example 5-1 Output of the show ccm-manager Command

```
router# sh ccm-manager

MGCP Domain Name: router
Priority        Status                  Host
============================================================
Primary         Registered              172.20.71.44
First backup    None
Second backup   None

Current active Call Manager:    172.20.71.44
Backhaul/Redundant link port:   2428
Failover Interval:              30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            00:54:14 (elapsed time: 00:00:13)
Last MGCP traffic time:         00:54:14 (elapsed time: 00:00:13)
Last switchover time:           None
Switchback mode:                Graceful
MGCP Fallback mode:             Not Selected
Last MGCP Fallback start time:  00:00:00
Last MGCP Fallback end time:    00:00:00

PRI Backhaul link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:   0
        Recv failures:   0
        Packets xmitted: 0
        Xmit failures:   0
    PRI Ports being backhauled:
       Slot 1, port 1
Configuration Auto-Download Information
=======================================
 No configurations downloaded
```

```
Current state: Automatic Configuration Download feature is disabled
Configuration Error History:
FAX relay mode: cisco-fax-relay
```

Configuring FXS and FXO Analog Ports

You use the same commands to configure both Foreign Exchange Service (FXS) and Foreign Exchange Office (FXO) ports. The gateway recognizes the type of voice interface card that is installed in each voice network module and applies the configuration you enter based on the port position you specify in the command.

To enable FXS or FXO ports with MGCP, enter the following commands:

```
Gateway(config)# dial-peer voice number pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port portnumber
```

To use these commands, replace *number* with a unique numeric ID, and replace *portnumber* with the port identifier in the form *slot#/voice module#/port#*. Use the **application MGCPAPP** command to place the port under control of the Cisco CallManager MGCP call agent.

For example, the following command string configures voice port 0 in voice interface card 1 with MGCP:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port 1/1/0
```

Configuring T1-CAS E&M Emulation

You can use MGCP with the following emulation types:

- E&M Wink Start
- E&M Delay Start

To configure T1-CAS E&M emulation with MGCP using Cisco CallManager Administrator, perform the following steps and configure the route pattern and dial plan:

Step 1 Identify the port number and then enter the configuration information provided by your local carrier, as in the following example:

```
Gateway(config)# controller T1 1/port#
Gateway(config-controller)# framing esf
Gateway(config-controller)# clock source internal
Gateway(config-controller)# linecode b8zs
```

Step 2 Assign time slots to the DS-0 group and identify the emulation type.

You can define each DS-0 group to use FXS, FXO, or E&M, using the following command:

```
Gateway(config-controller)# ds0-group group groupnumber timeslots
<timeslotnumber> type emulationtype
```

Replace *emulationtype* with e&m-wink-start or e&m-delay-dial.

Replace *groupnumber* with the DS-0 group number and replace *timeslotnumber* with the number of DS-0 time slots to allocate to the group. For example, the following command configures the first DS-0 group with one time slot using FXS emulation in loop-start mode:

```
Gateway(config-controller)# ds0-group 0 timeslots 1 type
fxs-loop-start
```

You can configure DS-0 hunt groups by assigning a range of time slots to a DS-0 group and then configuring multiple voice peers with the same destination pattern pointing to multiple voice ports.

For example, the following command assigns 12 time slots to DS-0 group 1:

```
Gateway(config-controller)# ds0-group 1 timeslots 1-12 type
fxs-loop-start
```

Step 3 Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
portnumber:ds0group
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Example 5-2 shows typical use of commands for configuring T-1 CAS E&M Emulation for MGCP.

Example 5-2 T-1 CAS E&M Emulation for MGCP

```
Gateway(config-controller)# ds0-group 0 timeslots 1 type
e&m-wink-start
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port 1/0:0
```

Configuring T1/E1 (ISDN-PRI) Ports

To configure an E1/T1 multiflex interface with ISDN-PRI signaling, use the Cisco IOS command line interface to perform the procedures in this section.

Configuring T1 Interfaces

To configure a new T1 interface or to change the configuration of an existing interface, perform the following procedure:

Step 1 Identify the port number and enter line-specific information provided by your local carrier.

a. Choose the T1/PRI interface to configure:

```
Gateway(config)# controller t1 1/0
```

This example configures a T1 interface in slot 1 and unit 0.

b. Specify which end of the circuit provides clocking:

```
Gateway(config-controller)# clock source line
```

Set the clock source to use internal clocking only for testing the network. Set one end of the T1 line to internal.

c. Specify the framing type:

```
Gateway(config-controller)# framing esf
```

d. Specify the line code format:

```
Gateway(config-controller)# linecode b8zs
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped:

```
Gateway(config-controller)# pri-group timeslots 1-24 service mgcp
```

For multiflex trunk interfaces, you can configure only channel 0.

b. Configure each PRI group as a virtual serial interface:

```
Gateway(config-controller)# interface serial 1/0:23
```

c. Specify the T1 interface and unit number to modify:

```
interface Serial 1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Step 3 Configure the PRI port by entering the following command:

```
Gateway(controller-t1)# pri-group timeslots 1-24 service mgcp
```

Step 4 Bind Layer 3 to the Cisco CallManager for PRI Q.931:

```
Gateway(config-if)# isdn bind-l3 ccm-manager
```

This command backhauls (tunnels) ISDN Layer 3 and above to the Cisco CallManager.

PRI/Q.931 signaling backhaul transports signals (Q.931 and higher layers) for processing from a PRI trunk to a MGCP call agent. The PRI trunk must be physically connected to an MGCP gateway.

The ISDN lower layer information (Q.921 and below) is terminated and processed on the gateway. The Layer 3 information (Q.931 and above) is transported over TCP to the Cisco CallManager (MGCP call agent).

Step 5 Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
<portnumber>:<ds0group>
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Step 6 To view the status of the PRI line, enter the following command:

```
Gateway # show ccm-manager backhaul
```

This command displays information about the status of the TCP backhaul link and the status of any PRI D-channels in the gateway. Example 5-3 shows the type of information the system displays.

Example 5-3 PRI Backhaul Status—T1

```
PRI Backhaul link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:   0
        Recv failures:   0
        Packets xmitted: 0
        Xmit failures:   0
    PRI Ports being backhauled:
       Slot 1, port 1
       Slot 1, port 0
```

Sample Configuration

The following example shows the overall configuration required to enable MGCP on a T1/PRI line:

```
isdn switch-type primary-5ess
controller T1 1/0
 framing crc4
 linecode hdb3
 pri-group timeslots 1-24 service mgcp
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
 isdn bind-l3 ccm-manager
dial-peer voice 1 pots
 application mgcpapp
 port 1/0:0
```

Configuring E1 Interfaces

Use the following procedure to configure a new E1 interface (balanced or unbalanced) or to change the configuration of an existing interface:

Step 1 Identify the port number and enter line-specific information provided by your local carrier.

a. Choose the E1/PRI interface to configure by entering the following command from Global configuration mode:

```
Gateway(config)# controller e1 1/0
```

This example configures an E1 interface in slot 1 and unit 0.

b. Specify the framing type:

```
Gateway(config-controller)# framing crc4
```

c. Specify the line code format:

```
Gateway(config-controller)# linecode hdb3
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped:

```
Gateway(config-controller)# pri-group timeslots 1-31 service mgcp
```

b. Configure each PRI group as a virtual serial interface:

```
Gateway(config-controller)# interface serial 1/0:15
```

c. Specify the E1 interface and unit number to modify:

```
interface Serial1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
```

Step 3 Configure the PRI port by entering the following command:

```
Gateway(controller-e1)# pri-group timeslots 1-31 service mgcp
```

Step 4 Bind Layer 3 to the Cisco CallManager for PRI Q.931:

```
Gateway(config-if)# isdn bind-l3 ccm-manager backhaul q931
```

This command backhauls (tunnels) ISDN Layer 3 and above to the Cisco CallManager.

PRI/Q.931 signaling backhaul transports signals (Q.931 and higher layers) for processing a PRI trunk that is physically connected to an MGCP call agent.

The ISDN lower layer information (Q.921 and below) is terminated and processed on the gateway. The Layer 3 information (Q.923 and above) is transported over TCP to the Cisco CallManager (MGCP call agent).

Step 5 Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
<portnumber>:<ds0group>
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Step 6 To view the status of the PRI line, enter the following command:

```
Gateway# show ccm-manager backhaul
```

This command displays information about the status of the PRI backhaul link and the status of any PRI D channels in the gateway. The following example shows the type of information the system displays.

Example 5-4 PRI Backhaul Status

```
PRI Backhaul link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:   0
        Recv failures:   0
        Packets xmitted: 0
        Xmit failures:   0
    PRI Ports being backhauled:
       Slot 1, port 1
       Slot 1, port 0
```

Sample Configuration

The following example shows the overall configuration required to enable MGCP on a E1/PRI line:

```
isdn switch-type primary-5ess
controller E1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-31 service mgcp
interface Serial 1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
 isdn bind-l3 ccm-manager backhaul q931
dial-peer voice 1 pots
 application mgcpapp
 port 1/0:0
```

Where to Go Next

At this point, make sure that Cisco CallManager is properly configured to provision the voice gateway and to configure MGCP endpoints or H.323 route patterns as required. Refer to the documentation and online help provided with Cisco CallManager. Refer to the Cisco IOS software configuration guide and command reference publications for more advanced configuration topics. For troubleshooting information, refer to the system error messages and debug command reference publications.

Access these publications on the Documentation CD-ROM that came with your gateway, or on the World Wide Web from the Cisco home page.

H.323 Gateway Configuration

Compared to Media Gateway Control Protocol (MGCP), H.323 requires more configuration on a gateway because the gateway must maintain the dial plan and route pattern. The gateway must have enough information to direct calls to the correct end point, which must be an H.323-capable device.

When using H.323, ensure that you configure Cisco CallManager correctly to provision the Catalyst 4224 as an H.323 gateway, with route patterns configured to route calls to a Catalyst 4224.

To provision the gateway using Cisco CallManager Administration, select the **Add a New Gateway** option from **Device > Gateway**. Assign the Gateway Type as H.323 Gateway, and the Device Protocol as H.225.

To configure a route pattern using CCM Administrator, select **Route Plan > Route Pattern** and enter the route pattern. Then, select **Cisco Catalyst 4224 gateway** from a drop-down list, click the **Route this option** button for the Route Option, and check the **Provide Outside Dial Tone** box for Offnet (the default is OnNet).

Perform the following steps to complete H.323 configuration:

Step 1 Identify the port number and enter line-specific information provided by your local carrier, as described in the following sections:

- Configuring T1-CAS Analog Emulation (H.323), page 5-19
- ISDN BRI Configuration (H.323), page 5-24
- T1/E1 Configuration (H.323), page 5-31
- E&M Trunk Line Configuration (H.323), page 5-35

Step 2 Configure parameters for the voice interface you are using, as described in the section referred to above.

Step 3 Configure H.323 endpoints connected to the Catalyst 4224 voice ports.

To configure plain old telephone service (POTS) dial peers, use the following command strings:

```
c4224(config)# dial-peer voice number pots
c4224(config-dial-peer)# destination-pattern endpoint#
c4224(config-dial-peer)# port 1/portnumber:DS0groupnumber
```

Replace:

- *number* with a unique numeric identifier for each dial peer
- *endpoint#* with the E.164 telephone extension of the POTS dial peer
- *portnumber* with 0 or 1, depending on which T1 port you are using
- *DS0groupnumber* with a numeric digit from 0 to 23 for each DS-0 group you are configuring

For example, the following commands could be used to route all calls with the prefix 222 to the DS-0 hunt group 1 of controller T1 1/0:

```
c4224(config)# dial-peer voice 222 pots
c4224(config-dial-peer)# destination-pattern 222....
c4224(config-dial-peer)# port 1/0:1
c4224(config-dial-peer)# prefix 222
```

The prefix command at the end is required to replace the digits that the Catalyst 4224 strips off from the dialed digit string based on the wildcard destination pattern.

Step 4 Configure H.323 endpoints connected to the Catalyst 4224 Ethernet port.

To configure H.323 endpoints, use the following command strings:

```
c4224(config)# dial-peer voice number voip
c4224(config-dial-peer)# destination-pattern endpoint#
c4224(config-dial-peer)# session target {ipv4:ipaddress|dns:hostname}
c4224(config-dial-peer)# codec codecid
```

Replace:

- *number* with a unique numeric identifier for each dial peer
- *endpointnumber* with the telephone extension of the dial peer
- *ipaddress* or *hostname* with the IP address or Domain Name System (DNS) host name of the VoIP dial peer

If you use the IP address, it must be preceded by the parameter **ipv4**. If you use the DNS host name, this must be preceded by the parameter **dns**, and the host name must resolve correctly to the IP address of the target. Finally, you must identify the coder-decoder (codec) used by the Voice over IP (VoIP) dial peer.

For example, the following commands assign extension 2001 to the IP device with the network address 192.168.100.1:

```
c4224(config)# dial-peer voice 1 voip
c4224(config-dial-peer)# destination-pattern 2001
c4224(config-dial-peer)# session target ipv4:192.168.100.1
c4224(config-dial-peer)# codec g711u1aw
```

Step 5 Direct calls using wildcard destination patterns, as needed.

You can use wildcard destination patterns to simplify your dial plan configuration. For instance, you can direct all incoming calls starting with specific digits, such as 525, to a Cisco CallManager configured as an H.323 endpoint. You might direct all calls starting with a 9 to voice ports connected to the Public Switched Telephone Network (PSTN), or direct all calls beginning with an 8 to a private branch exchange (PBX).

```
c4224(config-dial-peer)# destination-pattern pattern ....
```

For example, the following command directs all calls starting with 525 to a Cisco CallManager with the DNS host name CCM30:

```
c4224(config-dial-peer)# destination-pattern 525....
c4224(config-dial-peer)# session target dns:CCM30
```

The number of digits that you substitute for *pattern* plus the number of periods in the wildcard (...) must match the total number of digits configured for use by the Catalyst 4224 in Cisco CallManager Administration. Also, remember that the numbers that you substitute for *pattern* are removed by the Catalyst 4224. When the call is forwarded to the destination number, only the digits in the position of the wildcard pattern (...) are received by the destination endpoint. If you want to replace the digits that are stripped off (or add a different set of digits), use the **prefix** command.

Step 6 Complete and save the configuration by entering the following commands:

```
c4224# line con 0
c4224# transport input none
c4224# line aux 0
c4224# line vty 0 4
c4224# login
c4224# no scheduler allocate
c4224# end
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

Configuring T1-CAS Analog Emulation (H.323)

You can connect the T1-CAS (channel-associated signaling) port on a Catalyst 4224 to one of the following:

- The PSTN using Foreign Exchange Office (FXO) emulation
- A T1 channel bank using Foreign Exchange Station (FXS) emulation
- A PBX with a trunk (tie) line using Ear and Mouth (E&M) emulation

To configure T1-CAS analog emulation with H.323 T1, perform the following steps. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1 Identify the port number and then enter the configuration information provided by your telco, as in the following example:

```
c4224(config)# controller T1 1/port#
c4224(config-controller)# framing esf
c4224(config-controller)# clock source internal
c4224(config-controller)# linecode b8zs
```

Step 2 Assign time slots to the DS-0 group and identify the emulation type.

You can define each DS-0 group to use FXS, FXO, or E&M, using the following command:

```
c4224(config-controller)# dso-group group groupnumber
timeslots timeslotnumber type emulationtype
```

Replace:

- *groupnumber* with the DS-0 group number
- *timeslotnumber* with the number of DS-0 time slots to allocate to the group.
- *emulationtype* with one of the modes described in Table 5-2.

Table 5-2 T-1 Emulation Types

Emulation Type	Function
fxs-loop-start	Uses FXS emulation in loop-start mode.
fxs-ground-start	Uses FXS emulation in ground-start mode.
fxo-loop-start	Uses FXO emulation in loop-start mode.
fxo-ground-start	Uses FXO emulation in ground-start mode.
e&m-immediate-start	Uses E&M emulation in immediate-start mode.
e&m-wink-start	Uses E&M emulation in wink-start mode.
e&m-delay-dial	Uses E&M emulation in immediate-delay dial mode.

For example, the following command configures the first DS-0 group with one time slot using FXS emulation in loop-start mode:

```
c4224(config-controller)# dso-group 0 timeslots 1 type fxs-loop-start
```

You can configure DS-0 hunt groups by assigning a range of time slots to a DS-0 group, and then configuring multiple voice peers with the same destination pattern pointing to multiple voice ports.

For example, the following command assigns 12 time slots to DS-0 group 1:

```
c4224(config-controller)# dso-group 1 timeslots 1-12 type
fxs-loop-start
```

Note After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Managing Input Gain for Cisco IP Voice Applications

When using the FXO ports on a Catalyst 4224, set the input gain to a value provides adequate audio quality for Cisco IP voice applications or the Cisco 7960 IP Phone. Input gain values higher than 12 may cause dual tone multifrequency (DTMF) recognition difficulties.

Cisco recommends that you use the Cisco CallManager graphical user interface to set the input gain. However, you can also enter the following series of commands from the Cisco IOS command line to set input gain:

```
c4224# configure terminal
c4224(config)# voice-port x/x/x input gain value
```

Permitted entries for *value* are from -6 to 14.

FXS Emulation Example

By connecting the T1-CAS port on a Catalyst 4224 to a T1 channel bank using FXS emulation, you can achieve high port density when connecting POTS and VoIP endpoints. You can configure the dial plan for this configuration by treating Cisco CallManager as the only H.323 endpoint, or by configuring H.323

endpoints on a Catalyst 4224. If you configure Cisco CallManager as an H.323 endpoint, you must use Cisco CallManager Administration to define the route patterns required to route calls to the Catalyst 4224.

The following example illustrates how to configure a single DS-0 group. Repeat the relevant commands to configure additional groups. This example is for a scenario in which all of the POTS devices connected to a T1 channel bank are configured with a destination number beginning with 526. In this example, Cisco CallManager has the host name CCM30 and is configured as an H.323 endpoint that manages all the telephones and other devices on the IP network. The devices on the IP network have numbers beginning with 525.

```
c4224(config)# interface FastEthernet5/0
c4224(config)# ipaddress 172.20.71.48 255.255.255.0
c4224(config)# no ip directed-broadcast
c4224(config)# no keepalive
c4224(config)# duplex auto
c4224(config)# speed 10

c4224(config)# controller T1 1/0
c4224(config-controller)# framing esf
c4224(config-controller)# clock source internal
c4224(config-controller)# linecode b8zs
c4224(config-controller)# dso-group 0 timeslots 1 type fxo-loop-start

c4224(config)# dial-peer voice 1 pots
c4224(config-dial-peer)# destination-pattern 526....
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 525....
c4224(config-dial-peer)# session target dns:CCM30
c4224(config-dial-peer)# codec g711u1aw

c4224# line con 0
c4224# transport input none
c4224# line aux 0
c4224# line vty 0 4
c4224# login
c4224# no scheduler allocate
c4224# end
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

FXO Emulation Example

To use FXO emulation to connect the T1-CAS port to the PSTN, you must have Direct Inward Dialing (DID) enabled on incoming DS-0 groups. DID allows the gateway or Cisco CallManager to identify the extension to which each call on an incoming DS-0 group is directed. Because DID only works on incoming connections, you must have separate DS-0 groups allocated for incoming and outgoing calls. To configure the gateway to accept DID information, enter the following command:

```
c4224(config-dial-peer)# direct-inward-dial
```

The first and last parts of the configuration are the same as for the FXO example. However, you must configure your DS-0 groups for FXS by changing the emulation type and enabling direct inward dialing (DID). Then, enter the destination patterns required for routing voice calls to and from the PSTN. The commands required to make these changes are shown below:

```
c4224(config-controller)# dso-group 0 timeslots 1 type fxo-loop-start

c4224(config)# dial-peer voice 1 pots
c4224c4224(config-dial-peer)# direct-inward-dial
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 9.......
```

E&M Emulation Example

To connect the T1-CAS port to a trunk (tie) line using E&M emulation, you can enable one of the following modes:

- E&M immediate start
- E&M wink start
- E&M delay dial

The first and last parts of the configuration are the same as for the FXO example. However, you must configure your DS-0 groups for E&M by changing the emulation type. Then, enter the destination patterns required for routing voice

calls to and from the PBX to which the gateway is connected. The commands required to make these changes are shown below (all the extensions on the PBX begin with the prefix 625):

```
c4224(config-controller)# dso-group 0 timeslots 1 type
e&m-immediate-start
c4224(config)# dial-peer voice 1 pots
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 625....
```

ISDN BRI Configuration (H.323)

To configure an ISDN BRI interface, perform the following steps. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1 Enter an ISDN switch type by entering the following command in global configuration mode:

```
c4224(config)# isdn switch-type switch-type
```

See Table 5-3 for a list of ISDN switch types.

Note Switch types configured in interface configuration mode override this setting for the configured interface.

Table 5-3 **ISDN Switch Types**

Region	ISDN Switch Type	Description
Australia	basic-ts013	Australian TS013 switches
Europe	basic-1tr6	German 1TR6 ISDN switches
	basic-nwnet3	Norwegian NET3 ISDN switches (phase 1)
	basic-net3	NET3 ISDN switches (UK and others)
	vn2	French VN2 ISDN switches
	vn3	French VN3 ISDN switches
Japan	ntt	Japanese NTT ISDN switches
New Zealand	basic-nznet3	New Zealand NET3 switches
North America	basic-5ess	AT&T basic rate switches
	basic-dms100	NT DMS-100 basic rate switches
	basic-nil1	National ISDN-1 switches

Step 2 Assign the switch type to the interface by entering the following commands. The following example assigns the switch type basic-5ess:

```
c4224(config)# interface bri 0/0
c4224(config-if)# isdn switch-type basic-5ess
```

For details about configuring Cisco CallManager, refer to the *Cisco CallManager Administration Guide*.

Note After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Configuring ISDN BRI Lines

Before using a Catalyst 4224 with an ISDN BRI interface, you must order a correctly configured ISDN BRI line from your local telecommunications service provider.

The ordering process varies from provider to provider and from country to country; however, here are some general guidelines:

- Ask for two channels to be called by one number.
- Ask for delivery of calling line identification, also known as caller ID or Automatic Number Identification (ANI).

ISDN BRI Provisioning by Switch Type

ISDN BRI provisioning refers to the types of services provided by the ISDN BRI line. Although provisioning is performed by your ISDN BRI service provider, you must tell the provider what you want.

Table 5-4 lists the provisioning you should order for your Catalyst 4224 for each switch type.

Table 5-4 ISDN Provisioning by Switch Type

Switch Type	Provisioning
5ESS Custom BRI	For voice only: • (Use these values only if you have an ISDN telephone connected.) • 2 B channels for voice • Multipoint • Terminal type = D • 2 directory numbers assigned by service provider • 2 SPID[1]s required, assigned by service provider • MTERM = 2 • Number of call appearances = 1 • Display = No • Ringing/idle call appearances = 1 • Autohold = no • Onetouch = no • Request delivery of calling line ID on Centrex lines • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2

Table 5-4 ISDN Provisioning by Switch Type (continued)

Switch Type	Provisioning
5ESS National ISDN (NI-1) BRI	• Terminal type = A • 2 B channels for voice • 2 directory numbers assigned by service provider • 2 SPIDs required, assigned by service provider • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2
DMS-100 BRI	• 2 B channels for voice • 2 directory numbers assigned by service provider • 2 SPIDs required, assigned by service provider • Functional signaling • Dynamic TEI[2] assignment • Maximum number of keys = 64 • Release key = no, or key number = no • Ringing indicator = no • EKTS = no • PVC = 2 • Request delivery of calling line ID on Centrex lines • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2

1. Service profile identifier
2. Terminal endpoint identifier

Defining ISDN Service Profile Identifiers

Some service providers assign service profile identifiers (SPIDs) to define the services to which an ISDN device subscribes. If your service provider requires SPIDs, your ISDN device cannot place or receive calls until it sends a valid SPID to the service provider when initializing the connection.

A SPID is usually a seven-digit telephone number plus some optional numbers, but service providers might use different numbering schemes. SPIDs have significance at the local access ISDN interface only; remote Catalyst 4224s are never sent the SPID.

Currently, only DMS-100 and NI-1 switch types require SPIDs. Two SPIDs are assigned for the DMS-100 switch type, one for each B channel. The AT&T 5ESS switch type might support SPIDs, but Cisco recommends that you set up that ISDN service without SPIDs.

If your service provider assigns you SPIDs, you must define these SPIDs on the Catalyst 4224. To define SPIDs and the local directory number (LDN) on the gateway for both ISDN BRI B channels, use the following **isdn spid** commands:

```
c4224 (config-if)# isdn spid1 spid-number [ldn]
c4224 (config-if)# isdn spid2 spid-number [ldn]
```

Note Although the LDN is an optional parameter in the command, you might need to enter it so the gateway can answer calls made to the second directory number.

For further information on configuring ISDN, refer to the chapters “Configuring ISDN” and “Configuring DDR” in the *Wide-Area Networking Configuration Guide*.

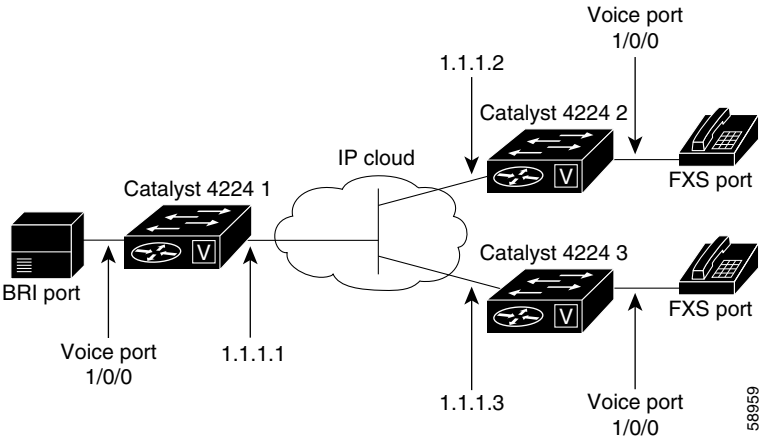
BRI Direct-Inward Dialing Configuration

The example illustrated in Figure 5-1 shows how to configure a BRI port for direct-inward dialing (DID). This configuration allows the called number information from the ISDN Q.931 setup message to be used for routing on an ISDN line.

In this example, a call comes in to Catalyst 4224 1 on the BRI port. The DID information allows the Catalyst 4224 to route the call based on the called number. If the called number is 2xxx, the call is routed to Catalyst 4224 2000; if the called number is 3xxx, the call is routed to Catalyst 4224 3000.

Figure 5-1 illustrates the topology of this connection example.

Figure 5-1 Configuring DID on a BRI Port



Gateway 1 Configuration

```
dial-peer voice 1 pots
 port 1/0/0
 destination-pattern 1...
 direct-inward-dial
dial-peer voice 2 voip
 session target ipv4:1.1.1.2
 destination-pattern 2...
dial-peer voice 3 voip
 session target ipv4:1.1.1.3
 destination-pattern 3...
```

Gateway 2 Configuration

```
dial-peer voice 1 pots
 port 1/0/0
 destination-pattern 2000
```

T1/E1 Configuration (H.323)

To configure an ISDN PRI interface or T1/E1 multiflex trunk interface on your Catalyst 4224, use configuration mode.

This section contains the following topics:

Configuring T1 Interfaces

Use the following procedure to configure a new T1 interface or to change the configuration of an existing interface. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1 Identify the port number and enter line-specific information provided by your local carrier.

a. Select the T1/PRI interface to configure.

```
c4224(config)# controller t1 1/0
```

This example configures a T1 interface in slot 1 and unit 0.

b. Specify which end of the circuit provides clocking.

```
c4224(config-controller)# clock source line
```

The clock source should be set to use internal clocking only for testing the network or if the full T1 line is used as the channel group. Only one end of the T1 line should be set to internal.

c. Specify the framing type.

```
c4224(config-controller)# framing esf
```

d. Specify the line code format.

```
c4224(config-controller)# linecode b8zs
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped.

```
c4224(config-controller)# pri-group timeslots 1-24
```

For multiflex trunk interfaces, only channel 0 can be configured.

b. Configure each pri-group as a virtual serial interface.

```
c4224(config-controller)# interface serial 1/0:15
```

c. Specify the T1 interface, unit number, and channel group to modify, as in the following example input:

```
interface Serial pri-group
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Note After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

T1/PRI Configuration Example

```
isdn switch-type primary-5ess
isdn voice-call-failure 0
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Configuring E1 Interfaces

Use the following procedure to configure a new E1 or CE1/PRI interface (balanced or unbalanced) or to change the configuration of an existing interface.

Step 1 Identify the port number and enter line-specific information provided by your local carrier.

a. Select the CE1/PRI interface to configure by entering the following command from global configuration mode. This example configures an E1 interface in slot 1 and unit 0.

```
c4224(config)# controller e1 1/0
```

b. Specify the framing type.

```
c4224(config-controller)# framing crc4
```

c. Specify the line code format.

```
c4224(config-controller)# linecode hdb3
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped.

```
c4224(config-controller)# pri-group timeslots 1-31
```

b. Configure each channel group as a virtual serial interface.

```
c4224(config-controller)# interface serial 1/0:31
```

c. Specify the E1 interface, unit number, and channel group to modify, as in the following example:

```
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Note After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

E1/PRI Configuration Example

```
isdn switch-type primary-5ess
isdn voice-call-failure 0
controller E1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-31
interface Serial 1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

E&M Trunk Line Configuration (H.323)

The section illustrates how to configure VoIP to link PBX users with E&M trunk lines.

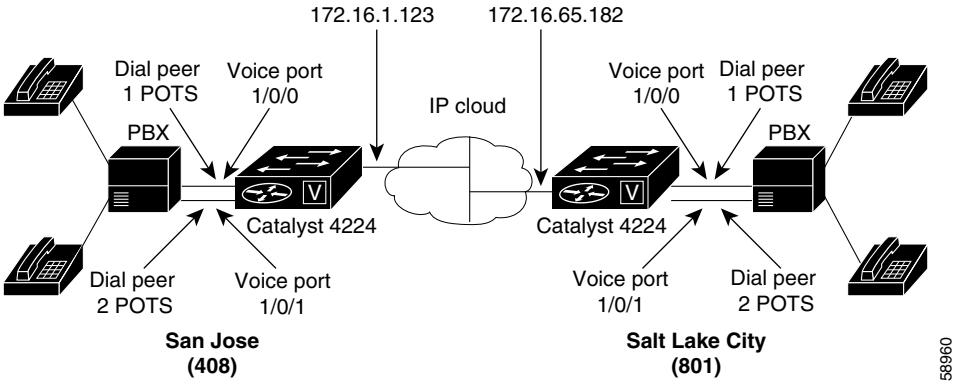
This section contains the following topics:

Scenario

Suppose that a company wants to connect two offices: one in San Jose, California, and the other in Salt Lake City, Utah. Each office has an internal telephone network using a PBX connected to the voice network with an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and ImmediateStart signaling. Each E&M interface connects to the gateway using two voice interface connections. Users in San Jose dial 8-569 and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial 4-527 and then the extension number to reach a destination in San Jose.

Figure 5-2 illustrates the topology of this configuration.

Figure 5-2 Linking PBX Users with E&M Trunk Lines



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

Handling Incoming Caller ID Digits on an E&M Port

When using an H.323 T1-CAS E&M port on the Catalyst 4224, Incoming Caller ID Digits may not be processed correctly by Cisco CRA, such as Cisco IP Auto-Attendant.

Depending on the T1-CAS line provisioning, incoming dialed number identification service (DNIS) digits received by the Catalyst 4224 after its first wink to the Central Office (CO) are treated as user-entered digits and are sent to the remote endpoint as out-of-band DTMF digits. If the remote endpoint is a Cisco CRA application, the out-of-band digits will be interpreted as a user entry and will change the application response.

There are two ways to handle this situation:

- Ask the T1 service provider to stop sending DNIS digits.
- Configure an IP phone with the same directory number as the incoming DNIS, then modify the "forward all" selection for this phone so that it sends the incoming call to the desired destination.

Gateway San Jose Configuration

```
hostname sanjose
!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern +527....
 port 1/0/0
!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern +527....
 port 1/0/1
!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern +569....
 session target ipv4:172.16.65.182
!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
voice-port 1/0/1
 signal immediate
 operation 4-wire
 type 2
!Configure the serial interface
interface serial 0/0
 description serial interface type dce (provides clock)
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Gateway Salt Lake City Configuration

```
hostname saltlake
!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern +569....
 port 1/0/0
!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern +569....
 port 1/0/1
!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern +527....
 session target ipv4:172.16.1.123
```

```
!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
!Configure the serial interface
interface serial 0/0
 description serial interface type dte
 ip address 172.16.65.182
 no shutdown
```

Note The PBXs should be configured to pass all DTMF signals to the gateway. Cisco recommends that you do not configure "store-and-forward" tone.

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

Note After completing E&M configuration, configure the route pattern and dial plan and save your configuration, as described in the "H.323 Gateway Configuration" section on page 5-16.

At this point in the configuration process, make sure that Cisco CallManager is properly configured to provision the Catalyst 4224 and to configure MGCP endpoints or H.323 route patterns, as required. Refer to the *Cisco CallManager Administration Guide* or to the online help for the application.

Refer to the Cisco IOS configuration guides and command references for details about specific VoIP commands and options.

Configuring VoIP

The Catalyst 4224 Access Gateway Switch (Catalyst 4224) provides Voice-over-IP (VoIP) gateway applications for a *micro branch* office. This section provides comprehensive information on how to configure VoIP on your Catalyst 4224.

This section contains the following topics:

Prerequisite Tasks

Before you can configure your Catalyst 4224 to use VoIP, you need to perform the following tasks:

- Establish a working IP network. For more information about configuring IP, refer to the "IP Overview," "Configuring IP Addressing," and "Configuring IP Services" sections in the *Network Protocols Configuration Guide, Part 1* for Cisco IOS Release 12.0 T.
- Install the voice interface cards (VICs) in your Catalyst 4224. For more information about installing a VIC in your Catalyst 4224, refer to the *Cisco WAN Interface Cards Hardware Installation Guide*.
- Complete your company dial plan.
- Establish a working telephony network based on your company dial plan.
- Integrate your dial plan and telephony network into your existing IP network topology. Merging your IP and telephony networks depends on your particular IP and telephony network topology. In general, Cisco recommends the following:
  - Use canonical numbers wherever possible. Avoid situations in which numbering systems are significantly different on different Catalyst 4224s or access servers in your network.
  - Make routing and dialing transparent to the user; for example, try to avoid secondary dial tones from secondary switches.
  - Contact your private branch exchange (PBX) vendor for instructions on how to reconfigure the appropriate PBX interfaces.

After you have analyzed your dial plan and decided how to integrate it into your existing IP network, you are ready to configure your network devices to support VoIP.

Configuration Tasks

To configure VoIP on your Catalyst 4224, perform the following steps:

Step 1 Configure your IP network to support real-time voice traffic. See the "Configure IP Networks for Real-Time Voice Traffic" section on page 6-3 for information about selecting and configuring the appropriate quality of service (QoS) tool or tools to optimize voice traffic on your network.

Step 2 (Optional) If you plan to run Voice over Frame Relay, you need to consider certain factors so that VoIP runs smoothly. For example, a public Frame Relay cloud provides no guarantees for QoS. See the "Configure Frame Relay for VoIP" section on page 6-37 for information about deploying VoIP over Frame Relay.

Step 3 Use the **num-exp** command to configure number expansion if your telephone network is configured so that you can reach a destination by dialing only an extension number of the full E.164 telephone number. See the "Configure Number Expansion" section on page 6-12 for information about number expansion.

Step 4 Use the **dial-peer voice** command to define dial peers and switch to the dial-peer configuration mode. See the "Configure Dial Peers" section on page 6-15 and the "Additional VoIP Dial-Peer Configurations" section on page 6-33 for additional information about configuring dial peers and dial-peer characteristics.

Step 5 Configure your Catalyst 4224 to support voice ports. See the "Configure Voice Ports" section on page 6-22 for information about configuring voice ports.

Configure IP Networks for Real-Time Voice Traffic

You need to have a well engineered end-to-end network when running delay-sensitive applications such as VoIP. Fine-tuning your network to adequately support VoIP involves a series of protocols and features to improve QoS. It is beyond the scope of this document to explain the specific details relating to wide-scale QoS deployment.

Cisco IOS software provides many tools for enabling QoS on your network backbone. These tools include Random Early Detection (RED), Weighted Random Early Detection (WRED), Fancy Queuing (meaning custom, priority, or

weighted fair queuing), and IP precedence. To configure your IP network for real-time voice traffic, you need to consider the entire scope of your network and then select the appropriate QoS tool or tools.

To improve voice network performance, QoS must be configured throughout your network, not just on your Catalyst 4224 running VoIP. Not all QoS techniques are appropriate for all network routers. Edge routers and backbone routers in your network do not necessarily perform the same operations, and the QoS tasks they perform might differ as well. To configure your IP network for real-time voice traffic, you need to consider the functions of both edge and backbone routers in your network and then select the appropriate QoS tool or tools.

In general, edge routers perform the following QoS functions:

- Packet classification
- Admission control
- Bandwidth management
- Queuing

In general, backbone routers perform the following QoS functions:

- High-speed switching and transport
- Congestion management
- Queue management

Scalable QoS solutions require cooperative edge and backbone functions.

Although not mandatory, some QoS tools can be valuable in fine-tuning your network to support real-time voice traffic. To configure your IP network for QoS, perform one or more of the following tasks:

- Configure RSVP for Voice, page 6-5
- Configure Multilink Point-to-Point Protocol with Interleaving, page 6-7
- Configure Real-Time Transport Protocol Header Compression, page 6-9
- Configure Custom Queuing, page 6-11
- Configure Weighted Fair Queuing, page 6-12

Each of these tasks is discussed in the following sections.

Configure RSVP for Voice

Resource Reservation Protocol (RSVP) enables routers to reserve enough bandwidth on an interface to provide reliable, high-quality performance. RSVP allows end systems to request a particular QoS from the network. Real-time voice traffic requires network consistency. Without consistent QoS, real-time traffic can experience jitter, insufficient bandwidth, delay variations, or information loss. RSVP works in conjunction with current queuing mechanisms. The interface queuing mechanism (such as weighted fair queuing or WRED) must implement the reservation.

RSVP works well on PPP, HDLC, and similar serial line interfaces. It does not work well on multiaccess LANs. RSVP can be equated to a dynamic access list for packet flows.

You should configure RSVP to ensure QoS if the following conditions describe your network:

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

Enable RSVP

To minimally configure RSVP for voice traffic, you must enable RSVP on each interface where priority is required.

By default, RSVP is disabled so that it is backward compatible with systems that do not implement RSVP. To enable RSVP for IP on an interface, use the following command:

ip rsvp bandwidth [*interface-kbps*] [*single-flow-kbps*]

interface-kbps represents the maximum amount of bandwidth, in kbps, that may be allocated by RSVP flows and *single-flow-kbps* represents the maximum amount of bandwidth, in kbps, that may be allocated to a single flow. The range for both arguments is from 1 to 10,000,000.

This command starts RSVP and sets the bandwidth and single-flow limits. The default maximum bandwidth is up to 75 percent of the bandwidth available on the interface. By default, the amount reservable by a flow can be up to the entire reservable bandwidth.

On subinterfaces, RSVP applies to the more restrictive of the available bandwidths of the physical interface and the subinterface.

Reservations on individual circuits that do not exceed the single flow limit normally succeed. However, if reservations have been made on other circuits adding up to the line speed, and a reservation is made on a subinterface that itself has enough remaining bandwidth, the reservation will be refused. The reason the reservation will be refused is because the physical interface lacks supporting bandwidth.

A Cisco 1750 running VoIP and configured for RSVP requests allocations by using the following formula:

bps = packet_size + ip/udp/rtp header size x 50 per second

For the codec standard G.729, wherein voice is coded into 8-kbps streams, the allocation works out to be 24,000 bps. For G.711, the allocation is 80,000 bps.

For more information about configuring RSVP, refer to the "Configuring RSVP" section of the *Network Protocols Configuration Guide, Part 1* for Cisco IOS Release 12.0 T.

RSVP Configuration Example

The following example enables RSVP and sets the maximum bandwidth to 100 kbps and the maximum bandwidth per single request to 32 kbps. The example presumes that both VoIP dial peers have been configured.

```
c4224(config)# interface serial 0/0
c4224(config-if)# ip rsvp bandwidth 100 32
c4224(config-if)# fair-queue
c4224(config-if)# end
```

After enabling RSVP, you must also use the **req-qos** dial-peer configuration command to request an RSVP session on each VoIP dial peer. Otherwise, no bandwidth is reserved for voice traffic.

```
c4224(config)# dial-peer voice 211 voip
c4224(config-dial-peer)# req-qos controlled-load
c4224(config)# dial-peer voice 212 voip
c4224(config-dial-peer)# req-qos controlled-load
```

Configure Multilink Point-to-Point Protocol with Interleaving

Multiclass multilink Point-to-Point Protocol (PPP) interleaving allows large packets to be multilink-encapsulated and fragmented into smaller packets to satisfy the delay requirements of real-time voice traffic. Small real-time packets, which are not multilink-encapsulated, are transmitted between fragments of the large packets.

The interleaving feature also provides a special transmit queue for the smaller, delay-sensitive packets, enabling them to be transmitted earlier than other flows. Interleaving provides the delay bounds for delay-sensitive voice packets on a slow link that is used for other best-effort traffic.

In general, multilink PPP with interleaving is used in conjunction with weighted fair queuing and RSVP or IP precedence to ensure voice packet delivery. Use multilink PPP with interleaving and weighted fair queuing to define how data is managed; use RSVP or IP precedence to give priority to voice packets.

You should configure multilink PPP if the following conditions describe your network:

- Point-to-point connection using PPP encapsulation
- Links slower than two Mbps

Note Do not use multilink PPP on links faster than two Mbps.

You can configure multilink PPP support for interleaving on virtual templates, dialer interfaces, and ISDN Basic Rate Interface (BRI) or Primary Rate Interface (PRI) interfaces. To configure interleaving, perform the following tasks:

- Configure the dialer interface or virtual template, as defined in the relevant chapters of the *Dial Solutions Configuration Guide* for Cisco IOS Release 12.0T.
- Configure multilink PPP and interleaving on the interface or template.

To configure multilink PPP and interleaving on a configured and operational interface or virtual interface template, use the following interface configuration commands:

	Task	Command
Step 1	Enable Multilink PPP.	**ppp multilink**
Step 2	Enable real-time packet interleaving.	**ppp multilink interleave**
Step 3	Optionally, configure a maximum fragment delay of up to 20 milliseconds.	**ppp multilink fragment-delay** *milliseconds*
Step 4	Reserve a special queue for real-time packet flows to specified destination User Datagram Protocol (UDP) ports, allowing real-time traffic to have higher priority than other flows. This applies only if you have not configured RSVP.	**ip rtp reserve** *lowest-UDP-port range-of-ports* [*maximum-bandwidth*]

Note You can use the **ip rtp reserve** command instead of configuring RSVP. If you configure RSVP, this command is not required.

For more information about multilink PPP, refer to the "Configuring Media-Independent PPP and Multilink PPP" section in the *Dial Solutions Configuration Guide* for Cisco IOS Release 12.0 T.

Multilink PPP Configuration Example

The following example defines a virtual interface template that enables multilink PPP with interleaving and a maximum real-time traffic delay of 20 milliseconds, and then applies that virtual template to the multilink PPP bundle:

```
c4224(config)# interface virtual-template 1
c4224(config-if)# ppp multilink
c4224(config-if)# encapsulated ppp
c4224(config-if)# ppp multilink interleave
c4224(config-if)# ppp multilink fragment-delay 20
c4224(config-if)# ip rtp reserve 16384 100 64

c4224(config)# multilink virtual-template 1
```

Configure Real-Time Transport Protocol Header Compression

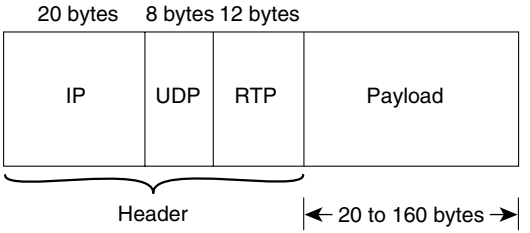
Real-Time Transport Protocol (RTP) is used for carrying audio traffic in packets over an IP network. RTP header compression compresses the RTP/UDP/IP header in an RTP data packet from 40 bytes to approximately 2 to 4 bytes, in most cases, as shown in Figure 6-1.

This compression feature is beneficial if you are running VoIP over slow links. Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if there is a lot of RTP traffic on that slow link.

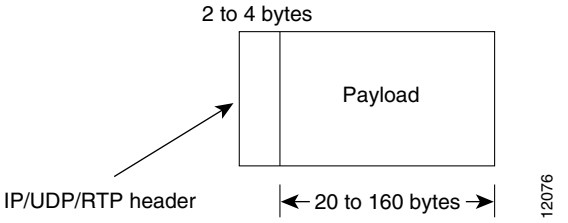
Typically, an RTP packet has a payload of approximately 20 to 160 bytes for audio applications that use compressed payloads. RTP header compression is especially beneficial when the RTP payload size is small (for example, compressed audio payloads between 20 and 50 bytes).

Figure 6-1 RTP Header Compression

Before RTP header compression:



After RTP header compression:



You should configure RTP header compression if the following conditions describe your network:

- Links slower than two Mbps
- Need to save bandwidth

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

Perform the following tasks to configure RTP header compression for VoIP. The first task is required, but the second task is optional.

- Enable RTP Header Compression on a Serial Interface, page 6-11
- Change the Number of Header Compression Connections, page 6-11

Enable RTP Header Compression on a Serial Interface

You need to enable compression on both ends of a serial connection. To enable RTP header compression, use the following interface configuration command:

```
c4224(config-if)# ip rtp header-compression [passive]
```

If you include the **passive** keyword, the software compresses outgoing RTP packets only if incoming RTP packets on the same interface are compressed. If you use the command without the **passive** keyword, the software compresses all RTP traffic.

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 following interface configuration command:

```
c4224(config-if)# ip rtp compression connections number
```

RTP Header Compression Configuration Example

The following example enables RTP header compression for a serial interface:

```
c4224(config)# interface serial0
c4224(config-if)# ip rtp header-compression
c4224(config-if)# encapsulation ppp
c4224(config-if)# ip rtp compression-connections 25
```

For more information about RTP header compression, see the "Configuring IP Multicast Routing" section of the *Network Protocols Configuration Guide, Part 1* for Cisco IOS Release 12.0 T.

Configure Custom Queuing

Some QoS features, such as IP RTP reserve and custom queuing, are based on the transport protocol and the associated port number. Real-time voice traffic is carried on UDP ports ranging from 16384 to 16624. This range is derived from the following formula:

16384 + (4 x number of voice ports in the c4224)

Custom queuing and other methods for identifying high-priority streams should be configured for these port ranges. For more information about custom queuing, refer to the "Managing System Performance" section in the *Configuration Fundamentals Configuration Guide* for Cisco IOS Release 12.0 T.

Configure Weighted Fair Queuing

Weighted fair queuing ensures that queues do not starve for bandwidth and that traffic gets predictable service. Low-volume traffic streams receive preferential service, while high-volume traffic streams share the remaining capacity, obtaining equal or proportional bandwidth.

In general, weighted fair queuing is used in conjunction with multilink PPP with interleaving and RSVP or IP precedence to ensure voice packet delivery. Use weighted fair queuing with multilink PPP to define how data is managed; use RSVP or IP precedence to give priority to voice packets. For more information about weighted fair queuing, refer to the "Managing System Performance" section in the *Configuration Fundamentals Configuration Guide* for Cisco IOS Release 12.0 T.

Configure Number Expansion

In most corporate environments, the telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. You can configure VoIP to recognize extension numbers and expand them into their full E.164 dialed number by using two commands in tandem: **destination-pattern** and **num-exp**. Before you configure these two commands, it helps to map individual telephone extensions with their full E.164 dialed numbers. You can do this by creating a number expansion table.

Create a Number Expansion Table

The example depicted in Figure 6-2 pertains to a small company that decides to use VoIP to integrate its telephony network with its existing IP network. The destination pattern (or expanded telephone number) associated with Catalyst 4224 1 (left of the IP cloud) is 408-555-xxxx, where xxxx identifies the individual dial peers by extension. The destination pattern (or expanded telephone number) associated with Catalyst 4224 2 (right of the IP cloud) is 729-555-xxxx.

Figure 6-2 Sample VoIP Network

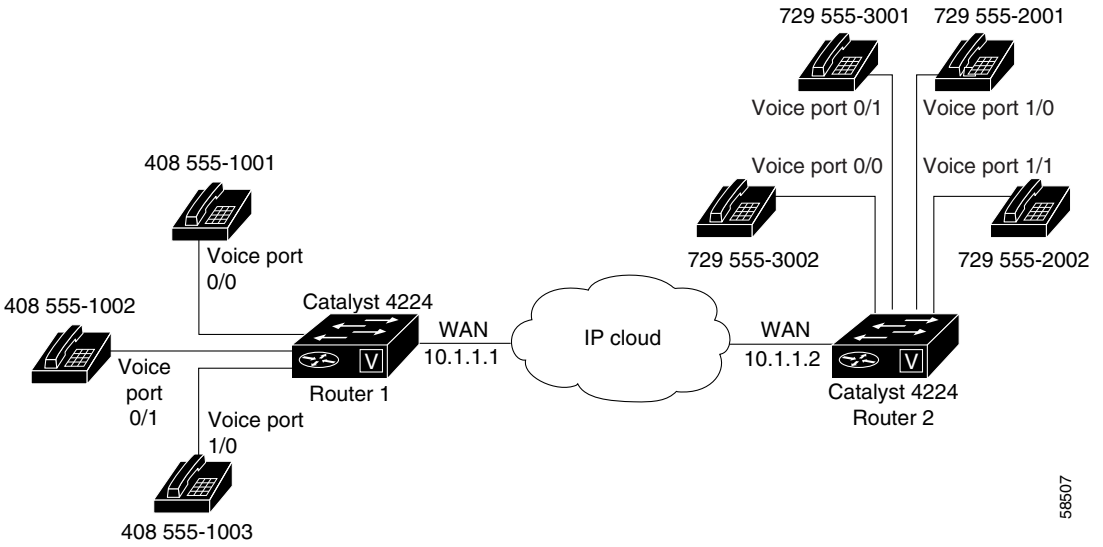


Table 6-1 shows the number expansion table for this scenario.

Table 6-1 Sample Number Expansion Table

Extension	Destination Pattern	Num-Exp Command Entry	Description
1...	14085551...	`num-exp 1... 14085551...`	Expands a four-digit extension beginning with the numeral 1 by prefixing 1408555 to the extension.
2...	17295552...	`num-exp 2... 17295552...`	Expands a four-digit extension beginning with the numeral 2 by prefixing 1408555 to the extension.
3...	17295553...	`num-exp 3... 17295553...`	Expands a four-digit extension beginning with the numeral 3 by prefixing 1408555 to the extension.

Note You can use a period (.) to represent variables, such as extension numbers, in a telephone number. A period is similar to a wildcard, which matches any entered digit.

The information included in this example needs to be configured on both Catalyst 4224 1 and Catalyst 4224 2. In this configuration, Catalyst 4224 1 can call any number string that begins with the digits *17295552* or *17295553* to connect to Catalyst 4224 2. Catalyst 4224 2 can call any number string that begins with the digits *14085551* to connect to Catalyst 4224 1.

Configure Number Expansion

To define how to expand an extension number into a particular destination pattern, use the following global configuration command:

num-exp *extension-number extension-string*

*extension-number represents the d*igit(s) defining an extension number to be expanded, whereas *extension-string represents the d*igit(s) defining an extension string to be expanded.

Use the **show num-exp** command to verify that you have mapped the telephone numbers correctly.

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

Configure Dial Peers

The key to understanding how VoIP functions is to understand dial peers. All of the voice technologies use dial peers to define the characteristics associated with a call leg. Dial peers are used to apply specific attributes to call legs and to identify call origin and destination.

As shown in Figure 6-3 and Figure 6-4, a *call leg* is a discrete segment of a call connection that lies between two points in the connection. Each call leg corresponds to a dial peer. An end-to-end call consists of four call legs, two from the perspective of the source Catalyst 4224 (Figure 6-3) and two from the perspective of the destination Catalyst 4224 (Figure 6-4). Attributes applied to a call leg include QoS, Coder-Decoder (codec), voice activity detection (VAD), and fax rate.

Figure 6-3 Dial Peer Call Legs, from the Perspective of the Source Catalyst 4224

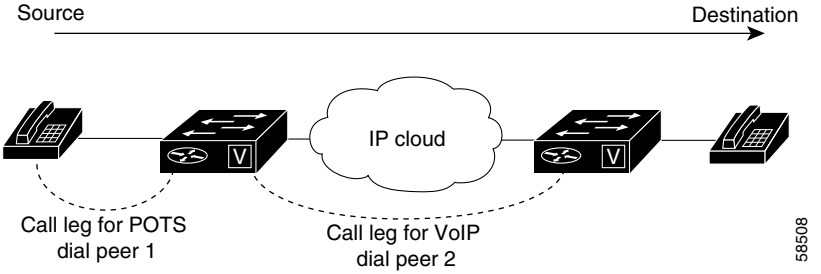
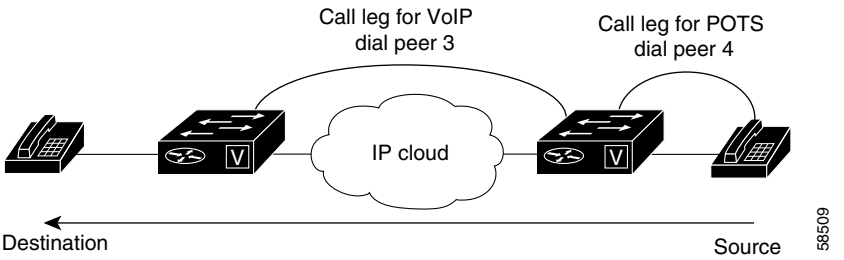


Figure 6-4 Dial Peer Call Legs, from the Perspective of the Destination Catalyst 4224



There are two types of dial peers with each voice implementation:

- POTS (plain old telephone service, or basic telephone service)—The dial peer associates a physical voice port with a local telephone device. The key commands you need to configure are the **destination-pattern** and **port** commands. The **destination-pattern** command defines the telephone number associated with the POTS dial peer. The **port** command associates the POTS dial peer with a specific logical dial interface, normally the voice port connecting your router to the local POTS network.
- VoIP—The dial peer associates a telephone number with an IP address. The key commands you need to configure are the **destination-pattern** and **session target** commands. The **destination-pattern** command defines the telephone number associated with the VoIP dial peer. The **session target** command specifies a destination IP address for the VoIP dial peer. In

addition, you can use VoIP dial peers to define characteristics such as IP precedence, additional QoS parameters (when RSVP is configured), codec, and VAD.

Inbound Versus Outbound Dial Peers

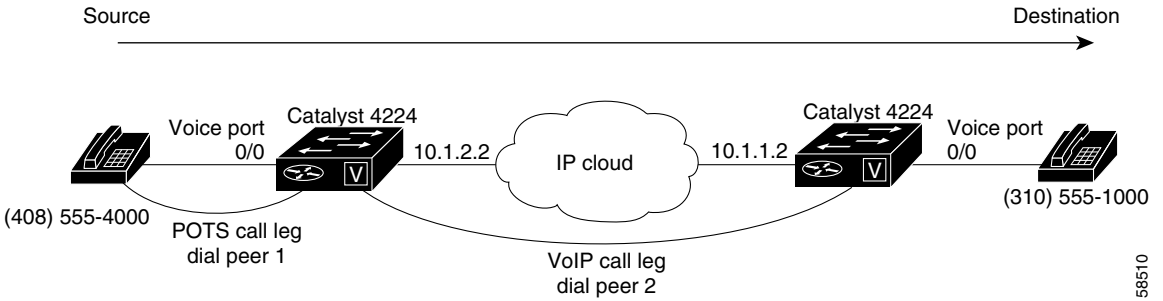
Dial peers are used for both inbound and outbound call legs. The terms *inbound* and *outbound* are defined from the *router* perspective. An inbound call leg means that an incoming call comes *to* the router. An outbound call leg means that an outgoing call is placed *from* the router.

For inbound call legs, a dial peer might be associated with the calling number or the voice-port number. Outbound call legs always have an associated dial peer. The destination pattern is used to identify the outbound dial peer. The call is associated with the outbound dial peer at setup time.

A POTS dial peer associates 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. VoIP dial peers point to specific devices (by associating destination telephone numbers with a specific IP address) so that incoming calls can be received and outgoing calls can be placed. To establish VoIP connections, you need both POTS and VoIP dial peers.

Establishing communication using VoIP is similar to configuring an IP static route; you are establishing a specific voice connection between two defined endpoints. As shown in Figure 6-5, the POTS dial peer establishes the source (via the originating telephone number or voice port) of the call. The VoIP dial peer establishes the destination by associating the destination telephone number with a specific IP address.

Figure 6-5 Outgoing Calls, from the Perspective of POTS Dial Peer 1



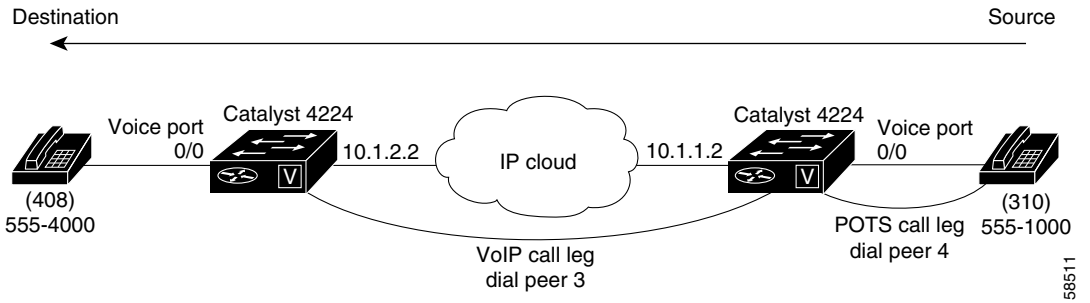
To configure call connectivity between the source and the destination as illustrated in Figure 6-5, enter the following commands on Catalyst 4224 10.1.2.2:

```
c4224(config)# dial-peer voice 1 pots
c4224(config-dial-peer)# destination-pattern 14085554000
c4224(config-dial-peer)# port 0/0

c4224(config)# dial-peer voice 2 voip
c4224(config-dial-peer)# destination-pattern 13105551000
c4224(config-dial-peer)# session target ipv4:10.1.1.2
```

Figure 6-6 shows how to complete the end-to-end call between dial peer 1 and dial peer 4.

Figure 6-6 Outgoing Calls, from the Perspective of POTS Dial Peer 2



To complete the end-to-end call between dial peer 1 and dial peer 4, as illustrated in Figure 6-6, enter the following commands on Catalyst 4224 10.1.1.2:

```
c4224(config)# dial-peer voice 4 pots
c4224(config-dial-peer)# destination-pattern 13105551000
c4224(config-dial-peer)# port 0/0

c4224(config)# dial-peer voice 3 voip
c4224(config-dial-peer)# destination-pattern 14085554000
c4224(config-dial-peer)# session target ipv4:10.1.2.2
```

Create a Dial-Peer Configuration Table

Each dial peer needs to be identified by its unique data before you can configure dial peers in VoIP. Do this is by creating a dial peer configuration table.

Using the example in Figure 6-2 on page 6-13, Router 1, with an IP address of 10.1.1.1, connects a small sales branch office to the main office through Router 2. There are three telephones in the sales branch office that need to be established as dial peers. Router 2, with an IP address of 10.1.1.2, is the primary gateway to the main office. There are four devices that need to be established as dial peers in the main office, and all are basic telephones connected to the PBX. Figure 6-2 on page 6-13 shows a diagram of this small voice network, and Table 6-2 shows the dial peer configuration table for the example in the figure.

Table 6-2 Dial Peer Example Configuration Table

		Commands				
Router	**Dial Peer Tag**	**Destination-Pattern**	**Type**	**Session Target**	**codec**	**QoS**
Cisco 1750 Router 1	10	1729555....	VoIP	IPV4 10.1.1.2	G.729	Best effort
Cisco 1750 Router 2	11	1408555....	VoIP	IPV4 10.1.1.1	G.729	Best effort

Configure POTS Dial Peers

POTS dial peers enable incoming calls to be received by a particular telephony device. To configure a POTS dial peer, you need to uniquely identify the dial peer (by assigning it a unique tag number), define its telephone number(s), and associate it with a voice port through which calls are established. Under most circumstances, the default values for the remaining dial peer configuration commands are sufficient to establish connections.

To enter the dial peer configuration mode (and select POTS as the method of voice-related encapsulation), use the **dial-peer voice** *number* **pots** global configuration command. The *number* value is a tag that uniquely identifies the dial peer. (This number has local significance only.)

To configure the identified POTS dial peer, use the **destination-pattern** *string* dial peer configuration command. The *string* value is the destination telephone number associated with this POTS dial peer.

Outbound Dialing on POTS Dial Peers

When a router receives a voice call, the router selects an outbound dial peer by comparing the called number with the number configured as the destination pattern for the POTS dial peer. The router then removes the left-justified numbers corresponding to the destination pattern that matches the called number. If you have configured a prefix, the prefix is put in front of the remaining numbers, creating a dial string, which the router then dials. If all numbers in the destination pattern are removed, the user receives a dial tone.

For example, suppose there is a voice call with the E.164 called number of *1-310-767-2222*. If you configure a destination-pattern of *1310767* and a prefix of *9*, the router removes *1310767* from the E.164 telephone number, leaving the extension number of *2222*. It will then create a prefix *9* at the front of the remaining numbers, so that the actual numbers dialed are *9,2222*. The comma in this example means that the router will pause for one second between dialing the *9* and the *2* to allow for a secondary dial tone.

Configure VoIP Dial Peers

VoIP dial peers enable outgoing calls to be made from a particular telephony device. To configure a VoIP dial peer, you need to identify the dial peer (by assigning it a unique tag number), define its destination telephone number, and define its destination IP address. The default values for the remaining dial peer configuration commands are usually adequate to establish connections.

To enter the dial peer configuration mode (and select VoIP as the method of voice-related encapsulation), use the **dial-peer voice** *number* **voip** global configuration command. The *number* value of the **dial-peer voice voip** command is a tag that uniquely identifies the dial peer.

To configure the identified VoIP dial peer, use the following dial peer configuration commands:

	Task	Command
Step 1	Define the destination telephone number associated with this VoIP dial peer.	**destination-pattern** *string*
Step 2	Specify a destination IP address for this dial peer.	**session target** {**ipv4:***destination-address* | **dns:***host-name*}

For examples of how to configure dial peers, see Appendix C, "VoIP Configuration Examples."

Verifying Your Configuration

You can check the validity of your dial peer 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. Use this command to display a specific dial peer or to display all configured dial peers.
- Use the **show dialplan number** command to show which dial peer is reached when a particular number is dialed.

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:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the "Configuring IP" section in the *Network Protocols Configuration Guide*, Part 1 for Cisco IOS Release 12.0 T.
- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.
- If you have configured a codec value, there can be a problem if the VoIP dial peers on either side of the connection have incompatible codec values. Make sure that both VoIP peers have been configured with the same codec value.
- Use the **debug vpm spi** command to verify that the router dials the correct output string.
- Use the **debug cch323 rtp** command to check RTP packet transport.
- Use the **debug cch323 h225** command to check the call setup.

Configure Voice Ports

Routers provide only analog voice ports for its implementation of VoIP. The type of signaling associated with analog voice ports depends on the voice interface card (VIC) installed in the device.

Each VIC is specific to a particular signaling type; therefore, VICs determine the type of signaling for the voice ports. Voice-port commands define the characteristics associated with a particular voice-port signaling type.

The voice ports support three basic voice signaling types:

- FXS—The foreign exchange station interface uses a standard RJ-11 modular telephone cable to connect directly to a standard telephone, fax machine, PBX or similar device. FXS supplies ring, voltage, and dial tone to the station.

- FXO—The foreign exchange office interface uses a RJ-11 modular telephone cable to connect local calls to a PSTN central office or to a PBX that does not support E&M signaling. This interface is used for off-premise extension applications.
- E&M—The E&M interface uses an RJ-48 telephone cable to connect remote calls from an IP network to PBX trunk lines (tie lines) for local distribution. It is a signaling technique for two-wire and four-wire telephone and trunk interfaces.

Configure FXS or FXO Voice Ports

Under most circumstances, the default voice port values are adequate to configure FXS and FXO ports to transport voice data over your existing IP network. However, if you need to change the default configuration for these voice ports, use the commands outlined in Table 6-3, beginning in privileged EXEC mode:

Table 6-3 Commands to Configure FXS and FXO Voice Ports

	Task	Required or Optional	Command
Step 1	Enter the global configuration mode.	Required	`configure terminal`
Step 2	Identify the voice port you want to configure, and enter the voice port configuration mode.	Required	`voice-port` *slot-number/port*
Step 3	(For FXO ports only) Select the appropriate dial type for out-dialing.	Required	`dial-type` {`dtmf` | `pulse`}
Step 4	Select the appropriate signal type for this interface.	Required	`signal` {`loop-start` | `ground-start`}
Step 5	Select the appropriate voice call progress tone for this interface. The default for this command is **us**.	Required	`cptone` *country*
Step 6	(For FXS ports only) Select the ring frequency (in Hz) specific to the equipment attached to this voice port and appropriate to the country you are in.	Required	`ring frequency` {`25` | `50`}

Table 6-3 Commands to Configure FXS and FXO Voice Ports (continued)

	Task	Required or Optional	Command
Step 7	(For FXO ports only) Specify the maximum number of rings before answering a call.	Required	**ring number** *number*
Step 8	Specify the private line auto ringdown (PLAR) connection if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number.	Optional	**connection plar** *string*
Step 9	Specify the threshold (in dB) for on-hold music. Valid entries are from –70 to –30 db.	Optional	**music-threshold** *number*
Step 10	Attach descriptive text about this voice port connection.	Optional	**description** *string*
Step 11	If voice activity detection (VAD) is activated, specify that background noise is generated.	Optional	**comfort-noise**

Verifying Your Configuration

You can check the validity of your voice port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and listen for a dial tone.
- Check for dual tone multifrequency (DTMF) detection. If the dial tone stops when you dial a digit, the voice port is configured properly.
- Use the **show voice-port** command to verify that the configured data is correct.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with the 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 ping the destination, refer to the *Network Protocols Configuration Guide, Part 1* for Cisco IOS Release 12.0T.
- Use the **show voice-port** command to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- Make sure the VICs are correctly installed. For more information about installing a VIC in your router, refer to the *Cisco WAN Interface Cards Hardware Installation Guide*.

Fine-Tune FXS and FXO Voice Ports

In most cases, the default values for voice-port tuning commands are sufficient. Depending on the specifics of your particular network, you might need to adjust voice parameters involving timing, input gain, and output attenuation for FXS or FXO voice ports. Collectively, these commands are referred to as voice-port tuning commands.

If you need to change the default tuning configuration for FXS and FXO voice ports, use the commands outlined in Table 6-4, beginning in privileged EXEC mode:

Table 6-4 Commands to Fine Tune FXS and FXO Voice Ports

	Task	Command	Valid Entries	Default Value
Step 1	Enter the global configuration mode.	`configure terminal`		
Step 2	Identify the voice port you want to configure, and enter the voice port configuration mode.	`voice-port` *slot-number/port*		

Table 6-4 Commands to Fine Tune FXS and FXO Voice Ports (continued)

	Task	Command	Valid Entries	Default Value
Step 3	Specify (in dB) the amount of gain to be inserted at the receiver side of the interface.	**input gain** *value*	–6 to 14 dB	0 dB
Step 4	Specify (in dB) the amount of attenuation at the transmit side of the interface.	**output attenuation** *value*	0 to 14 dB	0 dB
Step 5	Enable echo-cancellation of voice that is sent out of the interface and received back on the same interface.	**echo-cancel enable**		
Step 6	Adjust the size (in milliseconds) of the echo-cancel.	**echo-cancel coverage** *value*	8, 16, 24, and 32 ms	16 ms
Step 7	Enable nonlinear processing, which shuts off any signal if no near-end speech is detected. (Nonlinear processing is used with echo-cancellation.)	**non-linear**		
Step 8	Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits.	**timeouts initial** *seconds*	0 to 120 sec	10 sec
Step 9	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.	**timeouts interdigit** *seconds*	0 to 120 sec	10 sec
Step 10	If the voice-port dial type is DTMF, configure the DTMF digit signal duration.	**timing digit** *milliseconds*	50 to 100 ms	100 ms
Step 11	If the voice-port dial type is DTMF, configure the DTMF inter-digit signal duration.	**timing inter-digit** *milliseconds*	50 to 500 ms	100 ms

Table 6-4 Commands to Fine Tune FXS and FXO Voice Ports (continued)

	Task	Command	Valid Entries	Default Value
Step 12	(FXO ports only) If the voice-port dial type is pulse, configure the pulse digit signal duration.	**timing pulse-digit** *milliseconds*	10 to 20 ms	20 ms
Step 13	(FXO ports only) If the voice-port dial type is pulse, configure the pulse inter-digit signal duration.	**timing pulse-inter-digit** *milliseconds*	100 to 1000 ms	500 ms

Note After you change any voice port setting, Cisco recommends that you cycle the port by using the **shutdown** and **no shutdown** commands.

Configure E&M Voice Ports

Unlike FXS and FXO voice ports, the default E&M voice-port parameters are not sufficient to enable voice and data transmission over your IP network. Because of the inherent complexities of PBX networks, E&M voice port values must match those specified by the particular PBX device to which it is connected.

To configure E&M voice ports, use the commands outlined in Table 6-5, beginning in privileged EXEC mode:

Table 6-5 Commands to Configure E&M Voice Ports

	Task	Required or Optional	Command
Step 1	Enter the global configuration mode.	Required	**configure terminal**
Step 2	Identify the voice port you want to configure, and enter the voice port configuration mode.	Required	**voice-port** *slot-number/port*
Step 3	Select the appropriate dial type for out-dialing.	Required	**dial-type** {**dtmf** | **pulse**}

Table 6-5 Commands to Configure E&M Voice Ports (continued)

	Task	Required or Optional	Command
Step 4	Select the appropriate signal type for this interface.	Required	`signal {wink-start | immediate | delay-dial}`
Step 5	Select the appropriate voice call progress tone for this interface.	Required	`cptone {australia | brazil | china | finland | france | germany | japan | northamerica | unitedkingdom}`
Step 6	Select the appropriate cabling scheme for this voice port.	Required	`operation {2-wire | 4-wire}`
Step 7	Select the appropriate E&M interface type. • Type 1 is for the following lead configuration: E—Output, relay to ground M—Input, referenced to ground • Type 2 is for the following lead configuration: E—Output, relay to SG M—Input, referenced to ground SB—Feed for M, connected to –48V SG—Return for E, galvanically isolated from ground • Type 3 is for the following lead configuration: E—Output, relay to ground M—Input, referenced to ground SB—Connected to –48V SG—Connected to ground • Type 5 is for the following lead configuration: E—Output, relay to ground M—Input, referenced to –48V	Required	`type {1 | 2 | 3 | 5}`

Table 6-5 Commands to Configure E&M Voice Ports (continued)

	Task	Required or Optional	Command
Step 8	Specify a terminating impedance for an E&M voice port. The impedance value selected must match the specifications from the telephony system to which this voice port is connected.	Required	**impedance** {**600c** | **600r** | **900c** | **complex1** | **complex2**}
Step 9	Specify the private line auto ringdown (PLAR) connection if this voice port is used for a PLAR connection. The *string* value specifies the destination telephone number.	Optional	**connection plar** *string*
Step 10	Specify the threshold (in dB) for on-hold music. Valid entries are from –70 to –30 dB. The default is –38 dB.	Optional	**music-threshold** *number*
Step 11	Attach descriptive text about this voice port connection.	Optional	**description** *string*
Step 12	Specify that background noise is generated.	Optional	**comfort-noise**

Verifying Your Configuration

You can check the validity of your voice port configuration by performing the following tasks:

- Pick up the handset of an attached telephony device and listen for a dial tone.
- Check for DTMF detection. If the dial tone stops when you dial a digit, the voice port is configured properly.
- Use the **show voice-port** command to verify that the data configured is correct.

Troubleshooting Tips

If you are having trouble connecting a call and you suspect the problem is associated with the 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 ping the destination, refer to the *Network Protocols Configuration Guide, Part 1* for Cisco IOS Release 12.0 T.
- Use the **show voice-port command** to make sure that the port is enabled. If the port is offline, use the **no shutdown** command.
- If you have configured E&M interfaces, make sure that the values pertaining to your specific PBX setup, such as timing and type, are correct.
- Make sure the VICs are correctly installed. For more information, refer to the *Cisco WAN Interface Cards Hardware Installation Guide*.

Fine-Tune E&M Voice Ports

In most cases, the default values for voice port tuning commands are sufficient. Depending on the specifics of your particular network, you might need to adjust voice parameters involving timing, input gain, and output attenuation for E&M voice ports. Collectively, these commands are referred to as voice port tuning commands.

If you need to change the default tuning configuration for E&M voice ports, use the commands shown in Table 6-6, beginning in privileged EXEC mode:

Table 6-6 Commands to Fine Tune E&M Voice Ports

	Task	Command	Valid Entries	Default Values
Step 1	Enter the global configuration mode.	`configure terminal`		
Step 2	Identify the voice port you want to configure, and enter the voice port configuration mode.	`voice-port` *slot-number/port*		

Table 6-6 Commands to Fine Tune E&M Voice Ports (continued)

	Task	Command	Valid Entries	Default Values
Step 3	Specify (in dB) the amount of gain to be inserted at the receiver side of the interface.	**input gain** *value*	–6 to 14 dB	0 dB
Step 4	Specify (in dB) the amount of attenuation at the transmit side of the interface.	**output attenuation** *value*	0 to 14 dB	0 dB
Step 5	Enable echo-cancellation of voice that is sent out of the interface and received back on the same interface.	**echo-cancel enable**		
Step 6	Adjust the size (in milliseconds) of the echo-cancel.	**echo-cancel coverage** *value*	8, 16, 24, and 32 ms	16 ms
Step 7	Enable nonlinear processing, which shuts off any signal if no near-end speech is detected. (Nonlinear processing is used with echo-cancellation.)	**non-linear**		
Step 8	Specify the number of seconds the system will wait for the caller to input the first digit of the dialed digits.	**timeouts initial** *seconds*	0 to 120 sec	10 sec

Table 6-6 Commands to Fine Tune E&M Voice Ports (continued)

	Task	Command	Valid Entries	Default Values
Step 9	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.	**timeouts interdigit** *seconds*	0 to 120 sec	10 sec
Step 10	Specify the indicated timing parameters.	**timing clear-wait** *milliseconds* **timing delay-duration** *milliseconds* **timing delay-start** *milliseconds* **timing dial-pulse min-delay** *milliseconds* **timing digit** *milliseconds* **timing inter-digit** *milliseconds* **timing pulse** *pulses-per-second* **timing pulse-inter-digit** *milliseconds* **timing wink-duration** *milliseconds* **timing wink-wait** *milliseconds*	200 to 2000 ms 100 to 5000 ms 20 to 2000 ms 0 to 5000 ms 50 to 100 ms 50 to 500 ms 10 to 20 pps 100 to 1000 ms 100 to 400 ms 100 to 5000 ms	

Note After you change any voice port setting, Cisco recommends that you cycle the port by using the **shutdown** and **no shutdown** commands.

Additional VoIP Dial-Peer Configurations

Depending on how you have configured your network interfaces, you might need to configure additional VoIP dial-peer parameters. This section describes the following topics:

Configure IP Precedence for Dial Peers

Use the **ip precedence** command to give voice packets a higher priority than other IP data traffic. The **ip precedence** command should also be used if RSVP is not enabled and you would like to give voice packets a priority over other IP data traffic. IP precedence scales better than RSVP but provides no admission control.

To give real-time voice traffic precedence over other IP network traffic, use the following global configuration commands:

Table 6-7 Global Configuration Commands for Dial Peers

	Task	Command
Step 1	Enter dial peer configuration mode to configure a VoIP dial peer. The *number* argument is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647.	`dial-peer voice` *number* `voip`
Step 2	Select a precedence level for the voice traffic associated with that dial peer. The *number* argument specifies the IP Precedence value. Valid entries are from 0 to 7. A value of 0 means that no precedence (priority) has been set.	`ip precedence` *number*

In IP precedence, the numbers 1 through 5 identify classes for IP flows, while the numbers 6 and 7 are used for network and backbone routing and updates.

For example, to ensure that voice traffic associated with VoIP dial peer 103 is given a higher priority than other IP network traffic, enter the following:

```
c4224(config)# dial-peer voice 103 voip
c4224(config-dial-peer)# ip precedence 5
```

In this example, when an IP call leg is associated with VoIP dial peer 103, all packets transmitted to the IP network via this dial peer will have their precedence bits set to 5. If the networks receiving these packets have been configured to recognize precedence bits, the packets are given priority over packets with a lower configured precedence value.

Configure RSVP for Dial Peers

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

For example, to configure controlled-load QoS for VoIP dial peer 108, enter the following global configuration commands:

```
c4224(config)# Dial-peer voice 108 voip
c4224(config-dial-peer)# req-qos controlled-load
c4224(config-dial-peer)# session target ipv4:10.0.0.8
```

In this example, every time a connection is made through VoIP dial peer 108, an RSVP reservation request is made between the local router, all intermediate routers in the path, and the final destination router.

Note Cisco recommends that you select **controlled-load** for the requested QoS. The controlled-load service uses admission (or capacity) control to ensure that preferential service is provided even when the bandwidth is overloaded.

To generate Simple Network Management Protocol (SNMP), use the following commands beginning in global configuration mode:

Table 6-8 Configuration Commands to Generate SNMP

	Task	Command
Step 1	Enter the dial peer configuration mode to configure a VoIP dial peer.	**dial-peer voice** *number* **voip**
Step 2	Generate an SNMP event if the QoS for a dial peer drops below a specified level.	**acc-qos** [**best-effort** | **controlled-load** | **guaranteed-delay**]

Note RSVP reservations are only one-way. If you configure RSVP, the VoIP dial peers on either side of the connection must be configured for RSVP.

Configure codec and VAD for Dial Peers

Coder-decoder (codec) typically is used to transform analog signals into a digital bit stream and digital signals back into analog signals. The codec specifies the voice coder rate of speech for a dial peer. Voice activity detection (VAD) is used to disable the transmission of silence packets. codec and VAD values for a dial peer determine how much bandwidth the voice session uses.

Configure codec for a VoIP Dial Peer

To specify a voice coder rate for a selected VoIP dial peer, use the following commands, beginning in global configuration mode:

Table 6-9 Configuration Commands to Specify a Voice Coder Rate

	Task	Command
Step 1	Enter the dial peer configuration mode to configure a VoIP dial peer.	**dial-peer voice** *number* **voip**
Step 2	Specify the desired voice coder rate of speech.	**codec** [**g711alaw** | **g711ulaw** | **g729r8** | **g729r8 pre-ietf**]

The default for the **codec** command is **g729r8**, which is normally the most desirable setting. However, if you are operating on a high-bandwidth network and voice quality is of the highest importance, you should configure the **codec** command for **g711alaw** or **ulaw**. Using **g711alaw results** in better voice quality, but it also requires higher bandwidth usage for voice.

For example, to specify a codec rate of **g711alaw** for VoIP dial peer 108, enter the following:

```
c4224(config)# dial-peer voice 108 voip
c4224(config-dial-peer)# codec g711alaw
```

Note Prior to Cisco IOS Release 12.0(5)T, **g729r8** is implemented in the pre-IETF format; thereafter, it is implemented in the standard IETF format. When new images, such as Release 12.0(5)T or later, interoperate with older versions of VoIP (when the **g729r8** codec was not compliant with the IETF standard), users can hear garbled voices and ringback on either side of the connection. To avoid this problem, configure the dial peers with the **g729r8 pre-ietf** argument.

Configure VAD for a VoIP Dial Peer

To disable the transmission of silence packets and enable VAD for a selected VoIP dial peer, use the following global configuration commands:

Table 6-10 Commands to Disable the Transmission of Silence Packets

	Task	Command
Step 1	Enter dial peer configuration mode to configure a VoIP dial peer.	**dial-peer voice** *number* **voip**
Step 2	Disable the transmission of silence packets.	**vad**

The **vad** command is enabled by default, which is normally the most desirable setting. If you are operating on a high-bandwidth network and voice quality is of the highest importance, you should disable VAD. Disabling VAD results in better voice quality, but it also requires higher bandwidth usage for voice.

For example, to enable VAD for VoIP dial peer 108, enter the following:

```
c4224(config)# Dial-peer voice 108 voip
c4224(config-dial-peer)# vad
```

Configure Frame Relay for VoIP

When you are configuring VoIP, you need to take certain factors into consideration so that it runs smoothly over Frame Relay. A public Frame Relay cloud provides no QoS guarantee. For real-time traffic to be transmitted in a timely manner, the data rate must not exceed the committed information rate (CIR), or packets might be dropped. In addition, Frame Relay traffic shaping and RSVP are mutually exclusive. This is particularly important to remember if multiple data-link connection identifiers (DLCIs) are carried on a single interface.

For Frame Relay links with slow output rates (less than or equal to 64 kbps), where data and voice are transmitted over the same permanent virtual circuit (PVC), Cisco recommends the following solutions:

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

Note Lowering the MTU size impacts data throughput speed.

- Set CIR equal to line rate—Ensure that the data rate does not exceed the CIR through generic traffic shaping.
  - Use RSVP or IP precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet header size.

- Shape the traffic—Use adaptive traffic shaping to slow the output rate based on the backward explicit congestion notification (BECN). If the feedback from the switch is ignored, packets (both data and voice) might be discarded. Because the Frame Relay switch does not distinguish between voice and data packets, voice packets could be discarded, which would result in a deterioration of voice quality.
  - Use RSVP, compressed RTP, reduced MTU size, and adaptive traffic shaping based on BECN to hold the data rate to CIR.
  - Use generic traffic shaping to obtain a low interpacket wait time. For example, set committed burst (Bc) to 4000 to obtain an interpacket wait time of 125 milliseconds.

In Cisco IOS Release 12.0 T, Frame Relay traffic shaping is not compatible with RSVP. Cisco suggests one of the following workarounds:

- Provision the Frame Relay permanent virtual circuits (PVC) to have the CIR equal to the port speed.
- Use generic traffic shaping with RSVP.

Frame Relay for VoIP Configuration Example

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

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

interface Serial1 point-to-point
 mtu 300
 ip address 40.0.0.7 255.0.0.0
 ip rsvp bandwidth 48 48
 no ip route-cache
 no ip mroute-cache
 bandwidth 64
```

```
traffic-shape rate 32000 4000 4000
frame-relay interface-dlci 16
frame-relay ip rtp header-compression
```

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

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

The subinterface is configured as follows:

- MTU size is inherited from the main interface.
- IP address for the subinterface is specified.
- RSVP is enabled to use the default value, which is 75 percent of the configured bandwidth.
- Bandwidth is set to 64 kbps.
- Generic traffic shaping is enabled with 32-kbps CIR where committed burst (Bc) = 4000 bits and excess burst (Be) = 4000 bits.
- Frame Relay DLCI number is specified.
- IP RTP header compression is enabled.

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

For more information about configuring Frame Relay for VoIP, refer to the "Configuring Frame Relay" section in the *Wide-Area Networking Configuration Guide* for Cisco IOS Release 12.0 T.

Configuring the Eight-Port FXS RJ-21 Module

The Eight-port RJ-21 FXS Module is a high-density analog phone and fax relay interface. By providing service to analog phones and fax machines, the eight Foreign Exchange Station (FXS) ports emulate a Public Switched Telephone Network (PSTN) central office (CO) or private branch exchange (PBX).

This section describes how to configure the eight-port FXS module on the Catalyst 4224. This section contains the following topics:

Eight-Port RJ-21 FXS Module User Interface Conventions

The eight-port Foreign Exchange Station (FXS) module is similar to the two-port FXS analog interface card (VIC-2FXS). Because the eight-port FXS module is located in slot 4, the eight ports are numbered 4/0 to 4/7. The front panel of this module has two rows of four LEDs each. The LEDs numbered 0 through 3 on the left side represent ports 4/0 through 4/3, and the LEDs numbered 4 through 7 on the right side represent ports 4/4 through 4/7.

Configuring FXS Voice Ports

The default values for FXS voice ports are usually adequate but this section includes information on changing the defaults if necessary. This section describes the following procedures:

Changing Default Configurations

To configure FXS voice ports, use the following commands in privileged EXEC mode:

Command	Purpose
`configure terminal`	Enters global configuration mode.
`voice-port` *`slot-number/port`*	Identifies the voice slot and port number you want to configure, and entesr voice port configuration mode.
`signal` {**`loop-start`** | **`ground-start`**}	Selects the appropriate signal type for this interface.

Command	Purpose
cptone *country*	Selects the appropriate voice call progress tone for this interface. The default for this command is **us**. For a list of supported countries, refer to the *Voice, Video, and Home Applications Command Reference*.
ring frequency {**25** | **50**}	Selects the appropriate ring frequency (in Hertz) specific to the equipment attached to this voice port.
connection plar *string*	(Optional) Specifies the PLAR[1] connection if this voice port is used for a PLAR connection. The *string* value is any series of digits that specifies the destination E.164 telephone number.
music-threshold *number*	(Optional) Specifies the threshold (in decibels) for music on hold. Valid entries are from –70 to –30.
description *string*	(Optional) Attaches descriptive text about this voice port connection.
comfort-noise	(Optional) Specifies that background noise will be generated.

1. Private line automatic ringdown

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

The following example shows how to use the FXS configuration commands:

```
Gateway# conf t

Enter configuration commands, one per line.  End with CNTL/Z.

Gateway(config)# voice-port 4/0
Gateway(config-voiceport)# signal loopStart
Gateway(config-voiceport)# cptone IN
Gateway(config-voiceport)# ring frequency 50
Gateway(config-voiceport)# connection plar 5265761
Gateway(config-voiceport)# music-threshold -50
Gateway(config-voiceport)# description "Connection to PBX"
Gateway(config-voiceport)# comfort-noise
```

To display the values configured, use the **show running-config** command.

Validating the Configuration

To validate your voice port configuration, perform one or both of the following tasks:

- Pick up the handset of a telephony device attached to your IP network and check for a dial tone. The corresponding LED turns green to indicate "off-hook" and "call-in-progress" conditions. If the dial tone stops when you dial a digit, then the voice port is probably configured properly.
- To confirm that the data is configured correctly, use the **show voice port** command as follows:

```
Gateway# sh voice port 4/0

Foreign Exchange Station 4/0 Slot is 4, Port is 0
 Type of VoicePort is FXS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is "Connection to PBX"
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Music On Hold Threshold is Set to -50 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 8 ms
 Playout-delay Mode is set to default
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Connection Mode is plar
 Connection Number is 5265761
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Ringing Time Out is set to 180 s
 Companding Type is u-law
 Region Tone is set for IN
Analog Info Follows:
 Currently processing none
 Maintenance Mode Set to None (not in mtc mode)
 Number of signaling protocol errors are 0
 Impedance is set to 600r Ohm
 Wait Release Time Out is 30 s
 Station name None, Station number None
Voice card specific Info Follows:
 Signal Type is loopStart
```

```
Ring Frequency is 50 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
Ring Cadence is defined by CPTone Selection
Ring Cadence are [4 2] [4 20] * 100 msec
```

Troubleshooting the Configuration

If you are having trouble placing a call and you suspect the problem is associated with the voice port configuration, you might be able to resolve the problem by performing one or more of the following tasks:

- Ping the associated IP address to confirm connectivity, as follows:

```
Gateway# ping 172.20.59.93

Type escape sequence to abort.
Sending 5, 100-byte ICMP Echos to 172.20.59.93, timeout is 2
seconds:
!!!!!
Success rate is 100 percent (5/5), round-trip min/avg/max = 1/1/4
ms
```

  If you cannot successfully ping your destination, refer to the *Network Protocols Configuration Guide, Part 1*.

- Use the **show voice port** command to ensure that the port is enabled (administrative state is UP), as follows:

```
Gateway# sh voice port 4/0

Operation State is DORMANT
Administrative State is UP
```

  If the port state is DOWN, as in the following display, use the **no shutdown** command to enable the port. (See the "Activating the Voice Port" section on page 7-8.)

```
Operation State is DOWN
Administrative State is DOWN
```

Fine-Tuning FXS Voice Ports

Depending on the specifics of your particular network, you might need to fine-tune the FXS voice port settings. Under most circumstances, the default values will suffice; however, if you need to change them, use the following commands in privileged EXEC mode.

Command	Purpose
configure terminal	Enters global configuration mode.
voice-port *slot_number/port*	Identifies the voice slot and port number you want to configure, and enters voice port configuration mode.
input gain *value*	Specifies (in decibels) the amount of gain to be inserted at the receiver side of the interface. Acceptable values are from –6 to 14.
output attenuation *value*	Specifies (in decibels) the amount of attenuation at the transmit side of the interface. Acceptable values are from 0 to 14.
echo-cancel enable	Enables echo-cancellation of voice that is sent out through the interface and received back on the same interface.
echo-cancel coverage *value*	Adjusts the size (in milliseconds) of the echo-cancel. Acceptable value is 8.
impedance *value*	Specifies the impedance of the port. The functional values are 600r (the default) and complex2 (an 820 ohm in series with (220 ohm | 120 nF)).
non-linear	Enables nonlinear processing, which shuts off any signal if no near-end speech is detected. (Nonlinear processing is used with echo-cancellation.)
timeouts initial *seconds*	Specifies the number of seconds that the system will wait for the caller to enter the first digit of the digits to be dialed. Valid entries are from 0 to 120.
timeouts interdigit *seconds*	Specifies the number of seconds the system will wait (after the caller has entered the initial digit) for the caller to enter a subsequent digit. Valid entries are from 0 to 120.

Command	Purpose
timing digit *milliseconds*	If the voice port dial type is dual tone multifrequency (DTMF), configures the duration (in milliseconds) of the DTMF digit signal. The range of the duration is from 50 to 100; the default is 100.
timing inter-digit *milliseconds*	If the voice port dial type is DTMF, configures the duration (in milliseconds) of the DTMF interdigit signal. The range of the duration is from 50 to 500; the default is 100.

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

The following example shows how to use the fine-tune FXS commands:

```
Gateway# conf t

Enter configuration commands, one per line.  End with CNTL/Z.

Gateway(config)# voice-port 4/0
Gateway(config-voiceport)# input gain 10
Gateway(config-voiceport)# output attenuation 10
Gateway(config-voiceport)# echo-cancel enable
Gateway(config-voiceport)# echo-cancel coverage 8
Gateway(config-voiceport)# non-linear
Gateway(config-voiceport)# timeouts initial 10
Gateway(config-voiceport)# timeouts interdigit 10
Gateway(config-voiceport)# timing digit 60
Gateway(config-voiceport)# timing inter-digit 60
```

To display the values configured, use the **show running-config** command, as follows:

```
Gateway# sh running-config
!
voice-port 4/0
input gain 10
output attenuation 10
echo-cancel coverage 8
timeouts initial 10
timeouts interdigits 10
timing digit 60
timing inter-digit 60
!
```

Activating the Voice Port

By default, configured voice ports are active. However, if you need to activate a port because it has been shut down explicitly, use the **no shutdown** command, as follows:

```
Gateway# conf t

Enter configuration commands, one per line.  End with CNTL/Z.

Gateway(config)# voice-port 4/0
Gateway(config-voiceport)# no shutdown
Gateway(config-voiceport)#
00:55:53:%LINK-3-UPDOWN:Interface Foreign Exchange Station 4/0,
changed state to up
```

To deactivate a port, use the **shutdown** command, as follows:

```
Gateway# conf t

Enter configuration commands, one per line.  End with CNTL/Z.

Gateway(config)# voice-port 4/0
Gateway(config-voiceport)# shutdown
Gateway(config-voiceport)#
00:55:23:%LINK-3-UPDOWN:Interface Foreign Exchange Station 4/0,
changed state to Administrative Shutdown
```

Sample Configuration

This section provides a sample configuration for sending a fax or a call from the Cisco 2610 (a voice-enabled router) to the eight-port FXS module on a Catalyst 4224, and vice versa.

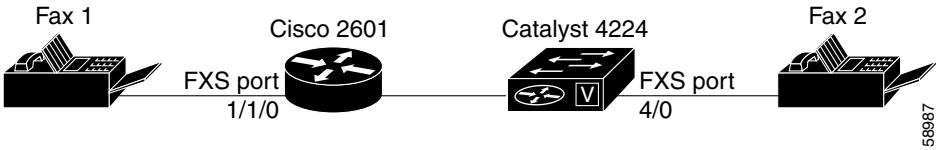
Note You can substitute any voice-enabled router for the Cisco 2610 and any Fast Ethernet interface connected to an IP network.

In the sample configuration illustrated in Figure 7-1, Fax1 is connected through an FXS port to the Cisco 2610 router. The router is connected through Fast Ethernet to the eight-port FXS module, which is connected through an FXS port to Fax2.

This section includes the following configuration-related samples:

- Cisco 2600 Sample Configuration, page 7-10
- FXS Module Sample Configuration, page 7-10
- Displaying Cisco 2600 Configuration Values, page 7-11
- Displaying FXS Module Configuration Values, page 7-12

Figure 7-1 Configuration for Connecting Faxs through Catalyst 4224



The following template can be used to implement this configuration:

Dial-Peer Tag[1]	Destination Pattern[2]	Type	Voice Port	Session Target[3]	codec
For Cisco 2610					
1	10	POTS	1/1/0	—	G.711 (default)
2	20	VoIP	—	172.20.59.93	G.729 (default)
For Eight-Port RJ-21 FXS Module					
1	20	POTS	4/0	—	G.711 (default)
2	10	VoIP	—	172.20.59.61	G.729 (default)

1. Assigns a unique number (1, 2,...) to a dial peer. Has only local significance.
2. Assigns phone numbers to dial peers. The router directs voice calls based on these patterns.

3. Identifies the remote end of the VoIP call, which can be specified using an IP address (as shown in the configuration) or a DNS name.

Cisco 2600 Sample Configuration

Using the configuration template shown, you could configure the Cisco 2600 as follows:

```
>[Configure the fast ethernet interface]
>2600# conf t

>Enter configuration commands, one per line.  End with CNTL/Z.

>2600(config)# interface FastEthernet0/0
>2600(config-if)# ip address 172.20.59.61 255.255.255.0
>
>[Configure the POTS call leg, as shown in the template above]
>2600(config-if)# dial-peer voice 1 pots
>2600(config-dial-peer)# destination-pattern 10
>2600(config-dial-peer)# port 1/1/0
>
>[Configure the VOIP call leg, as shown in the template above]
>2600(config-dial-peer)# dial-peer voice 2 voip
>2600(config-dial-peer)# destination-pattern 20
>2600(config-dial-peer)# session target ipv4:172.20.59.93
```

FXS Module Sample Configuration

Similarly, the eight-Port RJ-21 FXS module could be configured as follows:

```
>[Configure the fast ethernet interface]
>Gateway# conf t
>
>Enter configuration commands, one per line.  End with CNTL/Z.
>
>Gateway(config)# interface FastEthernet5/0
>Gateway(config-if)# ip address 172.20.59.93 255.255.0.0
>
>[Configure the POTS call leg as shown in the template above]
>Gateway(config-if)# dial-peer voice 1 pots
>Gateway(config-dial-peer)# destination-pattern 20
>Gateway(config-dial-peer)# port 4/0
>
>[Configure the VOIP call leg, as shown in the template above]
```

```
>Gateway(config-dial-peer)# dial-peer voice 2 voip
>Gateway(config-dial-peer)# destination-pattern 10
>Gateway(config-dial-peer)# session target ipv4:172.20.59.61
```

At this point, you should be able to send a fax or phone call from the Cisco 2600 to the FXS module, and vice versa.

Displaying Cisco 2600 Configuration Values

To display the values configured on the Cisco 2600, use the **show running-config** command, as follows:

```
>2600# sh running-config
>Building configuration...
>
>Current configuration :951 bytes
>!
>version 12.1
>no service single-slot-reload-enable
>service timestamps debug uptime
>service timestamps log uptime
>no service password-encryption
>!
>hostname 2600
>!
>no logging buffered
>no logging buffered
>logging rate-limit console 10 except errors
>!
>ip subnet-zero
>no ip finger
>!
>frame-relay switching
>no mgcp timer receive-rtcp
>!
>interface FastEthernet0/0
> ip address 172.20.59.61 255.255.255.0
> duplex auto
> speed auto
>!
>interface FastEthernet0/1
> ip address 192.100.1.156 255.255.255.0
> shutdown
> duplex auto
> speed auto
>!
```

```
>ip classless
>ip route 8.1.1.0 255.255.255.0 30.1.1.1
>no ip http server
>!
>snmp-server packetsize 4096
>call rsvp-sync
>!
>voice-port 1/1/0
>!
>voice-port 1/1/1
>!
>dial-peer cor custom
>!
>dial-peer voice 1 pots
> destination-pattern 10
> port 1/1/0
>!
>dial-peer voice 2 voip
> destination-pattern 20
> session target ipv4:172.20.59.93
>!
>line con 0
> transport input none
>line aux 0
>line vty 0 4
> login
>!
>end
```

Displaying FXS Module Configuration Values

To display the values configured on the eight-Port RJ-21 FXS module, use the **show running-config** command, as follows:

```
>--------------------
>Gateway# sh running-config
>Building configuration...
>
>Current configuration :1062 bytes
>!
>version 12.1
>no service single-slot-reload-enable
>no service pad
>service timestamps debug uptime
>service timestamps log uptime
```

```
>no service password-encryption
>!
>hostname Gateway
>!
>no logging buffered
>no logging buffered
>logging rate-limit console 10 except errors
>!
>ip subnet-zero
>no ip finger
>!
>ip audit notify log
>ip audit po max-events 100
>!
>voicecard mode toll-by-pass
>!
>interface FastEthernet0/0
> ip address 172.20.59.93 255.255.0.0
> duplex auto
> speed auto
>!
>interface GigabitEthernet0/0
> ip address 1.1.1.1 255.255.255.0
> no negotiation auto
>!
>ip default-gateway 172.20.59.1
>ip classless
>no ip http server
>!
>call rsvp-sync
>!
>voice-port 3/0
>!
>voice-port 3/1
>!
>voice-port 4/0
>!
>voice-port 4/1
>!
>voice-port 4/2
>!
>voice-port 4/3
>!
>voice-port 4/4
>!
>voice-port 4/5
>!
>voice-port 4/6
```

```
>!
>voice-port 4/7
>!
>dial-peer voice 1 pots
> destination-pattern 20
> port 4/0
>!
>dial-peer voice 2 voip
> destination-pattern 10
> session target ipv4:172.20.59.61
>!
>gatekeeper
> shutdown
>!
>line con 0
> exec-timeout 0 0
> transport input none
>line vty 0 4
> login
>!
>end
```

Configuring Survivable Remote Site Telephony

Survivable Remote Site Telephony (SRST) provides Cisco CallManager with fallback support for Cisco IP Phones that are attached to the Catalyst 4224 Access Gateway Switch (Catalyst 4224) on your local Ethernet.

SRST enables the Catalyst 4224 to provide call handling support for Cisco IP Phones when a Cisco IP Phones lose connection to the remote primary, secondary, or tertiary Cisco CallManager or when the WAN connection is down.

This section contains following topics:

Overview of Survivable Remote Site Telephony

Cisco CallManager Releases 3.0 and later support Cisco IP Phones across the WAN that are attached to a branch office Catalyst 4224. Without the SRST feature, when the WAN connection between the branch office Catalyst 4224 and Cisco CallManager fails or when connectivity with Cisco CallManager is lost, the Cisco IP Phones at the remote site are unusable for the duration of the failure.

The SRST feature overcomes this problem, enabling the basic features of the Cisco IP Phones by providing call-handling support on the Catalyst 4224 for its attached Cisco IP Phones. The system automatically detects the failure and uses the Simple Network Auto Provisioning (SNAP) technology to autoconfigure the Catalyst 4224 to provide call processing for the local Cisco IP Phones.

When the WAN link or connection to the primary Cisco CallManager is restored, call-handling capabilities for the Cisco IP Phones switch back to the primary Cisco CallManager. During a failure when the SRST feature is enabled, the Cisco IP Phone displays a message stating that the Cisco IP Phones are in the Cisco CallManager fallback mode and are able to perform limited functions.

Note The Cisco IP Phone 7960 and 7940 displays the message "CM Fallback Service Operating." The Cisco IP Phone 7910 displays the message "CM Fallback Service."

Restrictions

Restrictions on the SRST feature are as follows:

- Supports only the Cisco IP Phone 7960, Cisco IP Phone 7940, and Cisco IP Phone 7910 models

Note First-generation Cisco IP Phones, such as Cisco IP Phone 30 VIP and Cisco IP Phone 12 SP+ are not supported.

- Does not support other Cisco CallManager applications or services such as Cisco IP SoftPhone, Cisco Unity, or Cisco IP Contact Center
- Supports 24 Cisco IP Phones on the Catalyst 4224

Prerequisites

Prerequisites for employing the SRST feature on the Catalyst 4224 are as follows:

- IP routing must be enabled.
- The SRST Catalyst 4224 must be configured as the default for the Cisco IP Phones.
- Cisco IOS Release 12.1(5)YD or higher is required.
- Cisco CallManager Release 3.0.5 or higher is required.
- You must use the appropriate Cisco IP Phone load versions that support the SRST feature for the Cisco IP Phone 7960, Cisco IP Phone 7940, and Cisco IP Phone 7910 models.

For further details, refer to your Cisco CallManager documentation.

Supported Features

The following features are supported on the Cisco IP Phones:

- Re-homing of Cisco IP Phones to use call processing on the local Catalyst 4224
- Cisco IP Phone and plain old telephone service (POTS) telephones on the Catalyst 4224
- Graying out of all Cisco IP Phone function keys that are not supported during SRST mode:
  - CFwdAll (call forward all)
  - MeetMe
  - PickUp
  - GPickUp (group pickup)
  - Park
  - Confrn (conference)
- Extension-to-extension dialing
- Direct inward dialing (DID)
- Direct outward dialing (DOD)

- Calling party ID (Caller ID/ANI) display
- Calling party name display
- Last number redial
- Maintaining local extension-to-extension calls when WAN link fails
- Maintaining local extension to public switched telephone network (PSTN) calls when WAN link fails
- Maintaining existing calls when failed WAN link is reestablished
- Call transfer of local calls (blind transfer)
- Multiple lines per Cisco IP Phone
- Multiple line appearance across telephones
- Call hold (shared lines)
- Analog Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports
- BRI support for EuroISDN
- PRI support for NET5 switch type
- Support for a maximum of 48 Cisco IP Phones

Note When the primary Cisco CallManager is restored, the displayed message is removed and you can access all the previous functionality of the Cisco IP Phone.

Fallback Behavior

When the Cisco IP Phones lose contact with all primary, secondary, and tertiary Cisco CallManagers, the phones re-home to the local SRST Catalyst 4224 to provide the call processing capability required to place and receive calls. (Re-homing is a major network change involving moving the local loop termination from one Central Office wire center to another.) The Cisco IP Phone lists the IP address of the local SRST Catalyst 4224 as the default Catalyst 4224 in the Network Configuration area of the Settings menu. This list supports up to five default Catalyst 4224 entries; however, Cisco CallManager uses a maximum

of three entries. When a secondary Cisco CallManager is not configured, the SRST Catalyst 4224 is listed as the stand-by Cisco CallManager during normal operation hosted by a single Cisco CallManager.

When the WAN link fails, calls in progress are sustained if possible for the duration of the call. Calls in transition must be attempted again after the Cisco IP Phones re-home to the local SRST Catalyst 4224. The telephone service is unavailable from the time the connection is lost from the remote Cisco CallManager until the Cisco IP Phone is re-homed to the Catalyst 4224 with the SRST feature.

The time taken to re-home to the remote Cisco CallManager depends in part on the keep-alive period set for Cisco CallManager. Typically, it takes three times the keep-alive period for the phone to discover that its connection to Cisco CallManager has failed. The default keep-alive period is 30 seconds.

If the phone has an active standby connection established with the SRST Catalyst 4224, the fallback process itself takes 10 to 20 seconds after the primary Cisco CallManager has failed. An active standby connection to the SRST Catalyst 4224 exists only if the phone has a single Cisco CallManager in its Cisco CallManager list. Otherwise, the phone will maintain a standby connection to its secondary Cisco CallManager.

If the phone has multiple Cisco CallManagers in its Cisco CallManagers list, it must search through the list of secondary and tertiary Cisco CallManagers, which means the fallback time increases. The phone attempts to connect to each of its alternate Cisco CallManagers in turn before attempting to connect to the SRST Catalyst 4224 as a last resort.

Each Cisco CallManager connection attempt will take about one minute. A message is displayed on the Cisco IP Phone indicating that the Cisco IP Phones are in Cisco CallManager fallback mode.

These telephone services are restricted to those Cisco IP Phones that are supported by SRST Catalyst 4224.

Note The Cisco IP Phone 7960 and 7940 display the message “CM Fallback Service Operating” at the bottom of the screen. The Cisco 7910 IP Phone displays the message “CM Fallback Service” for 5 seconds every 30 seconds. The message is brief because the Cisco IP Phone 7910 has only a two-line display area and the display panel is also used to display the telephone number.

Cisco IP Phones periodically attempt to reestablish connections with the remote Cisco CallManagers. When a connection is reestablished with a remote Cisco CallManager, the Cisco IP Phones unregister from the local Catalyst 4224 with the SRST feature and register with the remote Cisco CallManager. The connection to the primary Cisco CallManager at the remote office cannot be reestablished if the telephone has active calls.

Figure 8-1 shows a branch office with several Cisco IP Phones connected to a Catalyst 4224. The Catalyst 4224 is connected to the remote centralized Cisco CallManager by the WAN and PSTN.

Figure 8-1 Branch Office Cisco IP Phones Connected to the Remote Central Cisco CallManager

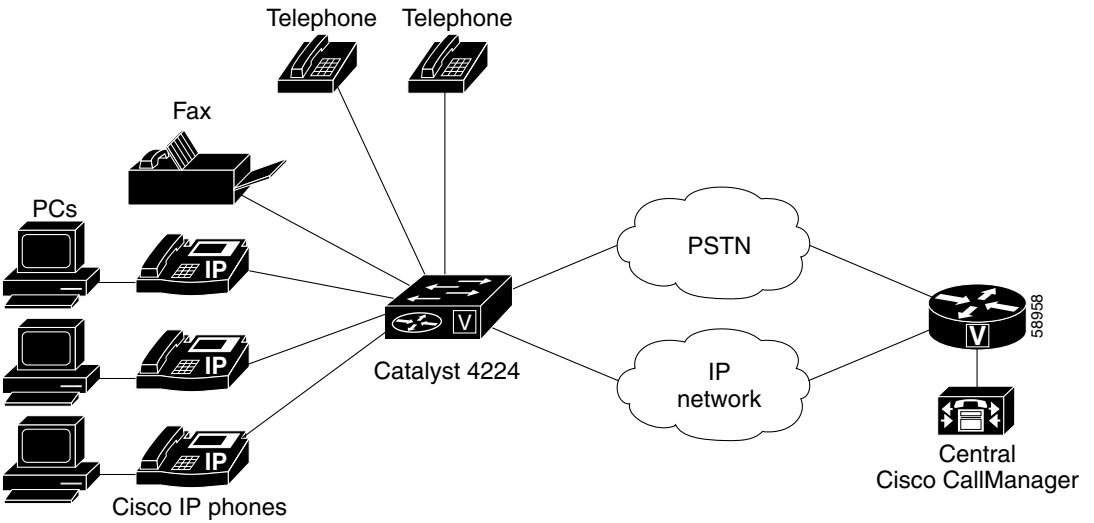
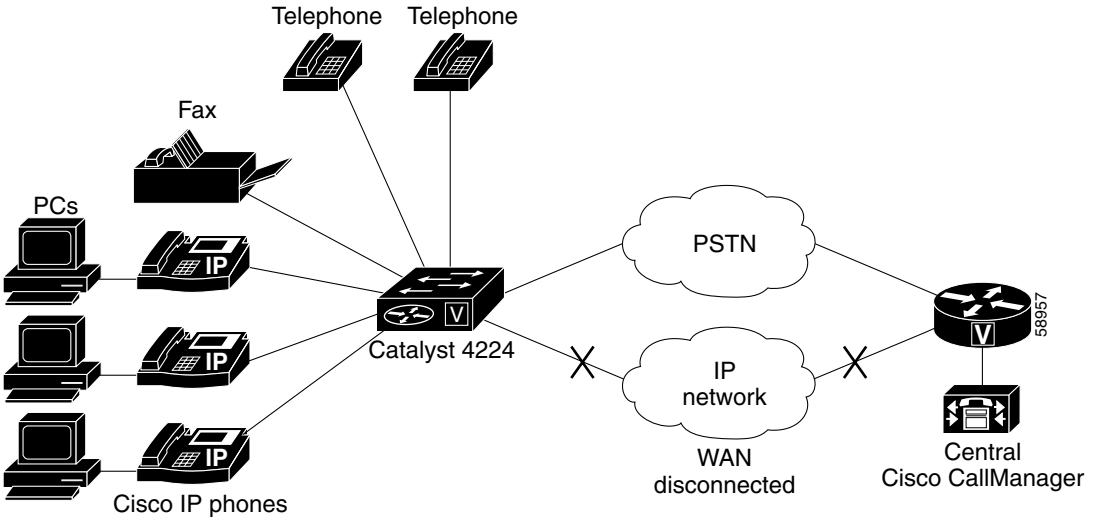


Figure 8-2 shows that the WAN connection to the branch office is down and that the Cisco IP Phones are able to make calls by being connected to the local SRST Catalyst 4224. This Catalyst 4224 acts as a fallback Cisco CallManager. The branch office Cisco IP Phones on the LAN network, connected to the PSTN, are capable of making off-net calls outside the network using the Catalyst 4224s.

Figure 8-2 Branch Office Cisco IP Phones Operating in Survivable Remote Site Telephony (SRST) Mode



Configuring Survivable Remote Site Telephony

When SRST is enabled, Cisco IP Phones do not need to be reconfigured individually during Cisco CallManager fallback mode because Cisco IP Phones retain the same configuration used with the primary Cisco CallManager.

To configure SRST on the Catalyst 4224, use the following steps, beginning in global configuration mode:

	Task	Command
Step 1	Enable SRST support and enters Cisco CallManager fallback mode.	`C4224(config)# call-manager-fallback`
Step 2	Enable the Catalyst 4224 to receive messages from the Cisco IP Phones through the specified IP addresses and ports.	`C4224(config-cm-fallback)# ip source-address` *ip-address* `port` *port*

	Task	Command
Step 3	Configure the allowable number of Cisco IP Phones that can be supported by the Catalyst 4224. **Note** You cannot reduce the allowable number of Cisco IP Phones once the maximum allowable number is configured, without rebooting the Catalyst 4224.	`C4224(config-cm-fallback)# max-ephones` *max phone*
Step 4	Set the allowable number of directory numbers that can be supported by the Catalyst 4224. **Note** You cannot reduce the allowable number of the directory numbers once the maximum allowable number is configured, without rebooting the Catalyst 4224.	`C4224(config-cm-fallback)# max-dn` *max directory number*
Step 5	(Optional) Configure the time interval between keepalive messages sent by Cisco IP Phones to the Catalyst 4224. E This setting is effective during Cisco CallManager fallback mode when SRST is enabled.	`C4224(config-cm-fallback)# keepalive` *seconds*
Step 6	(Optional) Assign a default destination number for incoming telephone calls.	`C4224(config-cm-fallback)# default-destination` *telephone number*
Step 7	(Optional) Create a global prefix that can be used to expand the abbreviated extension numbers into a fully qualified E.164 numbers. The **extension-length** keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller ID display, received, and missed-call lists.	`C4224(config-cm-fallback)# dialplan-pattern` *tag pattern* `extension-length` *number*
Step 8	(Optional) Allow transfer of telephone calls by Cisco IP Phones to other phone numbers (IP and non-IP phone numbers).	`C4224(config-cm-fallback)# transfer-pattern` *transfer-pattern*

	Task	Command
Step 9	(Optional) Configure trunk access codes for each type of line—Basic Rate Interface (BRI), E&M, Foreign Exchange Office (FXO), and Primary Rate Interface (PRI)—so that the Cisco IP Phones can access the trunk lines during Cisco CallManager fallback mode when the SRST feature is enabled.	`C4224(config-cm-fallback)# access-code {bri | e&m |fxo | pri}`
Step 10	(Optional) Configure the telephone number that is speed-dialed when the message button on a Cisco IP Phone is pressed.	`C4224(config-cm-fallback)# voicemail` *phone-number*
Step 11	Exit Cisco CallManager fallback configuration mode.	`C4224(config-cm-fallback)# exit`
Step 12	Exit global configuration mode.	`C4224(config)# exit`

Note In Cisco CallManager fallback mode, huntstop is set by default. (Huntstop disables all further dial-peer hunting for the dial peers associated with the Cisco IP Phone lines if a call fails using hunt groups.) Use the **no huntstop** command only if you want to disable huntstop.

Verifying Survivable Remote Site Telephony

To verify that the SRST feature is enabled, follow these steps:

Step 1 Enter the **show run** command to verify the configuration.

Step 2 Enter the **show call-manager-fallback all** command to verify that the SRST feature is enabled.

Step 3 Verify that the default Catalyst 4224 IP address on the Cisco IP Phone is the same as the IP address of the SRST Catalyst 4224. Use the settings display on a Cisco IP Phone to do so.

Step 4 Temporarily block the TCP port 2000 Skinny Client Protocol (SCCP) connection for one of the Cisco IP Phones. This forces the Cisco IP Phone to lose its connection to Cisco CallManager and register with the SRS Catalyst 4224. Perform the following:

1. Use the appropriate **access-list** command to temporarily disconnect a Cisco IP Phone from Cisco CallManager.

Note The Cisco IP Phone 7960 and 7940 display the message "CM Fallback Service Operating" at the bottom of the display screen. The Cisco IP Phone 7910 displays the message "CM Fallback Service" for five seconds every 30 seconds. This message is brief because the Cisco IP Phone 7910 has a two-line display area and the display panel is also used to display the telephone number.

2. Delete the **access-list** command to restore normal service for the phone by entering the **no** form of the appropriate **access-list** command.
3. Use the **debug ephone register** command to observe the registration process of the Cisco IP Phone on the SRST Catalyst 4224.
4. You can also enter the **show ephone** command to display the Cisco IP Phones that have registered to the SRST Catalyst 4224.

Troubleshooting Survivable Remote Site Telephony

Use the following commands to troubleshoot the SRST feature.

Command	Purpose
`debug ephone keepalive`	Sets keepalive debugging for the Cisco IP Phone.
`debug ephone register`	Sets registration debugging for the Cisco IP Phone.
`debug ephone state`	Sets state debugging for the Cisco IP Phone.
`debug ephone detail`	Sets detail debugging for the Cisco IP Phone.
`debug ephone error`	Sets error debugging for the Cisco IP Phone.
`debug ephone statistics`	Sets call statistics debugging for the Cisco IP Phone.

Command	Purpose
`debug ephone pak`	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
`debug ephone raw`	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages.

Monitoring and Maintaining Survivable Remote Site Telephony

Use the following commands to monitor and maintain the Catalyst 4224 with the SRST feature:

Command	Purpose
`C4224# show run`	Displays the configuration.
`C4224# show call-manager-fallback all`	Displays the detailed configuration of all the Cisco IP Phones, voice ports, and dial peers of the SRST Catalyst 4224.
`C4224# show call-manager-fallback dial-peer`	Displays the output of the dial peers of the SRST Catalyst 4224.
`C4224# show call-manager-fallback ephone-dn`	Displays the Cisco IP Phone destination number.
`C4224# show call-manager-fallback voice-port`	Displays output for the voice ports.
`C4224# show ephone`	Displays Cisco IP Phone output.
`C4224# show ephone-dn`	Displays the Cisco IP Phone destination number.
`C4224# show ephone summary`	Displays a summary of the Cisco IP Phone output.
`C4224# show voice port summary`	Displays a summary of all voice ports.
`C4224# show dial-peer voice summary`	Displays a summary of all voice dial peers.

SRST Configuration Example

This section provides a configuration example for the SRST feature:

```
!
version 12.1
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Garfield
!
logging rate-limit console 10 except errors
!
ip subnet-zero
!
no ip finger
no ip domain-lookup
!
ip dhcp pool PHONE1
   host 10.1.0.2 255.255.0.0
   client-identifier 0100.3094.c337.cb
   option 150 ip 172.198.0.2
   default-C4224 10.1.0.1
!
ip dhcp pool PHONE2
   host 10.1.0.3 255.255.0.0
   client-identifier 0100.3094.c3f9.6a
   default-C4224 10.1.0.1
   option 150 ip 172.198.0.2
!
interface FastEthernet0/0
 ip address 10.1.0.1 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 172.198.0.1 255.255.0.0
 duplex auto
 speed auto
!
interface Ethernet1/0
 no ip address
 shutdown
 half-duplex
!
interface Serial1/0
```

```
 no ip address
!
interface TokenRing1/0
 no ip address
 shutdown
!
ip kerberos source-interface any
ip classless
no ip http server
!
snmp-server packetsize 4096
snmp-server manager
call rsvp-sync
!
mgcp modem passthrough voip mode ca
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
!
call-manager-fallback
 ip source-address 10.0.0.1 port 2000 strict-match
 max-ephones 24
 max-dn 24
 dialplan-pattern 1 408734.... extension-length 4
 transfer-pattern 510650....
 voicemail 11111
!
line con 0
 transport input none
line aux 0
line vty 0 4
 login
!
end
```

Implementing Fax over IP on Cisco Voice Gateways

Fax over IP enables interoperability of traditional analog fax machines with IP telephony networks. The fax image is converted from an analog signal and is carried as digital data over the packet network.

This section contains the following topics:

Note For detailed Fax information, configuration, best practices and troubleshooting tips, see *Implementing Fax Over IP on Cisco Voice Gateways* at the following location:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_access/fxmdmnt.htm

Overview

In its original form, fax data is digital. However, to transmit across a traditional PSTN, it is modulated and converted to analog. Fax over IP reverses this analog conversion, transmitting digital data over the packet network, and then reconverting the digital data to analog for the receiving fax machine.

Most Cisco voice gateways currently support two methods to transmit fax traffic across the IP network:

- Fax Pass-Through—In fax pass-through mode, the gateways do not distinguish a fax call from a voice call.
- Cisco Fax Relay—In fax relay mode, the gateways terminate the T.30 fax signaling.

Fax relay mode is the preferred method to transmit fax traffic. However, if a specific gateway does not support Cisco fax relay, the gateway supports fax pass-through.

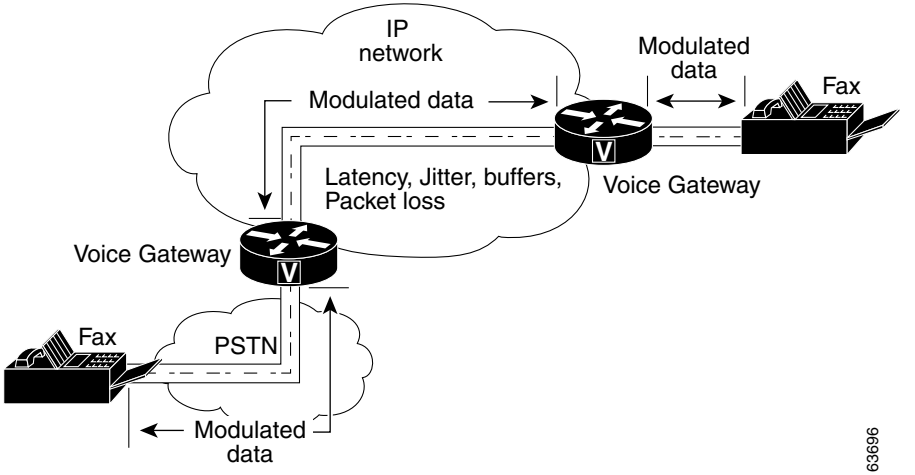
Fax Pass-Through

In fax pass-through mode, the gateways do not distinguish a fax call from a voice call. The fax communication between the two fax machines is carried in-band over a "voice" call in its entirety. All Cisco voice gateways support fax pass-through.

On Cisco voice gateways, you can achieve more reliable fax transmissions using fax pass-through mode if you disable VAD on the gateway dial peers.

Figure 9-1 illustrates how fax pass-through works. The fax traffic is transparently carried across the quality of service (QoS)-enabled IP infrastructure, and the data is not demodulated within the IP network.

Figure 9-1 Fax Pass-Through



Cisco Fax Relay

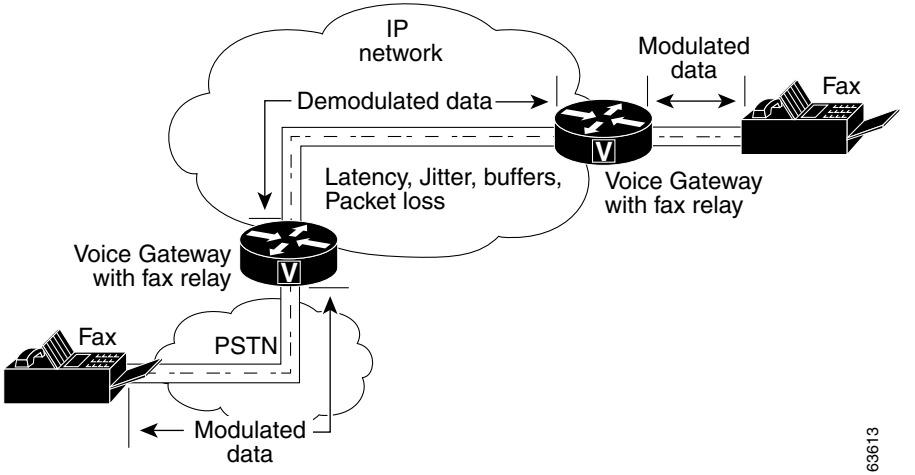
Cisco fax relay does not involve Cisco CallManager; it is a gateway-controlled fax mode. Most of the fax processing occurs in the digital signal processors (DSP), requiring only packet switching from the main processor (CPU) and some limited signaling to switch to fax mode. Therefore, the CPU overhead is very similar to a normal voice call.

Initially, a voice call is established. Once the V.21 preamble is detected at the terminating gateway, the originating and the terminating gateway negotiate the codec type. If the two sides cannot negotiate on a common codec and speed, the fax fails. If negotiation is successful, fax transmission begins.

Cisco fax relay is supported using MGCP or H.323, depending on the specific gateway type. The Cisco voice gateways support as many concurrent fax calls as G.711 voice calls.

As illustrated in Figure 9-2, the voice gateway demodulates the fax data before crossing the IP network. The voice gateway at the other end of the IP network demodulates it for transfer across the PSTN.

Figure 9-2 Cisco Fax Relay



Supported Platforms and Features

Table 9-1 shows the supported fax protocol, modes, and required software version for the Catalyst 4224 Access Gateway Switch. You can obtain the latest versions of Cisco software from the Cisco software downloads site: http://www.cisco.com/kobayashi/sw-center/index.shtml

Table 9-1 Fax Support on Cisco Catalyst 4224 Access Gateway Switch

Platform	Protocol	Transport Mode	Minimum Software Version
Catalyst 4224	H.323	Pass-through	12.1.5YE2
		Relay	12.1.5YE2

CHAPTER 10

Traffic Shaping

Cisco IOS Quality of Service (QoS) software includes four types of traffic shaping:

- Generic Traffic Shaping (GTS)
- Class-Based Traffic Shaping
- Frame Relay Traffic Shaping (FRTS)
- Distributed Traffic Shaping (DTS)

All four traffic shaping methods are similar in implementation, though their command line interfaces (CLIs) differ somewhat and they use different types of queues to contain and shape traffic that is deferred. If a packet is deferred, GTS and Class-Based Shaping use a weighted fair queue to hold the delayed traffic. FRTS uses either a custom queue or a priority queue.

This section explains how traffic shaping works and describes the Cisco IOS QoS traffic shaping mechanisms. It also described traffic-shaping feature called Low Latency Traffic Shaping (LLQ).

This section contains the following topics:

About Traffic Shaping

Traffic shaping allows you to control outgoing traffic on an interface to match the traffic speed of the remote target interface and to ensure that the traffic conforms to specific policies. Traffic that adheres to a particular profile can be shaped to meet downstream requirements, thereby eliminating bottlenecks in topologies caused by data-rate mismatches.

Why Use Traffic Shaping?

The primary reasons to use traffic shaping are to control access to available bandwidth, to ensure that traffic conforms to specific policies, and to regulate the flow of traffic in order to avoid congestion. Some example reasons for using traffic shaping follow:

- Control access to bandwidth when policy dictates that the average rate of a given interface should not exceed a certain rate.
- Configure traffic shaping on an interface if you have a network with differing access rates. Suppose that one end of the link in a Frame Relay network runs at 256 kbps and the other end of the link runs at 128 kbps. Sending packets at 256 kbps could cause failure of the applications using the link.

  A similar, more complicated case would be a link-layer network giving indications of congestion with differing access rates on different attached data terminal equipment (DTE) devices. The network may be able to deliver more transit speed to a given DTE device at a specific time than at another time.
- If you offer a subrate service, traffic shaping enables you to use the router to partition your T1 or T3 links into smaller channels.

Traffic shaping prevents packet loss. Its use is especially important in Frame Relay networks because the switch cannot determine which packets take precedence or which packets should be dropped when congestion occurs.

Traffic Shaping and Rate of Transfer

Traffic shaping limits the rate of transmission of data. You can limit the data transfer to one of the following:

- A specific configured rate
- A derived rate based on the level of congestion

The rate of transfer depends on three components that constitute the token bucket: burst size, mean rate, and measurement (time) interval. The mean rate is equal to the burst size divided by the interval.

When traffic shaping is enabled, the bit rate of the interface does not exceed the mean rate over any integral multiple of the interval. During every interval, the burst size is usually the maximum number of bits that can be sent. Within the interval, however, the bit rate may be faster than the mean rate at any given time.

One additional variable applies to traffic shaping: Excess Burst Size (called the *Be size)*. The Be Size corresponds to the number of noncommitted bits—those outside the committed information rate (CIR)—that are still accepted by the Frame Relay switch but are marked as discard eligible (DE).

The Be size allows more than the burst size to be sent during a time interval in certain situations. The switch allows the packets belonging to the Excess Burst to go through but it will mark them by setting the DE bit. Whether the packets are sent depends on how the switch is configured.

When the Be size equals 0, the interface sends no more than the burst size every interval, achieving an average rate no higher than the mean rate. However, when the Be size is greater than 0, the interface can send as many as Bc + Be bits in a burst, if the maximum amount was not sent in a previous time period. Whenever the number of bits sent during an interval is less than the burst size, the remaining number of bits can be sent in a later interval.

Discard Eligible Bit

You can specify which Frame Relay packets have low priority or low time sensitivity. These packets are the first to be dropped when a Frame Relay switch is congested. The Discard Eligible (DE) bit allows a Frame Relay switch to identify such packets.

You can define DE lists that identify the characteristics of packets to be eligible for discarding, and you can also specify DE groups to identify the data-link connection identifier (DLCI) that is affected.

You can specify DE lists based on the protocol or the interface. You can also specify DE lists that are based on characteristics such as fragmentation of the packet, a specific TCP or User Datagram Protocol (UDP) port, an access list number, or a packet size.

Differences Between Shaping Mechanisms

GTS, Class-Based Shaping, DTS, and FRTS are similar in implementation, sharing the same code and data structures, but they differ in regard to their CLIs and the queue types they use.

Here are a few ways in which these mechanisms differ:

- For GTS, the shaping queue is a weighted fair queue. For FRTS, the queue can be a weighted fair queue (configured by the **frame-relay fair-queue** command), a strict priority queue with weighted fair queueing (WFQ) (configured by the **frame-relay ip rtp priority** command in addition to the **frame-relay fair-queue** command), custom queueing (CQ), priority queueing (PQ), or first-in, first-out queueing (FIFO).
- For Class-Based Shaping, GTS can be configured on a class, rather than only on an access control list (ACL). You must first define traffic classes based on match criteria including protocols, ACLs, and input interfaces. You can then apply traffic shaping to each defined class.
- FRTS supports shaping on a per-DLCI basis; GTS and DTS are configurable per interface or subinterface.
- DTS supports traffic shaping based on a variety of match criteria, including user-defined classes, and differentiated services code point (DSCP).

Table 10-1 summarizes these differences.

Table 10-1 Differences Between Shaping Mechanisms

Mechanism	GTS	Class-Based	DTS	FRTS
Command-Line Interface	• Applies parameters per subinterface • **traffic group** command supported	• Applies parameters per interface or per class	• Applies parameters per interface or subinterface	• Classes of parameters • Applies parameters to all virtual circuits (VCs) on an interface through inheritance mechanism • No traffic group command
Queues Supported	• WFQ per subinterface	• class-based weighted fair queuing (CBWFQ) inside GTS	• WFQ, strict priority queue with WFQ, CQ, PQ, first-come, first-served (FCFS) per VC	• WFQ, strict priority queue with WFQ, CQ, PQ, FCFS per VC

You can configure GTS to behave the same as FRTS by allocating one DLCI per subinterface and using GTS plus backward explicit congestion notification (BECN) support.

Traffic Shaping and Queueing

Traffic shaping smooths traffic by storing traffic above the configured rate in a queue.

When a packet arrives at the interface for transmission, the following sequence occurs:

1. If the queue is empty, the arriving packet is processed by the traffic shaper.
   - If possible, the traffic shaper sends the packet.
   - Otherwise, the packet is placed in the queue.
2. If the queue is not empty, the packet is placed in the queue.

When packets are in the queue, the traffic shaper removes the number of packets it can send from the queue at each time interval.

Generic Traffic Shaping

Generic Traffic Shaping (GTS) shapes traffic by reducing outbound traffic flow to avoid congestion. GTS constrains traffic to a particular bit rate using the token bucket mechanism. See the section "What is a Token Bucket" in the *Cisco IOS Quality of Service Solutions Configuration Guide, Release 12.2*:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt4/qcfpolsh.htm#1000909

How It Works

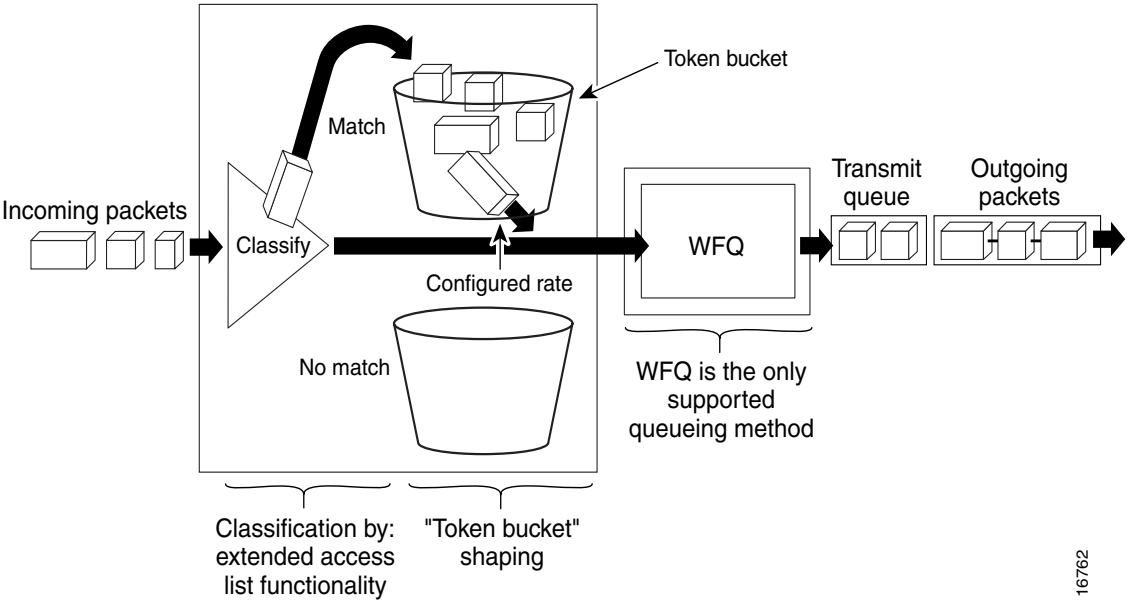
GTS applies traffic shaping on a per-interface basis and can use access lists to select the traffic to shape. GTS works with a variety of Layer 2 technologies, including Frame Relay, ATM, Switched Multimegabit Data Service (SMDS), and Ethernet.

On a Frame Relay subinterface, GTS can be set up to adapt dynamically to available bandwidth by integrating backward explicit congestion notification (BECN) signals. GTS also can be shape traffic to a specified rate. GTS can be configured on an ATM/ATM Interface Processor (AIP) interface to respond to the Resource Reservation Protocol (RSVP) feature signalled over statically configured ATM permanent virtual circuits (PVCs).

GTS is supported on most media and encapsulation types on the router. GTS can be applied to a specific access list on an interface.

Figure 10-1 shows how GTS works.

Figure 10-1 Generic Traffic Shaping



Configuration and Commands

For information on how to configure GTS, see the chapter "Configuring Generic Traffic Shaping" in the *Cisco IOS Quality of Service Solutions Configuration Guide, Release 12.2*:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt4/qcfgts.htm#80560

For information on traffic shaping commands, see the *Cisco IOS Quality of Service Solutions Command Reference, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_r/index.htm

Class-Based Traffic Shaping

Class-Based Traffic Shaping can be enabled on any interface that supports GTS.

How It Works

Using Class-Based Traffic Shaping, you can perform the following tasks:

- Configure GTS on a traffic class to provide greater flexibility for configuring traffic shaping. Previously, this ability was limited to the use of ACLs.
- Specify average rate or peak rate traffic shaping. This type of shaping allows more data than the CIR to be sent if the bandwidth is available.
- Configure class-based weighted fair queueing (CBWFQ) inside GTS. CBWFQ allows you to specify the exact amount of bandwidth to allocate for a specific class of traffic. You can configure up to 64 classes and control their distribution.

  Flow-based WFQ applies weights to traffic to classify the traffic into conversations and determine how much bandwidth each conversation is allowed. These weights and traffic classifications are dependent on and limited to the seven IP Precedence levels.

  CBWFQ allows you to define what constitutes a class based on criteria that exceed the confines of flow. CBWFQ allows you to use ACLs and protocols or input interface names to define how traffic will be classified, thereby providing coarser granularity. You need not maintain traffic classification on a flow basis. Moreover, you can configure up to 64 discrete classes in a service policy.

Configuration and Commands

For information on how to configure Class-Based Shaping, see the chapter "Configuring Class-Based Shaping" in the *Cisco IOS Quality of Service Solutions Configuration Guide, Release 12.2*:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt4/qcfcbshp.htm#80464

For information on traffic shaping commands, see the *Cisco IOS Quality of Service Solutions Command Reference, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_r/index.htm

Restrictions

Peak and average traffic shaping is configured on a per-interface or per-class basis, and cannot be used in conjunction with commands used to configure GTS from previous versions of Cisco IOS. These commands include the following:

- **traffic-shape adaptive**
- **traffic-shape fecn-adaptive**
- **traffic-shape group**
- **traffic-shape rate**

Adaptive traffic shaping for Frame Relay networks is not supported using the Class-Based Shaping feature. To configure adaptive GTS for Frame Relay networks, you must use the commands from releases prior to Release 12.1(2) of Cisco IOS software.

Frame Relay Traffic Shaping

Cisco has long provided support for FECN for DECnet and OSI, and BECN for Systems Network Architecture (SNA) traffic using Logical Link Control, type 2 (LLC2) encapsulation via RFC 1490 and DE bit support. FRTS builds upon this existing Frame Relay support with additional capabilities that improve the scalability and performance of a Frame Relay network, increasing the density of VCs and improving response time.

As is also true of GTS, FRTS can eliminate bottlenecks in Frame Relay networks that have high-speed connections at the central site and low-speed connections at branch sites. You can configure rate enforcement—a peak rate configured to limit outbound traffic—to limit the rate at which data is sent on the VC at the central site.

How It Works

Using FRTS, you can configure rate enforcement to either the CIR or some other defined value such as the excess information rate on a per-VC basis. The ability to allow the transmission speed used by the router to be controlled by criteria other than line speed provides a mechanism for sharing media by multiple VCs. You can allocate bandwidth to each VC, creating a virtual time-division multiplexing (TDM) network.

You can also define PQ, CQ, and WFQ at the VC or subinterface level. Using these queueing methods allows for finer granularity in the prioritization and queueing of traffic, providing more control over the traffic flow on an individual VC. If you combine CQ with the per-VC queueing and rate enforcement capabilities, you enable Frame Relay VCs to carry multiple traffic types such as IP, SNA, and Internetwork Packet Exchange (IPX) with bandwidth guaranteed for each traffic type.

Using information contained in the BECN-tagged packets received from the network, FRTS can also dynamically throttle traffic. With BECN-based throttling, packets are held in the buffers of the router to reduce the data flow from the router into the Frame Relay network. The throttling is done on a per-VC basis and the transmission rate is adjusted based on the number of BECN-tagged packets received.

With the Cisco FRTS feature, you can integrate ATM ForeSight closed-loop congestion control to actively adapt to downstream congestion conditions.

Derived Rates

In Frame Relay networks, BECNs and FECNs indicate congestion. BECN and FECN are specified by bits within a Frame Relay frame.

FECNs are generated when data is sent out a congested interface; they indicate to a DTE device that congestion was encountered. Traffic is marked with BECN if the queue for the opposite direction is deep enough to trigger FECNs at the current time.

BECNs notify the sender to decrease the transmission rate. If the traffic is one-way only (such as multicast traffic), there is no reverse traffic with BECNs to notify the sender to slow down. Thus, when a DTE device receives an FECN, it first determines if it is sending any data in return. If it is sending return data,

this data will get marked with a BECN on its way to the other DTE device. However, if the DTE device is not sending any data, the DTE device can send a Q.922 TEST RESPONSE message with the BECN bit set.

When an interface configured with traffic shaping receives a BECN, it immediately decreases its maximum rate by a large amount. If, after several intervals, the interface has not received another BECN and traffic is waiting in the queue, the maximum rate increases slightly. The dynamically adjusted maximum rate is called the derived rate.

The derived rate will always be between the upper bound and the lower bound configured on the interface.

Configuration and Commands

For more information on configuring Frame Relay, refer to the chapter "Configuring Frame Relay" in the *Cisco IOS Wide-Area Networking Configuration Guide, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fwan_c/wcffrely.htm

For information on configuring Frame Relay as it relates to voice traffic, refer to the chapter "Configuring Voice Over Frame Relay" in the *Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/vvfvofr.htm

For information on Frame Relay commands, see the section "Frame Relay Commands" in the *Cisco IOS Wide-Area Networking Command Reference, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fwan_r/frcmds/index.htm

For information on Frame Relay ATM commands, see the section "Frame Relay—ATM Internetworking Commands" in the *Cisco IOS Wide-Area Networking Command Reference, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fwan_r/frcmds/index.htm

Restrictions

FRTS applies only to Frame Relay PVCs and switched virtual circuits (SVCs).

Distributed Traffic Shaping

DTS provides a method of managing the bandwidth of an interface to avoid congestion, to meet remote site requirements, and to conform to a service rate that is provided on that interface.

DTS uses queues to buffer traffic surges that can congest a network and send the data to the network at a regulated rate. This ensures that traffic will behave to the configured descriptor, as defined by the CIR, Bc, and Be. With the defined average bit rate and burst size that is acceptable on that shaped entity, you can derive a time interval value.

Prerequisites

Distributed Cisco Express Forwarding (dCEF) must be enabled on the interface before DTS can be enabled.

A policy map and class maps must be created before DTS is enabled.

How It Works

The Be size allows more than the Bc size to be sent during a time interval under certain conditions. Therefore, DTS provides two types of **shape** commands: **average** and **peak**. When **shape average** is configured, the interface sends no more than the Bc size for each interval, achieving an average rate no higher than the CIR. When the **shape peak** command is configured, the interface sends Bc plus Be bits in each interval.

In a link layer network such as Frame Relay, the network sends messages with the forward explicit congestion notification (FECN) or BECN if there is congestion. With the DTS feature, the traffic shaping adaptive mode takes advantage of these signals and adjusts the traffic descriptors, thereby regulating the amount of traffic entering or leaving the interface accordingly.

DTS provides the following key benefits:

- Offloads traffic shaping from the Route Switch Processor (RSP) to the VIP.
- Supports up to 200 shape queues per VIP, supporting up to OC-3 rates when the average packet size is 250 bytes or greater and when using a VIP2-50 or better with eight MB of SRAM. Line rates below T3 are supported with a VIP2-40.
- Configures DTS at the interface level or subinterface level.
- Shaping based on the following traffic match criteria:
  - Access list
  - Packet marking
  - Input port
  - Other matching criteria. For information about other matching criteria, see the section "Creating a Traffic Class" in the *Cisco IOS Quality of Service Solutions Command Reference, Release 12.2* manual.
- Optional configuration to respond to Frame Relay network congestion by reducing the shaped-to rate for a period of time until congestion is believed to have subsided. Supports FECN, BECN, and ForeSight Frame Relay signaling.

This feature runs on Cisco 7500 series routers with VIP2-40, VIP2-50, or greater.

Configuration

For information on how to configure DTS, see the chapter "Configuring Distributed Traffic Shaping" in the *Cisco IOS Quality of Service Solutions Command Reference, Release 12.2* manual:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt4/qcfdts.htm

Restrictions

DTS does not support the following:

- Fast EtherChannel, Multilink PPP (MLP), Tunnel, VLANs, and dialer interface
- Any VIP below a VIP2-40

Note A VIP2-50 is strongly recommended when the aggregate line rate of the port adapters on the VIP is greater than DS3. A VIP2-50 card is required for OC-3 rates.

Low-Latency Queueing

The LLQ feature brings strict PQ to CBWFQ. Strict PQ allows delay-sensitive data such as voice to be dequeued and sent before packets in other queues are dequeued.

Without LLQ, CBWFQ provides WFQ based on defined classes with no strict priority queue available for real-time traffic. CBWFQ allows you to define traffic classes and then assign characteristics to these classes. For example, you can designate the minimum bandwidth delivered to the class during congestion.

For CBWFQ, the weight for a packet belonging to a specific class is derived from the bandwidth you assigned to the class at configuration. Therefore, the bandwidth assigned to the packets of a class determines the order in which packets are sent. All packets are serviced fairly based on weight; no class of packets may be granted strict priority. This scheme poses problems for voice traffic that is largely intolerant of delay, especially variation in delay. For voice traffic, variations in delay introduce irregularities of transmission, such as *jitter* in the heard conversation.

LLQ provides strict priority queueing for CBWFQ, reducing jitter in voice conversations. Configured by the priority command, LLQ enables use of a single, strict priority queue within CBWFQ at the class level. This allows you to direct traffic belonging to a class to the CBWFQ strict priority queue. To enqueue class traffic to the strict priority queue, specify the named class within a policy map and then configure the priority command for the class. (Classes to which the priority command is applied are considered priority classes.) Within a policy map, you

can give one or more classes priority status. When multiple classes within a single policy map are configured as priority classes, all traffic from these classes is enqueued to the same single, strict priority queue.

One of the ways in which the strict PQ used within CBWFQ differs from its use outside CBWFQ is in the parameters it takes. Outside CBWFQ, you can use the **ip rtp priority** command to specify the range of UDP ports whose voice traffic flows are given priority service.

Using the priority command, you are no longer limited to a UDP port number to stipulate priority flows because you can configure the priority status for a class within CBWFQ. Instead, all of the valid match criteria used to specify traffic for a class now apply to priority traffic. These methods of specifying traffic for a class include matching on access lists, protocols, and input interfaces. Moreover, within an access, list you can specify that traffic matches are allowed based on the IP differentiated services code point (DSCP) value. This value is set using the first six bits of the ToS byte in the IP header.

Although it is possible to enqueue various types of real-time traffic to the strict priority queue, Cisco strongly recommends that you direct only voice traffic to this queue. The reason is that voice traffic is well-behaved, whereas other types of real-time traffic are not well-behaved. Moreover, voice traffic requires nonvariable delays to avoid jitter.

Real-time traffic such as video could introduce variation in delay, thereby thwarting the steadiness of delay required for successful voice traffic transmission.

For more conceptual information about LLQ, see the section “Weighted Fair Queueing” in the *Cisco IOS Quality of Service Solutions Configuration Guide, Release 12.2*:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt2/qcfconmg.htm#xtocid46014

For information on how to configure LLQ, see the chapter "Configuring Weighted Fair Queueing" in the *Cisco IOS Quality of Service Solutions Configuration Guide, Release 12.2*:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fqos_c/fqcprt2/qcfwfq.htm

Configuring Encryption Services

The Encryption Service Adapter (ESA) is a high-performance data encryption module that offloads some of the encryption processing from the Catalyst 4224 main processor and improves performance. The ESA implements data encryption and authentication algorithms on the Catalyst 4224 through a software service called a crypto engine.

The ESA includes a public key math processor and a hardware random number generator. These features support public key cryptography for key generation, exchange, and authentication. The ESA can encrypt and authenticate two full-duplex T1 or two E1 communication links.

Each data line can be channelized with a separate encryption context. The ESA uses Public Key (PK) technology based on the concept of the Protected Entity (PE) and employs IPSec Data Encryption Standard (DES) 56-bit and 3(Triple) DES 168-bit encryption to ensure that secure data and information can be transferred between similarly equipped hosts on your network.

This section details how to configure the ESA and includes the following topics:

Configuring the Encryption Service Adapter

Configuring the ESA requires four steps, as outlined below:

Step 1: Configure the T1 Channel Group

The first step toward configuring the ESA is to establish a T1 connection. You must define the characteristics of a configuration group (such as speed and slot number).

To configure the T1 channel group, follow this procedure:

	Task	Command
Step 1	Specify a controller and enter controller configuration mode.	`Gateway(config)# controller {t1|e1}slot/port`
Step 2	Specify the clock source for a link. **line** specifies that the link uses the recovered clock from the link and is the default setting. Generally, this setting is most reliable. **internal** specifies that the DS1 link uses the internal clock. **loop-timed** specifies that the T1 or E1 interface takes the clock from the Rx (line) and uses it for Tx. This setting decouples the controller clock from the system-wide clock set with the **network-clock-select** command.	`Gateway(config-controller)# clock source {line|internal|loop-timed}`
Step 3	Select frame clock.	`Gateway(config-controller)# frame-clock-select{priority} {E1/T1} {slot/port}`

	Task	Command
Step 4	Specify the framing type for the T1 or E1 data line. **sf** specifies Super Frame as the T1 frame type. **esf** specifies Extended Super Frame as the T1 frame type.	`Gateway(config-controller)#` **`framing {sf|esf}`**
Step 5	Specify the line code format. **ami** specifies alternate mark inversion (AMI) as the line-code type. Valid for T1 or E1 controllers; the default for T1 lines. **b8zs** specifies B8ZS as the line-code type. Valid for T1 controller only. **hdb3** specifies high-density bipolar 3 (hdb3) as the line-code type. Valid for E1 controller only; the default for E1 lines.	`Gateway(config-controller)#` **`linecode {ami|b8zs|hdb3}`**
Step 6	Specify the channel group and time slots to be mapped.	`Gateway(config-controller)#` **`channel-group`** *`channel_number`* **`timeslots`** *`range`*
Step 7	Return to global configuration mode.	`Gateway(config-controller)#` **`exit`**

Step 2: Configure the Internet Key Exchange Security Protocol

The second step is to establish an Internet Key Exchange (IKE) Security Protocol for encryption.

The Internet Key Exchange (IKE) protocol is a key management protocol standard that is used in conjunction with the IPSec standard. IPSec is an IP security feature that provides robust authentication and encryption of IP packets. IPSec can be configured without IKE, but IKE enhances IPSec by providing additional features, flexibility, and ease of configuration for the IPSec standard. (For more information on IPSec, see the "Step 3: Configure IPSec Network Security" section on page 11-5.)

To configure an IKE Security Protocol, follow this procedure:

	Task	Command
Step 1	Create an IKE policy[1] with a unique priority number and enter Internet Security Association and Key Management Protocol (ISAKMP[2]) policy configuration mode. **Note** You can configure multiple policies on each peer[3]. At least one of these policies must contain exactly the same encryption, authentication, and other parameters as one of the policies on the remote peer.	`Gateway(config)# crypto isakmp policy priority`
Step 2	Specify the authentication method to be used in an IKE policy.	`Gateway(config-isakmp)# authentication {rsa-sig|rsa-encr|pre-share}`
Step 3	Return to global configuration mode.	`Gateway(config-isakmp)# exit`
Step 4	Configure the authentication key for each peer that shares a key.	`Gateway(config)# crypto isakmp key keystring address peer_address|peer_hostname`

1. You must create IKE policies at each peer. An IKE policy defines a combination of security parameters to be used during the IKE negotiation. IKE negotiations must be protected, so each IKE negotiation begins by each peer agreeing on a common (shared) IKE policy. This policy states which security parameters will be used to protect subsequent IKE negotiations. After the two peers agree upon a policy, the security parameters of the policy are identified by a security association established at each peer, and these security associations apply to all subsequent IKE traffic during the negotiation.
2. A protocol framework that defines payload formats, the mechanics of implementing a key exchange protocol, and the negotiation of a security association.
3. In the context of this document, a peer refers to a Catalyst 4224 or other device that participates in IPSec and IKE.

For information on how to create a private or public key and to download a certificate, visit the following website:

http://www.cisco.com/univercd/cc/td/doc/product/software/ios121/121cgcr/secur_c/scprt4/scdipsec.htm

Step 3: Configure IPSec Network Security

The third step is to define how the T1 data will be handled. This requires that you use IPSec (IP Security Protocol) security.

IPSec is a framework of open standards that provides data confidentiality, data integrity, and data authentication between participating peers. IPSec provides these security services at the IP layer. IPSec uses IKE to handle negotiation of protocols and algorithms based on local policy, and to generate the encryption and authentication keys to be used by IPSec. IPSec can be used to protect one or more data flows between a pair of hosts, between a pair of security gateways, or between a security gateway and a host.

To configure IPSec network security, follow this procedure:

	Task	Command
Step 1	Specify the lifetime of a security association[1]. As a general rule, the shorter the lifetime (up to a point), the more secure your IKE negotiations will be. However, with longer lifetimes, future IPSec security associations can be set up more quickly. The default lifetimes are 3600 seconds (one hour) and 4608000 kilobytes (10 megabytes per second for one hour).	`Gateway(config)# crypto ipsec security-association lifetime seconds` *seconds* `kilobytes` *kilobytes*
Step 2	Specify a transform set[2] and enter transform-set configuration mode. To define a transform set, specify one to three "transforms"—each *transform* represents an IPSec security protocol (ESP or AH) plus the algorithm you want to use. When the particular transform set is used during negotiations for IPSec security associations, the entire transform set (the combination of protocols, algorithms and other settings) must match a transform set at the remote peer.	`Gateway(config)# crypto ipsec transform-set` *transform_set_name transform1 [transform2 [transform3]]*
Step 3	Return to global configuration mode.	`Gateway(cfg-crypto-trans)# exit`

	Task	Command
Step 4	Create a crypto map[3] denoted by *map-name*. Enter crypto map configuration mode, unless you use the dynamic keyword. *seq-num* is the number you assign to the crypto map entry. **ipsec-isakmp** indicates that IKE will be used to establish the IPSec security associations for protecting the traffic specified by this crypto map entry. **dynamic** is an optional argument specifying that this crypto map entry references a preexisting dynamic crypto map. Dynamic crypto maps are policy templates used in processing negotiation requests from a peer IPSec device. If you use this keyword, none of the crypto map configuration commands will be available. *dynamic-map-name* specifies the name of the dynamic crypto map set that should be used as the policy template.	`Gateway(config)# crypto map map_name seq_num ipsec-isakmp [dynamic dynamic_map_name] [discover]`
Step 5	Specify the same remote IPSec peer that you specified in Step 4 in the previous procedure, "Step 2: Configure the Internet Key Exchange Security Protocol" section on page 11-3.	`Gateway(config-crypto map)# set peer hostname|ip_address`
Step 6	For this crypto map entry, specify the same transform set that you specified in Step 2 of this procedure.	`Gateway(config-crypto map)# set transform-set transform_set_name`
Step 7	Specify an extended access list for a crypto map entry. This value should match the access-list-number or name argument of the extended access list.	`Gateway(config-crypto map)# match address [access_list_id | name]`

	Task	Command
Step 8	Return to global configuration mode.	`Gateway(cfg-crypto-trans)# exit`
Step 9	Create an access list.[4] access_list_number denotes an IP list number from 1 through 99. **permit** or **deny** specifies permit or deny condition for this list. IP-address is the IP address to which the router compares the address being tested. wild-mask is the wildcard mask bits for the address in 32-bit, dotted decimal notation.	`Gateway(config)# access-list access_list_number {permit | deny} {type_code wild_mask | address mask}`

1. A security association (SA) describes how two or more entities will utilize security services to communicate securely. For example, an IPSec SA defines the encryption algorithm (if used), the authentication algorithm, and the shared session key to be used during the IPSec connection. Both IPSec and IKE require and use SAs to identify the parameters of their connections. IKE can negotiate and establish its own SA. The IPSec SA is established either by IKE or by manual user configuration.
2. A transform set represents a specific combination of security protocols and algorithms. During the IPSec security association negotiation, the peers search for a transform set that is the same on both peers. When such a transform set is found, it is selected and applied to the protected traffic as part of both peers' IPSec security associations.
3. With IPSec you define what traffic should be protected between two IPSec peers by configuring access lists and applying these access lists to interfaces by way of crypto map sets. A crypto map set can contain multiple entries, each with a different access list. The crypto map entries are searched in order, and the Catalyst 4224 attempts to match the packet to the access list specified in that entry.
4. Packet filtering helps control packet movement through the network. Such control can help limit network traffic and restrict network use by certain users or devices. To permit or deny packets from crossing specified interfaces, Cisco provides access lists. An access list is a sequential collection of permit and deny conditions that apply to IP addresses.

Step 4: Configure Encryption on the T1 Channel Group Serial Interface

The fourth step is to configure a T1 serial interface with an IP address and a crypto map.

To configure encryption on the T1 channel group, follow this procedure:

	Task	Command
Step 1	Select the serial interface and enter interface configuration mode.	`Gateway (config)# interface serial slot|port:timeslot`
Step 2	Specify an IP address followed by the subnet mask for this interface.	`Gateway (config-if)# ip address address mask`
Step 3	Assign a crypto map to this interface.	`Gateway (config-if)# crypto map map_name`
Step 4	Return to global configuration mode.	`Gateway(config-if)# exit`
Step 5	Return to the enable mode.	`Gateway(config)# exit`
Step 6	Display the current operating configuration, including any changes just made.	`Gateway# show running-config`
Step 7	Display the configuration currently stored in nonvolatile random-access memory (NVRAM).	`Gateway# show startup-config`
Step 8	At the enable prompt, write your changes to NVRAM. **Note** The results of the **show running-config** and **show startup-config** commands differ if you have made changes to the configuration but have not yet written them to NVRAM.	`Gateway# copy running-config startup-config`

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

Verifying the Configuration

After configuring the new interface, use the following commands to verify that it is operating correctly:

- Use **show version** to display the router hardware configuration. Check that the list includes the new interface.
- Use **show controllers** to display all network modules and their interfaces.
- Use **show interfaces** [*type slot/port*] to display the details of a specified interface. Verify that the first line of the display shows the correct slot and port number and that the interface and line protocol are in the correct state (up or down).
- Use **show protocols** to display the protocols configured for the entire router and for individual interfaces. If necessary, add or remove protocol routing on the router or its interfaces.
- Use **show running-config** to display the running configuration.
- Use **show startup-config** to display the configuration stored in NVRAM.
- Use **ping** to send an echo request to a specified IP address.

Note Encryption is enabled by default when you install the ESA hardware. If you need to enable encryption, use the **no crypto engine accel** command. This command is useful for debugging problems with the ESA or for testing features available only with software encryption.

Sample Configurations

This section contains the following topics:

Encrypting Traffic Between Two Networks

The sample configurations in this section show you how to encrypt traffic between a private network (10.103.1.x) and a public network (98.98.98.x) using IPSec. The 98.98.98.x network knows the 10.103.1.x network by the private addresses. The 10.103.1.x network knows the 98.98.98.x network by the public addresses.

Configuration File for the Public Gateway

```
gateway-2b# show running config
Building configuration...

Current configuration:
!
version 12.0
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname gateway-2b
!
ip subnet-zero
!
ip audit notify log
ip audit po max-events 100
!
crypto isakmp policy 1
hash md5
authentication pre-share
crypto isakmp key cisco123 address 95.95.95.2
!
crypto ipsec transform-set rtpset esp-des esp-md5-hmac
!
crypto map rtp 1 ipsec-isakmp
set peer 95.95.95.2
set transform-set rtpset
match address 115
!
interface Ethernet0/0
ip address 98.98.98.1 255.255.255.0
no ip directed-broadcast
!
interface Ethernet0/1
ip address 99.99.99.2 255.255.255.0
```

```
no ip directed-broadcast
no ip route-cache
no ip mroute-cache
crypto map rtp
!
interface Ethernet0/2
no ip address
no ip directed-broadcast
shutdown
!
interface Ethernet0/3
no ip address
no ip directed-broadcast
shutdown
!
ip classless
ip route 0.0.0.0 0.0.0.0 99.99.99.1
no ip http server
!
access-list 115 permit ip 98.98.98.0 0.0.0.255 10.103.1.0 0.0.0.255
access-list 115 deny ip 98.98.98.0 0.0.0.255 any
!
line con 0
transport input none
line aux 0
line vty 0 4
login
!
end
```

Configuration File for the Private Gateway

```
gateway-6a# show running config
Building configuration...

Current configuration:
!
version 12.0
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname gateway-6a
!
enable secret 5 $1$S/yK$RE603ZNv8N71GDYDbdMWd0
enable password ww
```

```
!
ip subnet-zero
!
ip audit notify log
ip audit PO max-events 100
isdn switch-type basic-5ess
isdn voice-call-failure 0
!

crypto isakmp policy 1
hash md5
authentication pre-share
crypto isakmp key cisco123 address 99.99.99.2
!
crypto ipsec transform-set rtpset esp-des esp-md5-hmac
crypto map rtp 1 ipsec-isakmp
set peer 99.99.99.2
set transform-set rtpset
match address 115
!
interface Ethernet0/0
no ip address
no ip directed-broadcast
!
interface Serial0/0
no ip address
no ip directed-broadcast
no ip mroute-cache
shutdown
!
interface Ethernet0/1
no ip address
no ip directed-broadcast
!
interface Serial0/1
no ip address
no ip directed-broadcast
shutdown
!
interface BRI1/0
no ip address
no ip directed-broadcast
shutdown
isdn switch-type basic-5ess
!
interface Ethernet1/0
no ip address
no ip directed-broadcast
```

```
shutdown
!
interface Serial1/0
no ip address
no ip directed-broadcast
shutdown
!
interface TokenRing1/0
no ip address
no ip directed-broadcast
shutdown
ring-speed 16
!
interface Ethernet3/0
ip address 95.95.95.2 255.255.255.0
no ip directed-broadcast
ip nat outside
no ip route-cache
no ip mroute-cache
crypto map rtp
!
interface Ethernet3/1
no ip address
no ip directed-broadcast
shutdown
!
interface Ethernet3/2
ip address 10.103.1.75 255.255.255.0
no ip directed-broadcast
ip nat inside
!
interface Ethernet3/3
no ip address
no ip directed-broadcast
shutdown
!
ip nat pool FE30 95.95.95.10 95.95.95.10 netmask 255.255.255.0
ip nat inside source route-map nonat pool FE30 overload
ip classless
ip route 0.0.0.0 0.0.0.0 95.95.95.1
ip route 171.68.120.0 255.255.255.0 10.103.1.1
no ip http server
!
access-list 110 deny ip 10.103.1.0 0.0.0.255 98.98.98.0 0.0.0.255
access-list 110 permit ip 10.103.1.0 0.0.0.255 any
access-list 115 permit ip 10.103.1.0 0.0.0.255 98.98.98.0 0.0.0.255
access-list 115 deny ip 10.103.1.0 0.0.0.255 any
dialer-list 1 protocol ip permit
```

```
dialer-list 1 protocol ipx permit
route-map nonat permit 10
match ip address 110
!
tftp-server flash:cgateway-io3s56i-mz.120-7.T
!
line con 0
transport input none
line 65 72
line aux 0
line vty 0 4
password WW
login
!
end
```

Exchanging Encrypted Data Through an IPSec Tunnel

This section contains sample configuration files for two peer Catalyst 4224s set up to exchange encrypted data through a secure IPSec tunnel over a channelized T1 interface channel group, serial 1/0:0.

Configuration File for Peer 1

```
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Rose
!
logging buffered 100000 debugging
enable password lab
!
ip subnet-zero
no ip domain-lookup
!
crypto isakmp policy 10
 authentication pre-share
crypto isakmp key pre-shared address 6.6.6.2
!
crypto ipsec security-association lifetime seconds 86400
!
```

```
crypto ipsec transform-set transform-1 esp-des
!
crypto map cmap 1 ipsec-isakmp
 set peer 6.6.6.2
 set transform-set transform-1
 match address 101
!
controller T1 1/0
 framing esf
 linecode b8zs
 channel-group 0 timeslots 1-23 speed 64
 channel-group 1 timeslots 24 speed 64
!
controller T1 1/1
 channel-group 0 timeslots 1-23 speed 64
 channel-group 1 timeslots 24 speed 64
!
process-max-time 200
!
interface FastEthernet0/0
 ip address 111.0.0.2 255.0.0.0
 no ip directed-broadcast
 no ip route-cache
 no ip mroute-cache
 speed 10
!

interface Serial0/0
 no ip address
 no ip directed-broadcast
 shutdown
!
interface FastEthernet0/1
 ip address 4.4.4.1 255.0.0.0
 no ip directed-broadcast
 no ip route-cache
 no ip mroute-cache
 load-interval 30
 speed 10
!
interface Serial1/0:0
 bandwidth 1472
 ip address 6.6.6.1 255.0.0.0
 no ip directed-broadcast
 encapsulation ppp
 no ip route-cache
 load-interval 30
 no fair-queue
```

```
 crypto map cmap
!
interface Serial1/0:1
 no ip address
 no ip directed-broadcast
 fair-queue 64 256 0
!
interface Serial1/1:0
 no ip address
 no ip directed-broadcast
!
interface Serial1/1:1
 no ip address
 no ip directed-broadcast
 fair-queue 64 256 0
!
router rip
 network 4.0.0.0
 network 6.0.0.0
!
ip classless
ip route 0.0.0.0 0.0.0.0 111.0.0.1
no ip http server
!
access-list 101 deny   udp any eq rip any
access-list 101 deny   udp any any eq rip
access-list 101 permit ip 6.6.6.0 0.0.0.255 6.6.6.0 0.0.0.255
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 password lab
 login
!
end
```

Configuration File for Peer 2

```
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Peony
```

```
!
logging buffered 100000 debugging
enable password lab
!
ip subnet-zero
no ip domain-lookup
!
crypto isakmp policy 10
 authentication pre-share
crypto isakmp key pre-shared address 6.6.6.1
!
crypto ipsec security-association lifetime seconds 86400
!
crypto ipsec transform-set transform-1 esp-des
!
crypto map cmap 1 ipsec-isakmp
 set peer 6.6.6.1
 set transform-set transform-1
 match address 101
!
controller T1 1/0
 framing esf
 linecode b8zs
 channel-group 0 timeslots 1-23 speed 64
 channel-group 1 timeslots 24 speed 64
!
controller T1 1/1
 channel-group 0 timeslots 1-23 speed 64
 channel-group 1 timeslots 24 speed 64
!
process-max-time 200
!
interface FastEthernet0/0
 ip address 172.0.0.13 255.0.0.0
 no ip directed-broadcast
 no ip mroute-cache
 load-interval 30
 no keepalive
 speed 10
!
interface FastEthernet0/1
 ip address 3.3.3.2 255.0.0.0
 no ip directed-broadcast
 no ip route-cache
 no ip mroute-cache
 load-interval 30
 speed 10
!
```

```
interface Serial1/0:0
 bandwidth 1472
 ip address 6.6.6.2 255.0.0.0
 no ip directed-broadcast
 encapsulation ppp
 no ip route-cache
 load-interval 30
 no fair-queue
 crypto map cmap
!

interface Serial1/0:1
 no ip address
 no ip directed-broadcast
 fair-queue 64 256 0
!
interface Serial1/1:0
 no ip address
 no ip directed-broadcast
!
interface Serial1/1:1
 no ip address
 no ip directed-broadcast
 fair-queue 64 256 0
!
router rip
 network 3.0.0.0
 network 6.0.0.0
!
ip classless
ip route 0.0.0.0 0.0.0.0 111.0.0.1
no ip http server
!
access-list 101 deny   udp any eq rip any
access-list 101 deny   udp any any eq rip
access-list 101 permit ip 6.6.6.0 0.0.0.255 6.6.6.0 0.0.0.255
!
line con 0
 exec-timeout 0 0
 transport input none
line aux 0
line vty 0 4
 login
!!
end
```

CHAPTER 12

Configuring Other Routing Protocols

Cisco IOS software on the Catalyst 4224 Access Gateway Switch supports the following additional routing protocols:

Novell IPX

Novell Internetwork Packet Exchange (IPX) is derived from the Xerox Network Systems (XNS) Internet Datagram Protocol (IDP). IPX and XNS have the following differences:

- IPX and XNS do not always use the same Ethernet encapsulation format.
- IPX uses the Novell proprietary Service Advertising Protocol (SAP) to advertise special network services. File servers and print servers are examples of services that typically are advertised.
- IPX uses delay (measured in ticks) while XNS uses hop count as the primary metric in determining the best path to a destination.

The Cisco Implementation of Novell IPX

The Cisco implementation of the Novell IPX protocol is certified to provide full IPX routing functionality.

IPX MIB Support

Cisco supports the IPX MIB (currently, read-only access is supported). The IPX Accounting group represents one of the local Cisco-specific IPX variables Cisco supports. This group provides access to the active database that is created and maintained if IPX accounting is enabled on a router or access server.

IPX Enhanced IGRP Support

Cisco IOS software supports IPX Enhanced IGRP, which provides the following features:

- Automatic redistribution—IPX Routing Information Protocol (RIP) routes are automatically redistributed into Enhanced IGRP, and Enhanced IGRP routes are automatically redistributed into RIP. If desired, you can turn off redistribution. You also can completely turn off Enhanced IGRP and IPX RIP on the device or on individual interfaces.
- Increased network width—With IPX RIP, the largest possible width of your network is 15 hops. When Enhanced IGRP is enabled, the largest possible width is 224 hops. Because the Enhanced IGRP metric is large enough to support thousands of hops, the only barrier to expanding the network is the transport layer hop counter. Cisco works around this problem by incrementing the transport control field only when an IPX packet has traversed 15 routers, and when the next hop to the destination was learned via Enhanced IGRP. When a RIP route is being used as the next hop to the destination, the transport control field is incremented as usual.
- Incremental SAP updates—Complete SAP updates are sent periodically on each interface until an Enhanced IGRP neighbor is found, and thereafter only when changes are made to the SAP table. This procedure works by taking advantage of the Enhanced IGRP reliable transport mechanism, which means that an Enhanced IGRP peer must be present for incremental SAPs to be sent. If no peer exists on a particular interface, periodic SAPs will be sent on that interface until a peer is found. This functionality is automatic on serial interfaces and can be configured on LAN media.

LAN Support

Cisco IOS software supports routing IPX between Ethernet-emulated LANs and Token Ring-emulated LANs. For more information on emulated LANs and routing IPX between them, refer to the "Configuring LAN Emulation" chapter of the *Cisco IOS Switching Services Configuration Guide*.

VLAN Support

Cisco IOS software supports routing IPX between VLANs. Users with Novell NetWare environments can configure any one of the four IPX Ethernet encapsulations to be routed using ISL encapsulation across VLAN boundaries. For more information on VLANs and routing IPX between them over ISL, refer to the "Configuring Routing Between VLANs with ISL Encapsulation" chapter of the *Cisco IOS Switching Services Configuration Guide*.

Multilayer Switching Support

Cisco IOS software supports IPX Multilayer Switching (MLS). For more information on IPX MLS, refer to the "Multilayer Switching" chapter of the *Cisco IOS Switching Services Configuration Guide*.

IPX Configuration

For details on how to configure IPX protocol, refer to the following documentation, available online at Cisco.com:

- *Cisco IOS AppleTalk and Novell IPX Configuration Guide, Release 12.2*

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fatipx_c/index.htm

- *Cisco IOS AppleTalk and Novell IPX Command Reference, Release 12.2*

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fatipx_r/index.htm

IBM SNA

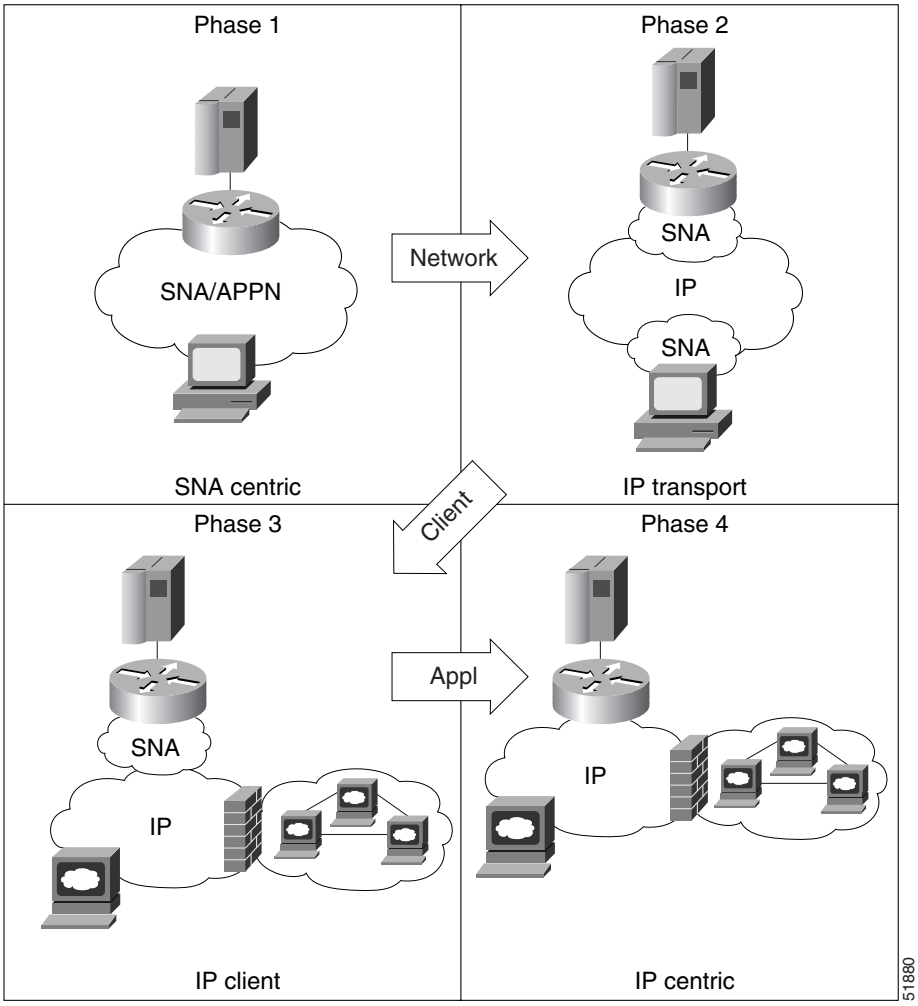
Adopting a TCP/IP infrastructure is the first logical step to creating a multiservice network that seamlessly accommodates data, voice, and video.

Enterprise organizations are heavily invested in mainframes and Systems Network Architecture (SNA) and mainframes are still a vital part of enterprise data centers. The goal for these enterprise organizations is to integrate the TCP/IP-based environment with the SNA-based environment. Cisco bridging and IBM networking technologies enable the delivery of SNA data over routers supporting TCP/IP.

The Cisco Four-Phase Model for SNA-to-IP Integration

Cisco has developed a high-level, four-phase model illustrating a typical integration path to incorporate TCP/IP into an SNA-based network. Figure 12-1 illustrates the four-phase integration path. The model helps to describe some common phases in SNA-to-IP integration. A single phase in this integration path might represent the network of some organizations, while two or more phases might represent the network implementation of other organizations in various sectors of their network.

Figure 12-1 The Cisco Four-Phase SNA-to-IP Integration Model



The phases can be differentiated by the protocol that runs in each of three key elements in the network: the mainframe/midrange computer, the network backbone, and the desktop. This section describes the characteristics of each of the phases along with the problems solved, types of products and technologies implemented, and challenges.

This section contains the following topics:

Phase One: SNA Centric

An SNA-centric network has SNA, Advanced Peer-to-Peer Networking (APPN), or APPN/High Performance Routing (HPR), protocols running on one or more mainframe/midrange systems.

An SNA-centric network is a very high-speed and dynamic network when compared to the traditional SNA network of the past. ACF/VTAM on the mainframe includes APPN/HPR protocols to support dynamic rerouting around failures and high-speed switching in the network. The mainframe complex, which now comprises multiple complementary metal-oxide semiconductor (CMOS) processors, implements Parallel Sysplex to provide the ultimate in redundancy and session persistence.

The FEP has often been replaced by a high-performance, channel-connected router such as the Channel Interface Processor (CIP) or the Channel Port Adapter (CPA). The network backbone comprises high-speed switches (ATM, Ethernet/Fast Ethernet/Gigabit Ethernet, or Token Ring) and routers running APPN/HPR. Shared Token Ring LANs are being replaced with Token Ring or Ethernet switching to the desktop, offering a dedicated LAN segment and bandwidth to each end user.

Most desktops have PCs running advanced SNA client emulation software such as TN3270 Server. Routers provide support, via features such as Dependent Logical Unit Requester (DLUR) and downstream physical unit (DSPU) concentration, to transport the traffic from the remaining traditional SNA terminals and controllers.

Phase Two: IP Transport

Beginning in the 1980s, large organizations began building TCP/IP-based networks to support client/server applications and systems. UNIX, a dominant operating system for client/server applications, natively supports TCP/IP. As the growth of TCP/IP-based systems continued, organizations often found that they had built parallel networks, one running SNA and one running TCP/IP. This setup is expensive because of the duplication of line costs, equipment, and personnel. To eliminate the duplication, organizations had a choice—run the TCP/IP traffic over the SNA backbone, or run the SNA traffic over the TCP/IP backbone.

Running TCP/IP over an SNA backbone was not a feasible choice because of the lack of redundancy and openness of SNA. Routers, which formed the core of the TCP/IP network, began to support the encapsulation of SNA in TCP/IP for transport across the TCP/IP network.

This encapsulation brings many benefits. First and foremost, while it is encapsulated in TCP/IP, the SNA traffic is dynamically routed around network failures, a benefit that only recently has been added to SNA networks with APPN/HPR. The encapsulation schemes also provide more flexible configurations for SNA devices and reduced polling traffic across the backbone. Cisco offered the first such encapsulation scheme with RSRB. Since then, the industry has adopted a standard, data-link switching (DLSw), that has been very widely accepted and implemented. Routers also provide features such as serial tunnel (STUN) and Block Serial Tunneling (BSTUN) to encapsulate other types of traffic (asynchronous, bisynchronous, and some proprietary protocols) in addition to SNA.

In this second phase of integration, many organizations find that the same end users who are running advanced SNA client emulators to access mainframe and midrange systems are also accessing TCP/IP systems. This means that each PC must run two different protocol stacks—SNA and TCP/IP—for access to host systems.

Phase Three: IP Client

In the third phase of SNA-to-IP integration, organizations eliminated the dual protocol stacks at end-user PCs by implementing emulation software that supports TCP/IP. The same rich functionality that end users relied on in their emulation software remains the same, only it now runs over a TCP/IP stack.

Cisco Transaction Connection (CTRC) provides TCP/IP end-users and servers with direct access to Customer Information Control System (CICS) and IBM DB2 databases. Organizations achieve protocol independence between end-users and hosts, enabling applications to communicate directly to DB2 or CICS without upgrades.

TN3270(E), TN5250, Distributed Relational Database Architecture (DRDA) and Inter-System Communications (ISC) protocol are widely implemented and widely accepted standards for achieving TCP/IP-based access to mainframes and AS/400s. TN3270 Server technology on the router provides support for the TN3270(E) clients. CTRC on the router supports access to IBM DB2 databases from ODBC and JDBC drivers. CTRC also supports access to transaction programs managed by IBM's CICS. In addition to eliminating a second protocol from each desktop, organizations reap the following benefits by implementing low-cost, standards-based solutions such as TN3270(E), TN5250, and CTRC:

- Availability of high-performance servers. Very high-capacity and high-performance gateway servers are available that offload the protocol processing of TN3270(E) or TN5250 from the mainframe or midrange host. These servers replace the low-capacity PC gateways that are based on proprietary gateway protocols.
- Integration with corporate intranet. Because the desktop is based upon TCP/IP, all the advances taking place in corporate intranets can be brought to mainframe and midrange connections. For example, virtual private networks (VPNs) can be created for secure remote host access. Encryption and authentication can become a new level of security for host access.
- Access from a browser. A whole new market, the Web-to-host market, is emerging that allows end users to access host systems using the browser as the standard interface. This setup brings enormous benefits by reducing the software distribution and administration chores for emulation software and this sets the stage for a new, browser-style interface to older applications. Organizations can look to these mission-critical applications to extend new services to their customers, as in the case of home banking, citizen access to government records, and insurance company applications.

Phase Four: IP Centric

In the fourth and final stage of SNA-to-IP integration, the mainframe and midrange systems natively support TCP/IP. They share files with and transfer data to other, non-SNA systems. Corporate databases are securely accessed in a standard way from a variety of different end-user applications. The remaining applications that are based on traditional "green-on-black," character-based terminals are accessed transparently through standard emulation screens or through intuitive, user-friendly Web pages.

TCP/IP-based mainframe and midrange systems offer advanced redundancy and high-availability features similar to those provided to SNA-based applications today. With the full, native support of TCP/IP, the mainframe and midrange systems can be fully participating members in the corporate intranet.

Summary of Four-Phase Model

The four-phase model of SNA-to-IP integration is based on Cisco experience helping to integrate some of the world's largest and most complex SNA networks. In reality, very few organizations go through a stepwise, linear migration from SNA centric to IP transport, to IP client, to IP centric.

For example, many large organizations have run TCP/IP stacks on their mainframes for years, alongside ACF/VTAM, whether they have implemented TCP/IP in the enterprise backbone network or not. Most large organizations will find elements from all four phases represented somewhere in their network. The model, however, is useful to describe the various issues of SNA-to-IP integration, their common solutions, and the characteristics of the network at various points in the change.

Scenarios for SNA-to-IP Integration

There are common elements or scenarios for integrating TCP/IP with SNA networks. This section describes three elements or scenarios, the corresponding phase from the Cisco four-phase integration model, and the Cisco products and software features deployed in these scenarios. This section discusses the following scenarios:

Line Consolidation

Line consolidation involves simplifying the network by providing a single network infrastructure, based on TCP/IP. This structure accommodates SNA and other traffic and allows the elimination of multiple single-protocol lines to each location.

Phase two of SNA-to-IP integration dictates the building of a single network backbone based upon TCP/IP. This setup often allows organizations to consolidate the number of communication lines in the network which simplifies the support and maintenance.

The primary product in a line consolidation project is a multiprotocol router that encapsulates and converts the traffic from the SNA lines. RSRB and DLSw+ are the Cisco IOS technologies used for this conversion. In addition, Cisco routers also support the tunneling of both bisynchronous and certain asynchronous protocols with Cisco IOS features such as STUN and BSTUN and the Airline Product Set (ALPS).

FEP Replacement

FEP replacement involves replacing FEPs (and possibly other special-purpose mainframe channel-attached equipment) with new channel-attached routers that offer high throughput, low costs, and flexible software functionality.

Throughout all phases of the SNA-to-IP integration, high-capacity throughput to the mainframe is a key requirement. Organizations are replacing FEPs with routers with direct channel attachments.

The primary product in a FEP replacement project is a channel-attached router. This router contains the mainframe channel connection hardware supporting either a bus-and-tag or ESCON interface (or multiple interfaces). It also runs the necessary channel protocol software and, in some cases, special software designed to offload communication processing from the mainframe. For example, the Cisco CIP and CPA both support TCP Offload, TCP Assist, CTRC, and TN3270 Server features to offload mainframe cycles.

Desktop Consolidation

In a desktop consolidation, desktops running multiple protocol stacks are simplified to utilize TCP/IP for access to all resources, including mainframes and AS/400s. This consolidation can be accomplished using traditional emulators that utilize TCP/IP instead of SNA for host communication, or it can be accomplished by leveraging new browser-based access approaches.

Phases three and four of the SNA-to-IP integration require end users to access host systems using TCP/IP.

The primary products in a desktop consolidation project are desktop devices, desktop software, and new gateway servers. Other products that may be considered for deployment are additional load-balancing domain name servers, firewalls, and other security devices.

Terminal emulation is, by definition, a client/server implementation. That is, PCs running terminal emulation software communicate with gateway software using either a proprietary or a standard protocol that is at a higher level than the TCP/IP transport. These gateways then communicate directly with the host applications using standard SNA protocols.

Most terminal emulators offer multiple choices of gateway connectivity. The only standard TCP/IP-based protocols for communication to mainframe and midrange systems are TN3270(E) and TN5250, respectively. Many organizations are implementing TN3270 and TN5250 because they are standards and they set the stage for Web-to-host solutions.

SNA Configuration

For details on how to configure SNA protocol, refer to the following documentation:

- *Cisco IOS Bridging and IBM Networking Configuration Guide, Release 12.2*

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fibm_c/index.htm

- *Cisco IOS Bridging and IBM Networking Command Reference, Volume 1, Release 12.2*

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fibm_r1/index.htm

- *Cisco IOS Bridging and IBM Networking Command Reference, Volume 2, Release 12.2*

  http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fibm_r2/index.htm

Command Reference for Voice VLAN

This section provides reference information for the following voice VLAN commands:

interface range

The **interface range** command allows you to configure multiple interfaces with the same configuration parameters. By using this command, you enter range configuration mode. While you are in this mode, the command parameters that you enter are attributed to all interfaces within the range specified.

Syntax

interface range {{ethernet | fastethernet} *slot/interface - interface***}**
[, {{ethernet | fastethernet} *slot/interface - interface***}]**

Syntax Description

slot

Slot number.

interface

Interface number.

Defaults

None.

Command Modes

Global configuration

Usage Guidelines

The space before the dash is required. You can enter up to five comma-separated ranges. You are not required to enter spaces before or after the comma.

If you enter multiple configuration commands while you are in interface range mode, each command is executed as it is entered. If you exit interface range configuration mode while the commands are being executed, some commands may not be executed on all interfaces in the range. Wait until the command prompt reappears before exiting interface range configuration mode.

Example

The following example shows how to select the Fast Ethernet interfaces 5/1 to 5/5:

```
Gateway (config)# interface range fastethernet 5/1 - 5
```

interface vlan

Use the **interface vlan** configuration command to configure an interface type, create a switch virtual interface to be used as the routed VLAN interface, and to enter interface configuration mode.

Syntax

interface vlan *number*

no interface vlan *number*

Syntax Description

number

Valid VLAN IDs are from 1 to 1005. Do not enter leading zeroes.

Defaults

The default management VLAN interface is VLAN 1.

Command Modes

Global configuration.

Usage Guidelines

To enable inter-VLAN routing, configure an interface VLAN with a VLAN number.

Example

The following example shows how to enter configuration mode for vlan 3:

```
Switch# configure terminal
Switch(config)# interface vlan 3
Switch(config-if)#
```

monitor session

Use the **monitor session** configuration command to set a port session.

Syntax

monitor session *number destination_interface*

monitor session *number source_interface*

Syntax Description

number

Session number

destination_interface

SPAN destination interface

source_interface

SPAN source interface

Defaults

None.

Command Modes

Interface configuration.

Usage Guidelines

You can span only selected interfaces. You cannot span VLANs.

Examples

The following example shows how to set up a port monitor session. For this example, we configure two sessions in global configuration mode.

In session 1, int Fas 5/2 will mirror all of its packets to 5/9:

```
sjc12-42a-sw1(config)#monitor session 1 source interface fast 5/2
sjc12-42a-sw1(config)#monitor session 1 dest int fas 5/9
```

In session 2, int fas 5/3 will mirror all of its packets to 5/13:

```
sjc12-42a-sw1(config)#monitor session 2 source int fas 5/3
sjc12-42a-sw1(config)#monitor session 2 destination int fas 5/13
```

You can sees what is being mirrored by entering the **show monitor** *session* command in privileged EXEC mode:

```
sjc12-42a-sw1#sho monitor
Session 1
Source Ports:
    Both:         FastEthernet5/2
Destination Ports:FastEthernet5/13

Session 2
Source Ports:
    Both:         FastEthernet5/3
Destination Ports:FastEthernet5/13
```

spanning-tree

Use the **spanning-tree** global configuration command to enable Spanning Tree Protocol (STP) on a VLAN. Use the **no** form of the command to disable STP on a VLAN.

Syntax

spanning-tree [*vlan stp-list*]

no spanning-tree [*vlan stp-list*]

Syntax Description

vlan stp-list

(Optional) List of spanning-tree instances. Each spanning-tree instance is associated with a VLAN ID. Valid IDs are from 1 to 1005. Enter each VLAN ID separated by a space. Do not enter leading zeroes. Ranges are not supported.

Defaults

STP is enabled.

Command Modes

Global configuration.

Usage Guidelines

Disabling STP causes the VLAN or list of VLANs to stop participating in STP. Ports that are administratively down remain down. Received Bridge Protocol Data Units (BPDUs) are forwarded like other multicast frames. The VLAN does not detect and prevent loops when STP is disabled.

You can disable STP on a VLAN that is not currently active, and verify the change by using the privileged EXEC **show running-config** or the **show spanning-tree vlan** *stp-list* command. The setting takes effect when the VLAN is activated.

If the variable *stp-list* is omitted, the command applies to the STP instance associated with VLAN 1.

You can enable STP on a VLAN that has no ports assigned to it.

Example

The following example shows how to disable STP on VLAN 5:

```
Switch(config)# no spanning-tree vlan 5
```

You can verify the previous command by entering the **show spanning-tree** command in privileged EXEC mode. In this instance, VLAN 5 does not appear in the list.

spanning-tree portfast

Use the **spanning-tree portfast** interface configuration command to enable the Port Fast feature on a port in all its associated VLANs. When the Port Fast feature is enabled, the port changes directly from a blocking state to a forwarding state without making the intermediate STP status changes. Use the **no** form of this command to return the port to default operation.

Syntax

spanning-tree portfast

no spanning-tree portfast

Syntax Description

This command has no keywords or arguments.

Defaults

The Port Fast feature is disabled; however, it is automatically enabled on dynamic-access ports.

Command Modes

Interface configuration.

Usage Guidelines

This feature should be used only on ports that connect to end stations. It affects all VLANs on the port.

A port with the Port Fast feature enabled is moved directly to the spanning-tree forwarding state.

Example

The following example shows how to enable the Port Fast feature on fixed port 2.

```
Switch(config-if)# spanning-tree portfast fa5/2
```

You can verify the previous command by entering the **show running-config** in privilege EXEC mode.

switchport access

Use the **switchport access** interface configuration command to configure a port as a static-access port. The port operates as a member of the configured VLAN.

Use the **no** form of this command to reset the access mode to the default VLAN for the switch.

Syntax

switchport access vlan *vlan-id*

no switchport access vlan *vlan-id*

Syntax Description

vlan *vlan-id*

ID of the VLAN. Valid IDs are from 1 to 1005. Do not enter leading zeroes.

Defaults

All ports are in static-access mode in VLAN 1.

Command Modes

Interface configuration.

Usage Guidelines

An access port can be assigned to only one VLAN.

When the **no switchport access vlan** form is used, the access mode is reset to static access on VLAN 1.

Example

The following example shows how to assign a port to VLAN 2 (instead of the default VLAN 1):

```
Switch(config-if)# switchport access vlan 2
```

You can verify the previous command by entering the **show interface** *interface-id* **switchport** command in privileged EXEC mode and examining information in the Administrative Mode and Operational Mode rows.

switchport voice vlan

Use the **switchport voice vlan** interface configuration command to configure the voice VLAN on the port. Use the **no** form of this command to return the setting to its default.

Syntax

switchport voice vlan {*vlan-id* **| dot1p | none | untagged}**

no switchport voice vlan

Syntax Description

vlan-id

VLAN used for voice traffic. Valid IDs are from 1 to 1001 (IDs 1002 to 4095 are not supported on Catalyst 4224 switches).

Do not enter leading zeroes.

dot1p

The telephone uses priority tagging and uses an 802.1Q VLAN ID of 0.

none

The telephone is not instructed through the CLI about the voice VLAN.

The telephone uses the configuration from the telephone key pad.

untagged

The telephone does not tag frames. The switch port can be an access port or an 802.1Q trunk port.

Defaults

The switch default is not to configure the telephone automatically (**none**).

The Cisco 7960 IP Phone default is to generate an 802.1Q/802.1P frame.

Command Modes

Interface configuration.

Usage Guidelines

Ports that are not configured as trunk ports but have a configured voice VLAN are access ports with a voice VLAN ID (VVID).

Example

The following example shows how to configure VLAN 2 as the voice VLAN:

```
Switch(config-if)# switchport voice vlan 2
```

You can verify the previous command by entering the **show interface** *interface-id* **switchport** command in privileged EXEC mode.

Synopsis of Basic VoIP Concepts

The Catalyst 4224 Access Gateway Switch (Catalyst 4224) provides Voice over IP (VoIP) gateway applications for a *micro branch* office. This chapter introduces some basic VoIP concepts.

This chapter contains these sections:

VoIP Overview

The VoIP application allows a Catalyst 4224 to convert analog voice signals such as telephone calls and faxes into digital IP packets and distribute these packets across a WAN. In VoIP technology, a digital signal processor (DSP) segments the voice signals into frames and stores them in packets. These packets are transported using IP in compliance with the International Telecommunication Union-Telecommunication Standardization Sector's (ITU-T's) specification H.323, the specification for transmitting multimedia (voice, video, and data) across a network.

Because VoIP is a delay-sensitive application, you need to fine-tune your network from end to end before implementing VoIP. Fine-tuning your network to support VoIP incorporates a series of protocols and features that improve quality of service (QoS). Furthermore, you must take traffic shaping into account to ensure the reliability of VoIP.

A Voice Primer

This section describes some basic telephony concepts that might help you understand VoIP:

How VoIP Processes a Typical Telephone Call

The general flow of a two-party call follows this process:

1. The caller picks up the handset. This signals an off-hook condition to the VoIP signaling application in the Catalyst 4224.
2. The session application issues a dial tone and waits for the caller to dial a telephone number.
3. The caller dials the telephone number. The session application stores the dialed digits.
4. After enough digits are accumulated to match a configured destination pattern, the telephone number is mapped to an IP host via the dial plan mapper. The IP host has a direct connection to either the destination telephone number or a private branch exchange (PBX) that is responsible for completing the call to the configured destination pattern.
5. The session application runs the H.323 session protocol to establish transmission and reception channels for each direction over the IP network. If the call is being handled by a PBX, the PBX forwards the call to the destination telephone. If Resource Reservation Protocol (RSVP) has been configured, RSVP reservations are put into effect to achieve the desired quality of service (QoS) over the IP network.

6. The coder-decoder compression schemes (codecs) are enabled for both ends of the connection using Real-Time Transport Protocol/User Datagram Protocol/Internet Protocol (RTP/UDP/IP) as the protocol stack.
7. Any call-progress indications (or other signals that can be carried inband) are cut through the voice path as soon as an end-to-end audio channel is established. Signaling that can be detected by the voice ports is also trapped by the session application at each end of the connection. Signaling carried over the IP network is encapsulated in Real-Time Transport Control Protocol (RTCP) using the RTCP application-defined (APP) extension mechanism.
8. When either person hangs up the phone, RSVP reservations are torn down (if RSVP is used) and the session ends. Each end becomes idle, waiting for the next off-hook condition to trigger another call setup.

Numbering Scheme

The standard Public Switched Telephone Network (PSTN) is a large, circuit-switched network. It uses a specific numbering scheme, which complies with the ITU-T's international public telecommunications numbering plan (E.164) recommendations. For example, in North America, the North American Numbering Plan (NANP) is used. NANP consists of an area code, an office code, and a station code:

- Area codes are assigned geographically.
- Office codes are assigned to specific switches.
- Station codes identify a specific port on that switch.

The format in North America is 1Nxx-Nxx-xxxx, where N = digits 2 through 9, and x = digits 0 through 9. Internationally, each country is assigned a one- to three-digit country code, and the country's dialing plan is dictated by the country code.

Analog Versus Digital

Analog transmission is not robust or efficient at recovering from line noise. Because analog signals degrade over distance, they need to be amplified periodically. This amplification boosts both the voice signal and the ambient line noise, resulting in degradation of the quality of the transmitted sound.

In response to the limitations of analog transmission, the telephony industry migrated to digital transmission using pulse code modulation (PCM) or adaptive differential PCM (ADPCM). In both cases, analog sound is converted into digital form by sampling the analog sound 8000 times per second and converting each sample into a numeric code.

codecs

PCM and ADPCM are examples of *waveform* codec and are compression techniques that exploit the redundant characteristics of the waveform itself. In addition to waveform codecs, there are source codecs that compress speech by sending only simplified parametric information about voice transmission. Thus, these codecs require less bandwidth. Source codecs include linear predictive coding (LPC), code-excited linear prediction (CELP) and multipulse-multilevel quantization (MP-MLQ).

Coding techniques for telephony and voice packet are standardized by the ITU-T in its G-series recommendations. The Catalyst 4224 uses the following coding standards:

- G.711—Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs.
- G.729—Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity, but both provide speech quality similar to 32-kbps ADPCM.

Mean Opinion Score

Each codec provides a certain quality of speech. The quality of transmitted speech is a subjective response of the listener. A common benchmark used to determine the quality of sound produced by specific codecs is the mean opinion score (MOS). With MOS, a wide range of listeners judge the quality of a voice sample (corresponding to a particular codec) on a scale of 1 (bad) to 5 (excellent). The scores are averaged to provide the MOS for that sample. Table B-1 shows the relationship between codecs and MOS scores.

Table B-1 Compression Methods and MOS Scores

Compression Method	Bit Rate (kbps)	Framing Size (ms)	MOS Score
G.711 PCM	64	0.125	4.1
G.729 CS-ACELP[1]	8	10	3.92
G.729 x 2 encodings[2]	8	10	3.27
G.729 x 3 encodings	8	10	2.68
G.729a[3] CS-ACELP	8	10	3.7

1. Conjugate structure-algebraic code-excited linear prediction.
2. A G.729 voice signal is tandem-encoded two times.
3. G.729 Annex A.

Although it might seem logical from a financial standpoint to convert all calls to low bit-rate codecs to save on infrastructure costs, you should be aware of the drawbacks of designing voice networks with low bit-rate compression. One of the main drawbacks is signal distortion due to multiple encodings (called tandem encodings). For example, when a G.729 voice signal is tandem-encoded three times, the MOS score drops from 3.92 (very good) to 2.68 (unacceptable). Another drawback of low bit-rate codecs is codec-induced delay.

Delay

One of the most important design considerations in implementing voice is minimizing one-way, end-to-end delay. Voice traffic is real-time traffic; if there is too long a delay in voice packet delivery, speech becomes unrecognizable. Delay is inherent in voice networking and is caused by a number of different factors. An acceptable delay is less than 200 milliseconds.

There are two kinds of delays inherent in today's telephony networks: propagation delay and handling delay.

Propagation delay is caused by the characteristics of the speed of light traveling via a fiber-optic-based or copper-based medium.

Handling delay (sometimes called serialization delay) is caused by the devices that handle voice information. Handling delays significantly degrade voice quality in a packet network.

Delays caused by codecs are considered handling delays. Table B-2 shows the delay introduced by different codecs.

Table B-2 codec-Induced Delays

codec	Bit Rate (kbps)	Framing size (ms)	Compression Delay (ms)
G.711 PCM	64	0.125	5
G.729 CS-ACELP	8	10	15
G.729a CS-ACELP	8	10	15

Another handling delay is the time it takes to generate a voice packet. In VoIP, the DSP generates a frame every 10 milliseconds. Two of these frames are then placed within one voice packet, so the packet delay is 20 milliseconds.

Another source of handling delay is the time it takes to move the packet to the output queue. Cisco IOS software expedites the process of determining packet destination and getting the packet to the output queue. The actual delay at the output queue is another source of handling delay and should be kept under 10 milliseconds whenever possible by using queuing methods that are optimal for your network.

In Voice over Frame Relay, you need to make sure that voice traffic is not crowded out by data traffic.

Jitter

Jitter is another factor that affects delay. Jitter is the variation between the time a voice packet is expected to be received and when it actually is received, causing discontinuity in the real-time voice stream. Voice devices such as the Cisco 3600, Cisco MC3810, and the Catalyst 4224 compensate for jitter by setting up a playout buffer to play back voice in a smooth fashion.

Playout control is handled through RTP encapsulation, either by selecting adaptive or non-adaptive playout-delay mode. In either mode, the default value for nominal delay is sufficient.

End-to-End Delay

Figuring out the end-to-end delay is not difficult if you know the end-to-end signal paths/data paths, the codec, and the payload size of the packets. Adding the delays from the endpoints to the codecs at both ends, the encoder delay (which is 5 milliseconds for the G.711 and G.726 codecs and 10 milliseconds for the G.729 codec), the packet delay, and the fixed portion of the network delay yields the end-to-end delay for the connection.

Echo

Echo is hearing your own voice in the telephone receiver while you are talking. When timed properly, echo is reassuring to the speaker. But if the echo exceeds approximately 25 milliseconds, it can be distracting and cause breaks in the conversation.

In a traditional telephony network, echo is normally caused by a mismatch in impedance from the four-wire network switch conversion to the two-wire local loop and is controlled by echo cancellers. In voice-packet-based networks, echo cancellers are built into the low bit-rate codecs and are operated on each DSP. Echo cancellers are, by design, limited by the total amount of time they will wait for the reflected speech to be received. This amount of time is called an *echo trail*. The echo trail is normally 32 milliseconds. VoIP has configurable echo trails of 8, 16, 24, and 32 milliseconds.

Signaling

Although there are various types of signaling used in telecommunications today, this document describes only those with direct applicability to Cisco voice implementations. One signaling type involves access signaling, which determines

when a line has gone off-hook or on-hook. Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) are types of access signaling. There are two common methods of providing this basic signal:

- Loop start is the most common technique for access signaling in a standard PSTN end-loop network. When a handset is picked up (goes off-hook), this action closes the circuit that draws current from the telephone company's central office (CO), indicating a change in status. This change in status signals the CO to provide a dial tone. An incoming call is signalled from the CO to the handset by a standard on/off pattern signal, causing the telephone to ring.
- Ground start is another access signaling method used to indicate on-hook/off-hook status to the CO, but this signaling method is primarily used on trunk lines or tie-lines between PBXs. Ground-start signaling works through ground and current detectors, allowing the network to indicate off-hook or seizure of an incoming call independent of the ringing signal.

Another signaling technique used mainly between PBXs or other network-to-network telephony switches is known as Ear and Mouth (E&M). There are five types of E&M signaling, as well as two different wiring methods.

VoIP Configuration Examples

This section uses four different scenarios to demonstrate how to configure Voice over IP (VoIP). The actual VoIP configuration procedure depends on the topology of your voice network. The following configuration examples should give you a starting point, but you will need to customize them to reflect your network topology.

Configuration procedures are supplied for the following scenarios:

FXS-to-FXS Connection Using RSVP

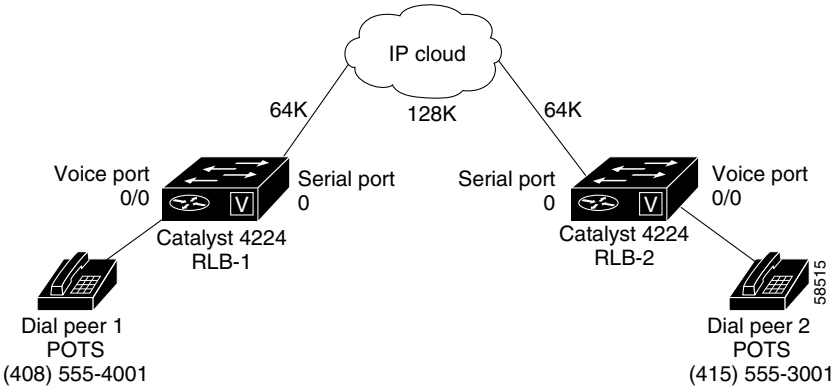
The following example shows how to configure VoIP for a simple FXS-to-FXS connection.

In this scenario, a very small company with two offices decides to integrate VoIP into its existing IP network. One basic telephony device is connected to Catalyst 4224 RLB-1; therefore, Catalyst 4224 RLB-1 is configured for one POTS dial peer and one VoIP dial peer. Catalyst 4224 RLB-w and Catalyst 4224 RLB-e establish the WAN connection between the two offices. Because one POTS telephony device is connected to Catalyst 4224 RLB-2, it is also configured for one POTS dial peer and one VoIP dial peer.

In this scenario, only the calling end (Catalyst 4224 RLB-1) is requesting RSVP.

Figure C-1 illustrates the topology of this FXS-to-FXS connection example.

Figure C-1 FXS-to-FXS Connection (Example)



Configuration for Catalyst 4224 RLB-1

```
hostname RLB-1

! Create voip dial-peer 2
dial-peer voice 2 voip

! Define its associated telephone number and IP address
 destination-pattern 14155553001
 sess-target ipv4:40.0.0.1

! Request RSVP
 req-qos controlled-load

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

! Define its associated telephone number and voice port
 destination-pattern 14085554001
 port 0/0

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

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

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

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

Configuration for Catalyst 4224 RLB-w

```
hostname RLB-w

! Configure serial interface 0
interface Serial0
 ip address 10.0.0.2 255.0.0.0

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

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

! Configure serial interface 1
interface Serial1
 ip address 20.0.0.1 255.0.0.0

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

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

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

Configuration for Catalyst 4224 RLB-e

```
hostname RLB-e

! Configure serial interface 0
interface Serial0
 ip address 40.0.0.2 255.0.0.0

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

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

! Configure serial interface 1
interface Serial1
 ip address 20.0.0.2 255.0.0.0

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

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

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

Configuration for Catalyst 4224 RLB-2

```
hostname RLB-2

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

! Define its associated telephone number and voice-port
 destination-pattern 14155553001
 port 0/0

! Create voip dial-peer 1
dial-peer voice 1 voip

!Define its associated telephone number and IP address
 destination-pattern 14085554001
 sess-target ipv4:10.0.0.1

! Configure serial interface 0
interface Serial0
 ip address 40.0.0.1 255.0.0.0
 no ip mroute-cache

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

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

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

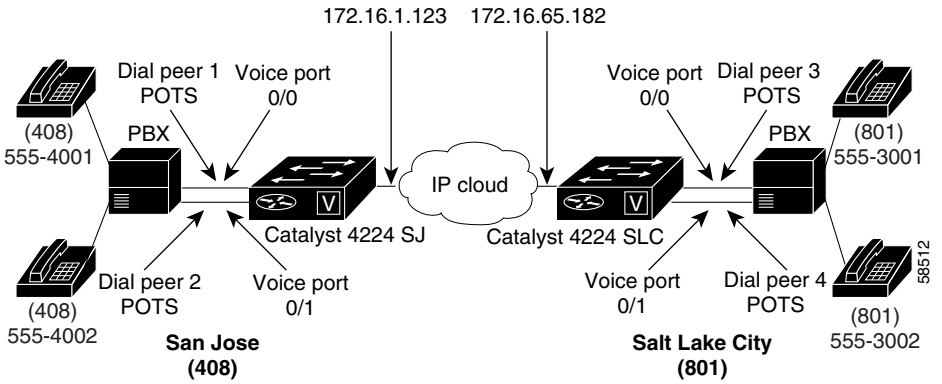
Linking PBX Users with E&M Trunk Lines

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

In this scenario, a company decides to connect two offices: one in San Jose, California, and the other in Salt Lake City, Utah. Each office has an internal telephone network using a PBX connected to the voice network by an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and Immediate Start signaling. Each E&M interface connects to the Catalyst 4224 using two voice interface connections. Users in San Jose dial *801-555* and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial *408-555* and then the extension number to reach a destination in San Jose.

Figure C-2 illustrates the topology of this scenario.

Figure C-2 Linking PBX Users with E&M Trunk Lines (Example)



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

Router San Jose Configuration

```
hostname router SJ

!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1408555....
 port 0/0

!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern 1408555....
 port 0/1

!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern 1801555....
 session target ipv4:172.16.65.182
 ip precedence 5

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

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

!Configure the serial interface 0
interface serial0
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

!Configure pots dial-peer 3
dial-peer voice 3 pots
 destination-pattern 1801555....
 port 0/0

!Configure pots dial-peer 4
dial-peer voice 4 pots
 destination-pattern 1801555....
 port 0/1

!Configure voip dial-peer 1
dial-peer voice 1 voip
 destination-pattern 1408555....
 session target ipv4:172.16.1.123
 ip precedence 5

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

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

!Configure the serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```

Note PBXs should be configured to pass all dual tone multifrequency (DTMF) signals to the router. Cisco recommends that you do not configure, store, or forward tone.

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

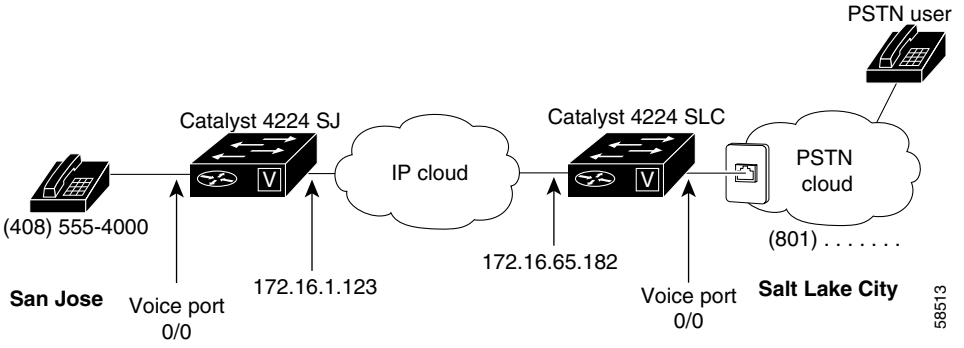
FXO Gateway to PSTN

Foreign Exchange Office (FXO) interfaces provide a gateway from the VoIP network to the analog public switched telephone network (PSTN) or to a PBX that does not support Ear and Mouth (E&M) signaling.

In this scenario, users connected to Catalyst 4224 SJ in San Jose, California, can reach PSTN users in Salt Lake City, Utah, via Catalyst 4224 SLC. Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure C-3 illustrates the topology of this scenario.

Figure C-3 FXO Gateway to PSTN (Example)



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

Router San Jose Configuration

```
hostname router SJ

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

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 1801.......
 session target ipv4:172.16.65.182
 ip precedence 5

! Configure serial interface 0
interface serial0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1801.......
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 14085554000
 session target ipv4:172.16.1.123
 ip precedence 5

! Configure serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```

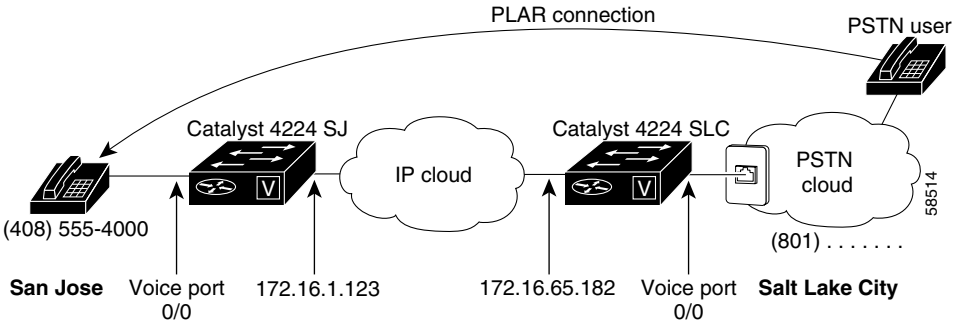
FXO Gateway to PSTN (PLAR Mode)

The following scenario shows an FXO gateway to PSTN connection in PLAR mode.

In this scenario, PSTN users in Salt Lake City, Utah, can dial a local number and establish a private line connection in a remote location. As in the previous scenario, Catalyst 4224 SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure C-4 illustrates the topology of this scenario.

Figure C-4 FXO Gateway to PSTN (PLAR Mode) (Example)



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

Router San Jose Configuration

```
hostname router SJ

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

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 1801.......
 session target ipv4:172.16.65.182
 ip precedence 5

! Configure the serial interface 0
interface serial0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1801.......
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 14085554000
 session target ipv4:172.16.1.123
 ip precedence 5

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

! Configure the serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```

A

B

C

D

E

F

G

H

I

J

L

M

N

P

Q

R

S

T

U

V

W

X