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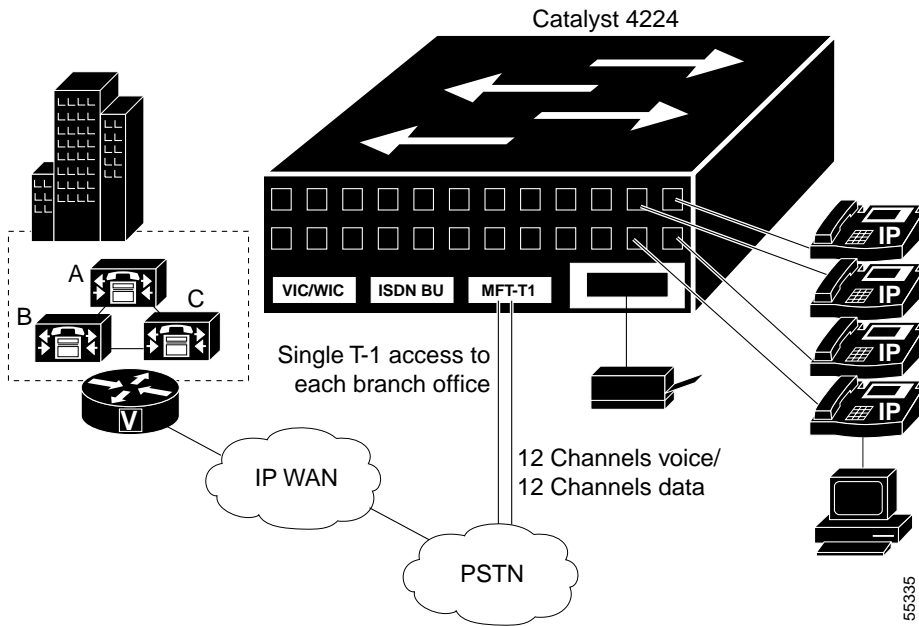
In the diagram, a Cisco CallManager cluster at a central site uses Simple Client Control Protocol (SCCP) to control IP phones at two branch offices. In the branch VoIP network, a Catalyst 4224 acts as an H.323 gateway, interconnecting the analog devices, the PSTN, and IP WAN. This system uses ISDN Basic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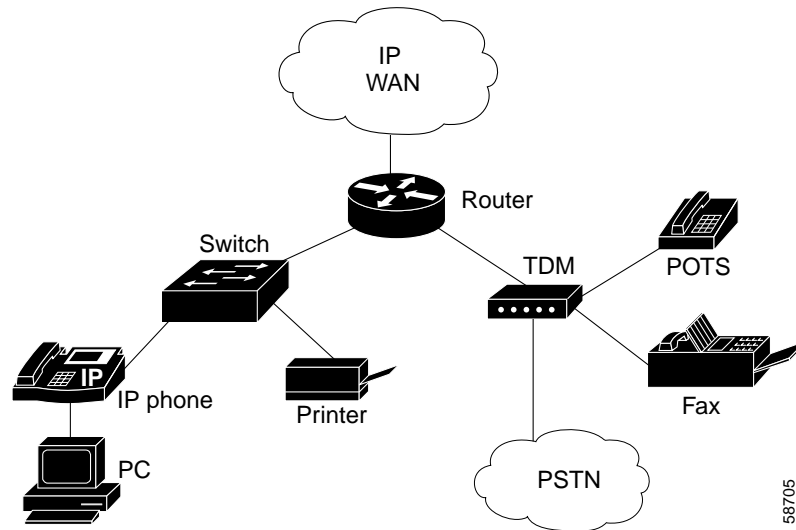
DSP Allocation

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.

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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
```

Chapter 1   Product Overview

Recommended Configurations

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:

- No VTP or DTP Support, page 1-22

- Creating a VLAN, page 1-22

- Defining a VLAN on a Trunk Port, page 1-23

- Trunking, page 1-23

- Fractional PRI Configuration, page 1-23

- No Ring Back Tone Generated, page 1-25

- MTP Required on Cisco CallManager, page 1-26

- H323-Gateway VOIP Bind SRCADDR Command, page 1-27

- Port Fast Not Enabled on Trunk Ports, page 1-28

- Priority Queuing on Frame Relay, page 1-28

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- 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