CHAPTER 36

Configuring a Voice-over-IP Network

This chapter describes how to configure a Voice-over-IP (VoIP) network. While this chapter introduces a number of Cisco networking products related to VoIP, the primary focus of the chapter is to provide configuration information for integrating Catalyst 6000 family products into your VoIP network.

This chapter consists of these sections:

Software and Hardware Requirements

The software and hardware requirements for the Catalyst switches and Cisco CallManager are as follows:

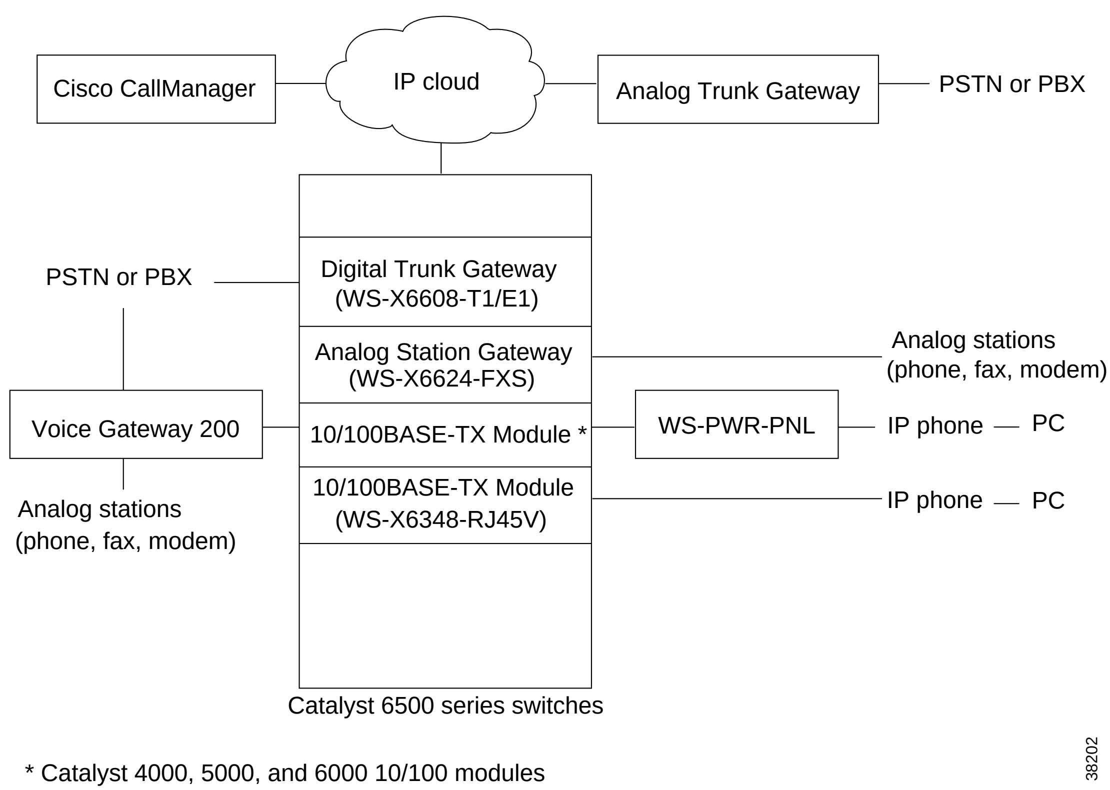
- Catalyst 4000, 5000, and 6000 family switches running supervisor engine software release 6.1(1) or later
- Cisco CallManager release 3.0 or later

Understanding a Voice-Over-IP Network

A telephony system built on an IP network instead of the traditional circuit-switched Private Branch Exchange (PBX) is called an IP PBX system. Figure 36-1 shows an IP PBX system; the individual components of this system are described in these sections:

- Cisco IP Phone 7960, page 36-3
- Cisco CallManager, page 36-4
- Access Gateways, page 36-5
  - Analog Station Gateway, page 36-5
  - Analog Trunk Gateway, page 36-6
  - Digital Trunk Gateway, page 36-6
  - Converged Voice Gateway, page 36-7
- How a Call Is Made, page 36-7

Figure 36-1 IP PBX System



Cisco IP Phone 7960

The Cisco IP Phone 7960 provides connectivity to the IP PBX system. The IP phone has two RJ-45 jacks for connecting to external devices, a LAN-to-phone jack and a PC-to-phone jack. The jacks use either Category 3 or Category 5 unshielded twisted-pair (UTP) cable. The LAN-to-phone jack is used to connect the phone to the LAN using a crossover cable; a workstation or a PC can be connected to the PC-to-phone jack using a straight-through cable.

The IP phone is Dynamic Host Configuration Protocol (DHCP) capable, providing mobility with a plug-and-play capability, and optionally, the IP phone can be programmed with a static IP address.

The IP phone can be powered by the following sources:

- External power source—Optional transformer and power cord for connecting to a standard wall receptacle.
- WS-X6348-RJ45V 10/100 switching module—Provides inline power to the IP phone.
- WS-PWR-PNL—Inline-power patch panel provides inline power to the IP phone that allows the IP phone to be connected to existing Catalyst 4000, 5000, and 6000 family 10/100BaseTX switching modules.

Examples 1 through 4 in Figure 36-2 show the various ways to connect the Cisco IP Phone 7960 and PCs to the Catalyst switch.

Figure 36-2 Connecting the Cisco IP Phone 7960 to the Catalyst Switch

Example 1—Single Cisco IP Phone 7960

Example 1 shows one IP phone connected to the 10/100 port on the Catalyst switch. The PC-to-phone jack on the phone is not used. The phone can be powered through either the 10/100 port or wall-powered.

Example 2—Single PC

Example 2 shows one PC connected to the 10/100 port on the Catalyst switch. The PC is wall-powered.

Example 3—One Cisco IP Phone 7960 and One PC

Example 3 shows one IP phone connected to the 10/100 port on the Catalyst switch and one PC connected to the PC-to-phone jack on the phone. The PC behaves as if it is connected directly to the 10/100 port on the Catalyst switch. The phone can be powered through the 10/100 port or wall-powered. The PC must be wall-powered.

Example 4—Two Cisco IP Phone 7960s and One PC

Example 4 shows two IP phones connected to the 10/100 port on the Catalyst switch and one PC connected to the PC-to-phone jack on the phone. The PC behaves as if it is connected directly to the 10/100 port on the Catalyst switch. The first phone can be powered through the 10/100 port or wall-powered. The second phone and the PC must be wall-powered.

Note For information on configuring Cisco IP phones and third-party vendor phones, refer to the documentation that shipped with the phone.

Cisco CallManager

Cisco CallManager is an open and industry-standard call processing system—the nerve center of the IP PBX system. Cisco CallManager software runs on a Windows NT server and sets up and tears down calls between phones, integrating traditional PBX functionality with the corporate IP network. Cisco CallManager manages the components of the IP PBX system, the phones, access gateways, and the resources necessary for such features as call conferencing and media mixing. Each Cisco CallManager manages the devices within its *zone* and exchanges information with the Cisco CallManager in charge of another zone to make calls possible across multiple zones. Additionally, Cisco CallManager can work with existing PBX systems to route a call over the Public Switched Telephone Network (PSTN).

Note For information on configuring Cisco CallManager to work with the IP devices described in this chapter, refer to the *Cisco CallManager Administration Guide, Release 3.0*, the *Configuration Notes for Cisco CallManager Release 3.0,* and the *Cisco CallManager v3.0 Remote Serviceability Users Guide* publications.

Access Gateways

Access gateways allow the IP PBX system to talk to existing PSTN or PBX systems. Access gateways consist of analog station gateways, analog trunk gateways, digital trunk gateways, and a *converged* voice gateway.

Analog Station Gateway

The Catalyst 6000 family 24-port Foreign Exchange Station (FXS) analog interface module allows plain old telephone service (POTS) phones and fax machines to connect to the IP PBX network. The analog station gateway behaves like the PSTN side for the POTS equipment. It requires an IP address, is registered with Cisco CallManager in its domain, and is managed by Cisco CallManager.

The 24-port FXS analog interface module features are listed in Table 36-1.

To configure the analog station interfaces, see the "Catalyst Switch Configuration Procedures" section on page 36-9. To configure the interfaces to work with Cisco CallManager, refer to the *Cisco CallManager Administration Guide, Release 3.0* publication.

Table 36-1 24-Port FXS Analog Interface Module Features

Digital Signal Processing Per Port
G.711 and G.729 voice encoding
Silence suppression, voice activity detection
Comfort noise generation
Ringer, software programmable frequency and cadence, based on country
DTMF[1] detection
Signaling, loop start
Line echo cancellation (32 ms)
Impedance (600 ohms)
Programmable analog gain, signaling timers
Fax passthrough
SPAN[2] or port mirroring support
FXS Interface Features
Address signaling formats: In-band DTMF
Signaling formats: Loop start
Ringing tone: Programmable
Ringing voltage: Programmable, based on country
Ringing frequency: Programmable, based on country
Distance: 500-ohms maximum loop

1. DTMF = dual tone multifrequency.
2. SPAN = Switched Port Analyzer.

Analog Trunk Gateway

Cisco access analog trunk gateways allow the IP PBX to connect to the PSTN or PBX. The gateway supports up to eight trunks to the PSTN and appears like a phone to the trunk lines coming from the PSTN. Using this gateway, the IP PBX places an IP call through the PSTN. Similar to the analog station gateway, the analog trunk gateway provides line echo cancellation, dual tone multifrequency (DTMF) tone generation and detection. The analog trunk gateway does not provide ring voltage as it is not connected to POTS end devices such as POTS-phones or fax machines. The analog trunk gateway requires an IP address, is registered with Cisco CallManager in its domain, and is managed by Cisco CallManager.

To configure the analog trunk gateways, refer to the documentation that shipped with the gateway. To configure the interfaces to work with Cisco CallManager, refer to the *Cisco CallManager Administration Guide, Release 3.0* publication.

Digital Trunk Gateway

The Catalyst 6000 family 8-port T1/E1 PSTN interface module is a high-density, eight port, T1/E1 VoIP module that can support both digital T1/E1 connectivity to the PSTN or transcoding and conferencing. It requires an IP address, is registered with Cisco CallManager in its domain, and is managed by Cisco CallManager.

The module software is downloaded from a TFTP server. Depending upon which software you download, the ports can serve as T1/E1 interfaces or the ports will support transcoding and conferencing. Transcoding and conferencing functions are mutually exclusive. For every transcoding port in use, there will be one less conferencing port available and vice versa.

To configure the 8-port T1/E1 PSTN interfaces, see the "Catalyst Switch Configuration Procedures" section on page 36-9. To configure the interfaces to work with Cisco CallManager, refer to the *Cisco CallManager Administration Guide, Release 3.0* publication.

The 8-port T1/E1 PSTN interface module features are listed in Table 36-2.

Table 36-2 8-Port T1/E1 PSTN Interface Module Features

Digital Signal Processing Per T1/E1 Port
G.711 to G.723 and G.729A transcoding (maximum of 8 x 32 channels of transcoding)
Conference bridging, meet-me and ad-hoc conference modes (maximum of 8 x 16 channels of conferencing)
Comfort noise generation
Fax passthrough
Silence suppression, voice activity detection
Line echo cancellation
Common Channel Signaling (CCS) For T1: 23 DS0 channels for voice traffic; 24th channel is used for signaling For E1: 29 DS0 channels for voice traffic; 16th channel is reserved for signaling Any channel can be configured for CCS.
ISDN Primary Rate Interface (PRI) signaling: Each interface supports 23 channels for T1 and 30 channels for E1. The default mode is for the 24th T1 channel or 16th E1 channel to be reserved for signaling. Both network side and user side operation modes are supported.

Table 36-2 8-Port T1/E1 PSTN Interface Module Features (continued)

Digital Signal Processing Per T1/E1 Port
T1 binary 8-zero substitution/alternate mark inversion (B8ZS/AMI) line coding, u-law or a-law coding. E1 HDB3 line coding.
T1 line bit rate: 1.544 Mbps E1 line bit rate: 2.048 Mbps
T1 line code: AMI, B8ZS E1 line code: HDB3
Framing format: D4 superframe and extended superframe
Link Management
Facilities Data Link (FDL) is a link management protocol used to help diagnose problems and gather statistics on T1 lines.

Converged Voice Gateway

The Cisco Voice Gateway 200 (VG200) allows you to connect standard POTS phones (connected directly to the gateway or anywhere on the PSTN) with Cisco IP or any H.323-compliant telephony devices. When used with Cisco CallManager, the VG200 functions as a Media Gateway Control Protocol (MGCP) gateway. The Cisco VG200 Gateway provides a 10/100BaseT Ethernet port for connection to the data network. The following telephony connections are also available:

- One to four Foreign Exchange Office (FXO) ports for connecting to a central office or PBX
- One to four FXS ports for connecting to POTS telephony devices
- One or two T1 digital ports:
  - For connecting to the PSTN using FXO emulation
  - For connecting to a T1 channel bank using FXS emulation
  - For connecting to a PBX through a trunk (tie) line using Ear and Mouth (E&M) emulation

These ports can be used to integrate a VoIP network with POTS devices, PBXs, or the PSTN.

To configure the Cisco VG200 Gateway, refer to the documentation that shipped with the gateway. To configure the interfaces to work with Cisco CallManager, refer to the *Cisco CallManager Administration Guide, Release 3.0* publication.

How a Call Is Made

An IP phone is connected to a LAN either through a hub port or a switch port. The IP phone boots up and uses DHCP to get its IP address and the IP address of its TFTP file server. The IP phone uses its IP address to talk to the TFTP server and gets its configuration file. The configuration file has information including the IP address of the phone's Cisco CallManager(s). The phone then talks with Cisco CallManager and registers itself. Each time a phone boots up, it might get a different IP address. Cisco CallManager knows how to associate a consistent user phone number to a particular phone by using the MAC address of the phone. Cisco CallManager always maintains a table mapping the "phone MAC address" and "phone number." Each time a phone registers, the table is updated with the new IP address. During registration, Cisco CallManager downloads the key pad template and the feature capability for the phone. It also tells the phone which run-time image it should use. The phone then goes to the TFTP server to get its run-time image. Each phone has a dedicated TCP connection to Cisco CallManager called the “control channel.” All control information, such as key pressing, goes from the phone to Cisco CallManager through this channel. Instructions to generate ring tone, busy tone, and so on comes from Cisco CallManager to the phone through this channel.

Cisco CallManager stores the IP-address-to-phone-number mapping (and vice versa) in its tables. When a user wants to call another user, the user keys in the called party’s phone number.
Cisco CallManager translates the phone number to an IP address and generates an IP packet version of ring tone to the called IP phone through the TCP connection. When the called IP phone receives the packet, it generates a ring tone. When the user picks up the phone, Cisco CallManager instructs the called IP phone to start talking with the calling party and removes itself from the loop. From this point on, the call goes between the two IP phones through the Real-Time Transport Protocol (RTP) which runs over User Datagram Protocol (UDP). Because voice packets are sensitive to delays, TCP is not suitable for voice transmission as timeouts and retries increase the delay between packets. When any change occurs during the call due to a feature being pressed on one of the phones, or one of the users hanging up or pressing the flash button, the information goes to Cisco CallManager through the control channel.

If a call is made to a number outside of the IP PBX network, Cisco CallManager routes the call to an analog or digital trunk gateway which in turn routes it to the PSTN.

VLAN Overview

This section describes native VLANs and auxiliary VLANs. This section uses the following terminology:

- Auxiliary VLAN—Separate VLAN for IP phones
- Native VLAN—Traditional VLAN for data
- Auxiliary VLAN ID— VLAN ID of an auxiliary VLAN
- Native VLAN ID—VLAN ID of a native VLAN

Figure 36-3 shows how a Cisco IP Phone 7960 can be connected to a Catalyst switch.

Figure 36-3 Switch-to-Phone Connections

When the IP phone is connected to a 10/100 port on the Catalyst switch, the *access port* (PC-to-phone jack) of the IP phone can be used to connect a PC.

Packets to and from the PC and to and from the phone share the same physical link to the switch and the same port of the switch. Various configurations of connecting the phone and the PC are possible (see the "Cisco IP Phone 7960" section on page 36-3).

Introducing IP-based phones into existing switch-based networks raises the following issues:

- The current VLANs might be configured on an IP subnet basis, and additional IP addresses might not be available to assign the phone to a port so that it belongs to the same subnet as other devices (PC) connected to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.

These issues can be solved by isolating the voice traffic onto a separate VLAN on each of the ports connected to a phone. The switch port configured for connecting a phone would have separate VLANs configured for carrying:

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

Isolating the phones on a separate, auxiliary VLAN increases the quality of the voice traffic and allows a large number of phones to be added to an existing network where there are not enough IP addresses. A new VLAN means a new subnet and a new set of IP addresses.

Catalyst Switch Configuration Procedures

This section describes the command-line interface (CLI) commands used to configure the Catalyst switch for VoIP operation. This section provides the following descriptions and configuration procedures:

- Voice-Related CLI Commands, page 36-10
- Configuring Per-Port Power Management, page 36-10
- Configuring Auxiliary VLANs on Catalyst LAN Switches, page 36-19
- Configuring the Access Gateways, page 36-20
- Displaying Active Call Information, page 36-27
- Configuring QoS in the Cisco IP Phone 7960, page 36-29

Note CDP must be enabled on the Catalyst switch port connected to the IP phone in order to communicate information such as auxiliary VLAN ID, per-port power management details, and QoS configuration information.

Voice-Related CLI Commands

Table 36-3 lists the CLI commands described in the configuration procedures.

Table 36-3 Voice-Related CLI Command Module and Platform Support

CLI Commands	WS-X6348-RJ45V[1]	WS-X6608-T1/E1[2]	WS-X6624-FXS[3]
Inline-power related commands			
set port inlinepower	X[4]		
set inlinepower defaultallocation	X		
show port inlinepower	X		
show environment power	X	X	X
Voice-related commands			
set port auxiliaryvlan	X/X		
show port auxiliaryvlan	X/X		
set port voice interface		X	X
show port voice interface		X	X
show port voice	X	X	X
show port voice fdl		X	
show port voice active	X	X	X
QoS commands related to voice			
set port qos *mod/port* **cos-ext** **set port qos** *mod/port* **trust-ext**	X/X		
show port qos	X/X		

1. WS-X6348-RJ45V = 48-port 10/100BaseTX switching module with voice daughter card.
2. WS-X6608-T1 and WS-X6608-E1 = 8-port T1/E1 ISDN PRI modules.
3. WS-X6624-FXS = 24-port FXS analog station interface module.
4. X = Command supported on Catalyst 6000 family switch only; XX = Command supported on Catalyst 4000, 5000, and 6000 family switches (note that all modules listed in Table 36-3 are supported only on Catalyst 6000 family switches).

Configuring Per-Port Power Management

This section describes per-port power management and the CLI commands used to configure power management for IP phones.

Note To determine the exact power requirements for your configuration to ensure that you are within the system power budget, see the "Determining System Power Requirements" section on page 14-13.

Note This section applies to the WS-X6348-RJ45V 10/100BaseTX Ethernet switching module only. For information on powering IP phones connected to other Catalyst 10/100BaseTX switching modules, refer to the *Catalyst Family Inline-Power Patch Panel Installation Note* publication.

For each IP phone connected to the WS-X6348-RJ45V module, the supervisor engine software allocates part of the available system power to power up and run the phone. The power can be applied on an individual port basis.

Only one IP phone can be powered per port; the phone must be connected directly to the switch port. If a second phone is daisy chained off the phone connected to the switch port, the second phone cannot be powered by the switch.

This section describes the following:

- Using show Commands to Display Module Type and Version Information, page 36-11
- Power Management Modes, page 36-12
- Phone Detection Summary, page 36-14
- Error Detection and Handling, page 36-15
- set port inlinepower, page 36-16
- set inlinepower defaultallocation, page 36-17
- show port inlinepower, page 36-17
- show environment power, page 36-18

Using show Commands to Display Module Type and Version Information

There are three versions of the Catalyst 6000 family 48-port 10/100BaseTX Ethernet switching module; each version has a unique product number:

- WS-X6248-RJ-45—standard 10/100BaseTX switching module
- WS-X6348-RJ-45—enhanced 10/100BaseTX switching module (enhanced QoS features and 128k per port packet buffers), accepts field-upgradable voice daughter card
- WS-X6348-RJ45V—enhanced 10/100BaseTX switching module with voice daughter card

When you enter the **show module** command, the WS-X6348 modules both display as WS-X6348-RJ-45 in the “Model” field. To determine if the module has a voice daughter card installed, look at the “Sub” field. For example, in the following display the 10/100BaseTX module in slot 8 does not have a voice daughter card, while the module in slot 9 does have a voice daughter card.

Note that further down in the **show module** display there is a submodule field that provides information about submodules. The EARL daughter card is treated as a submodule while the MSFC internal router is not treated as a submodule. The model number for the voice daughter card, as shown in the display, is WS-F6K-VPWR.

```
Console> (enable) show module
Mod Slot Ports Module-Type               Model               Sub Status
--- ---- ----- ------------------------- ------------------- -----------
1   1    2     1000BaseX Supervisor      WS-X6K-SUP1A-2GE    yes ok
15  1    1     Multilayer Switch Feature WS-F6K-MSFC         no  ok
8   8    48    10/100BaseTX Ethernet     WS-X6348-RJ-45      no  ok
9   9    48    10/100BaseTX Ethernet     WS-X6348-RJ-45      yes ok

Mod Module-Name         Serial-Num
--- ------------------- -----------
1                       SAD03436055
15                      SAD03432597
9                       SAD03414268

Mod MAC-Address(es)                        Hw     Fw         Sw
--- -------------------------------------- ------ ------------------------
1   00-30-80-f7-a5-06 to 00-30-80-f7-a5-07 1.0    5.2(1)  6.2(0.32-Eng)FTL
    00-30-80-f7-a5-04 to 00-30-80-f7-a5-05
    00-30-a3-4a-a0-00 to 00-30-a3-4a-a3-ff
15  00-d0-bc-ee-d0-dc to 00-d0-bc-ee-d1-1b 1.2    12.0(3)XE1 12.0(3)XE1
8   00-d0-c0-c8-83-ac to 00-d0-c0-c8-83-db 1.1    4.2(0.24)V6.1(0.37)FTL
9   00-50-3e-7c-43-00 to 00-50-3e-7c-43-2f 0.201  5.3(1)

Mod Sub-Type                Sub-Model           Sub-Serial  Sub-Hw
--- ----------------------- ------------------- ----------- ------
1   L3 Switching Engine     WS-F6K-PFC          SAD03451187 1.0
9   Inline Power Module     WS-F6K-VPWR                     1.0
Console> (enable)
```

Use the **show version** command to show the version of modules and submodules:

```
Console> (enable) show version 9
Mod Port Model              Serial #    Versions
--- ---- ------------------- -------------------------------------------------
9   48   WS-X6348           SAD03414268 Hw :0.201
Fw :5.3(1)
Sw :6.1(0.32)FTL
WS-F6K-VPWR                             Hw :1.0
Console> (enable)
```

Power Management Modes

Each port is configured through the CLI, SNMP, or a configuration file to be in one of the following modes (configured through **set port inlinepower** CLI command):

- **Auto**—The supervisor engine directs the switching module to power up the port *only* if the switching module has discovered the phone.
- **Off**—The supervisor engine does not direct the switching module to power up the port even if an unpowered phone is connected.

Each port also has a status, defined as:

- On—Power is being supplied by the port.
- Off—Power is not being supplied by the port.
- Power-deny—The supervisor engine does not have enough power to allocate to the port; power is not being supplied by the port.
- Faulty—The port is unable to provide power to the connected device.

Unpowered Phone

When an unpowered phone is discovered on a switching module port, the switching module reports to the supervisor engine that an unpowered phone is present and on which module/port. If the port is configured in **Auto** mode, the supervisor engine determines if there is enough available system power to allow the switching module to power up and run the phone. If there is sufficient power, the supervisor engine removes the *default allocated power* required by a phone from the total available system power and then sends a message to the switching module instructing it to provide power to the port. If there is not enough available power for the phone, the supervisor engine sends a message to the switching module indicating that power is denied to the port.

After power is applied to the port, the supervisor engine monitors the port to ensure that the link comes up. If the link does not come up within 4 seconds, the supervisor engine instructs the switching module to turn power off. The entire cycle is then repeated, and the switching module performs discovery and reports to the supervisor engine if a device is present on the port.

Power Requirements

IP Phones may have different power requirements. The supervisor engine initially allocates the configured default of 7W (167 mA at 42V) to the Cisco IP Phone. When the correct amount of power is determined from the CDP messaging with the Cisco IP Phone, the supervisor engine reduces or increases the allocated power.

For example, the default allocated power is 7W. A Cisco IP Phone requiring 6.3W is plugged into a port. The supervisor engine allocates 7W for the Cisco IP Phone and powers it up. Once the Cisco IP Phone is operational, it sends a CDP message with the actual power requirement to the supervisor engine. The supervisor engine then decreases the allocated power to the required amount.

Wall-Powered Phones

When a wall-powered phone is present on a switching module port, the switching module cannot detect its presence. The supervisor engine discovers the phone through CDP messaging with the port. If the phone supports inline power (the supervisor engine determines this through CDP), and the mode is set to **Auto** or **Off**, the supervisor engine does not attempt to power on the port. If there is a power outage and the mode is set to **Auto**, the phone will lose power, but the switching module will discover the phone and inform the supervisor engine, which then applies inline power to the phone.

Powering Off the Phone

The supervisor engine can turn off power to a specific port by sending a message to the switching module. That power is then added back to the available system power. This only occurs when you power off the phone through the CLI or SNMP.

Phone Removal

The switching module informs the supervisor engine if a *powered* phone is removed using a link-down message. The supervisor engine then adds the allocated power for that port back to the available system power.

The switching module also informs the supervisor engine if an *unpowered* phone is removed.

Caution When a phone cable is plugged into a port and power is turned on, the supervisor engine has a 4-second timeout waiting for the link to go up on the line. During those 4 seconds, if the phone cable is unplugged and a network device is plugged in, the device could be damaged. We recommend that you wait at least 10 seconds between unplugging a device and plugging in a new device.

High Availability Support

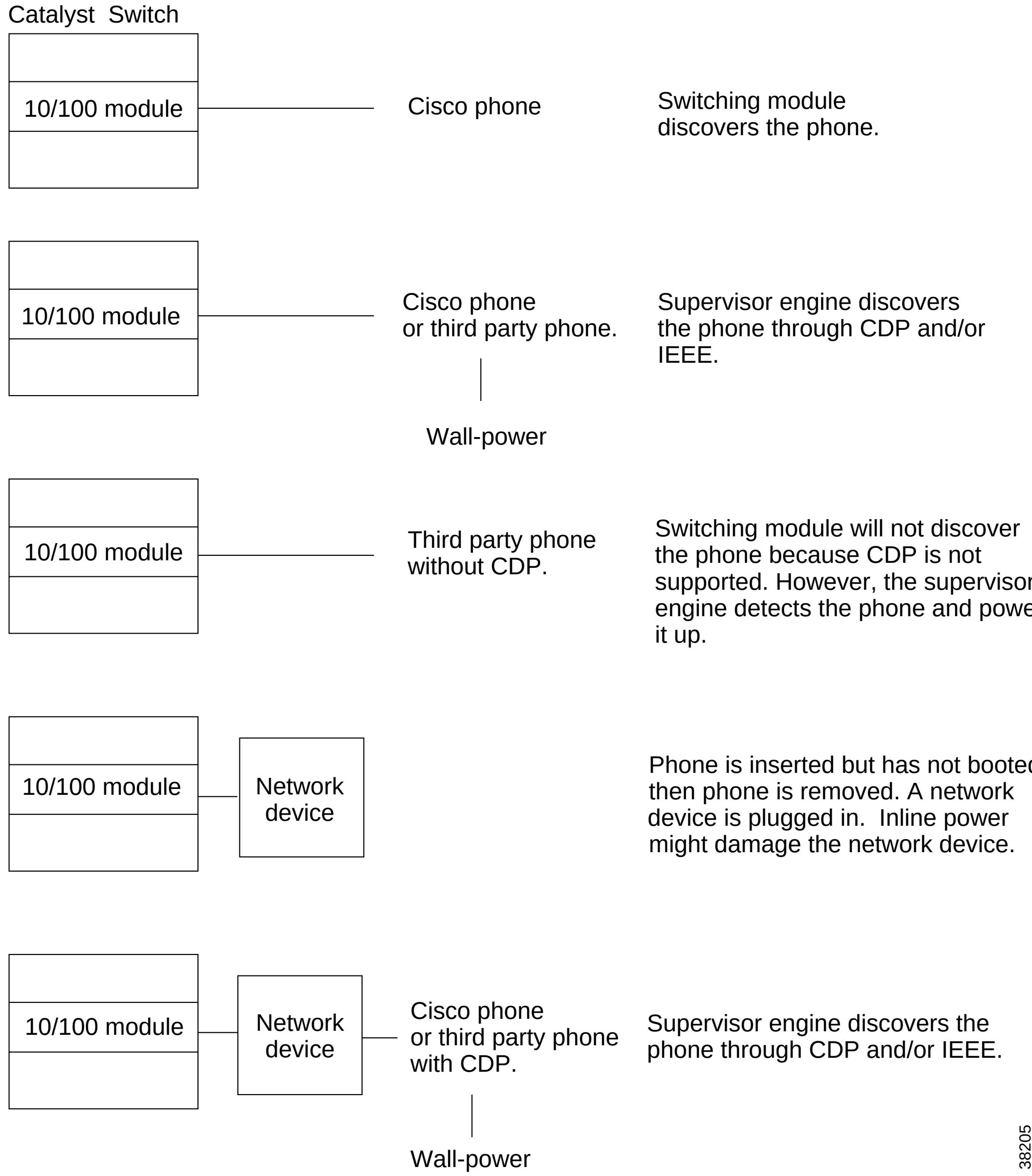
To support high availability during a failover from the active supervisor engine to the standby supervisor engine, the per-port power management and phone status information is synchronized between the active and standby supervisor engines.

The information to be synchronized (on a per-port basis) is the presence of a phone, the phone power status (on, off, denied, or faulty), and the amount of power consumed by the phone. The active supervisor engine sends this information to the standby supervisor engine and the standby supervisor engine updates its internal data structures. When a switchover occurs, the standby supervisor engine allocates power to the modules and ports from the available power, one module at a time. Once the power for each module has been allocated, the supervisor engine allocates power to the phones, beginning with the lowest slot number, until all inline powered ports have been either powered on, off, or denied.

Phone Detection Summary

Figure 36-4 shows how the system detects a phone connected to a Catalyst switch port.

Figure 36-4 Power Detection Summary



Error Detection and Handling

This section describes how the Catalyst 6000 family switch handles fault detection and errors related to per-port power management.

Device is Powered but Link is Not Up

The supervisor engine detects that the device is powered but the link is not up by setting a timeout when the switching module is directed to power up the device. If the timeout occurs and the supervisor engine has not received a “link up” for the port, the following syslog message is displayed:

```
1999 Jul 14 10:05:58 %SYS-5-PORT_DEVICENOLINK: Device on Port 4/7 powered, no link up.
```

The supervisor engine also directs the switching module to power off the port. The switching module then performs discovery again on the port.

Port is Unable to Provide Inline Power to the Device

The switching module detects when there is a problem providing inline power to the device and reports this to the supervisor engine. The following syslog message is displayed:

```
1999 Jul 14 10:05:58 %SYS-5-PORT_INLINEPWRFLTY: Port 4/7 reporting inline power as
faulty.
```

Not Enough Available Power to Power the Device

The supervisor engine keeps track of the available power left in the system and will not power up any ports if there is no available power remaining. The following syslog message is displayed:

```
1999 Jul 14 10:05:58 %SYS-5-PORT_NOPOWERAVAIL: Device on Port 4/7 will remain unpowered.
```

The supervisor engine informs the switching module that power to the port is denied.

Power Supply Configured from Nonredundant to Redundant

Depending upon the number and type of modules in the chassis, some modules might need to be powered off to prevent overdrawing power from the power supply. The supervisor engine first powers off and reallocates the power supplied by the ports and then starts the process of powering off and reallocating the power used by the module.

Power Supply Configured from Redundant to Nonredundant

Once a module that was powered down due to lack of power is powered up and comes online, the module begins discovery on the ports to determine the presence of unpowered connected devices (phones). The module reports discovered devices to the supervisor engine, which then directs the switching module to power up the device (if the port is configured to do so).

For modules that are already powered on, but have devices connected that are power denied, the supervisor engine attempts to power on the devices starting with the lowest numbered slot to the highest numbered slot, and from the lowest port number to the highest port number, one module at a time.

set port inlinepower

This example shows how to set the power mode of a port or group of ports:

```
Console> (enable) set port inlinepower help
Usage: set port inlinepower <mod/port> <auto|off>
Console> (enable) set port inlinepower 2/5 off
Inline power for port 2/5 set to off.
Console> (enable) set port inlinepower 2/3-9 auto
Inline power for ports 2/3-9 set to auto.
Console> (enable)
```

set inlinepower defaultallocation

This example shows how to set the default power allocation for a port:

```
Console> (enable) set inlinepower defaultallocation help
Usage: set inlinepower defaultallocation <value>
       (value = 2000..12500 (mWatt))
Console> (enable) set inlinepower defaultallocation 9500
Default inline power allocation set to 9500 mWatt per applicable port.
Console> (enable)
```

show port inlinepower

This example shows how to display the power status for modules and individual ports:

```
Console> (enable) show port inlinepower help
Usage: show port inlinepower [mod/port]
Console> show port inlinepower 3/2-6
Default Inline Power allocation per port: 9.500 Watts (0.22 Amps @42V)
Total inline power drawn by module 3: 0 Watt
Port      InlinePowered     PowerAllocated
      Admin Oper   Detected mWatt mA @42V
----- ----- ------ -------- ----- --------
 3/2  auto  on     yes      10.00 0.250
 3/3  auto  on     yes      9.8   0.198
 3/4  auto  denied yes      0     0
 3/5  off   off    no       0     0
 3/6  off   off    yes      0     0
Console> (enable)
```

Operational (Oper) status field descriptions:

- on—Power is being supplied by the port.
- off—Power is not being supplied by the port.
- denied—System does not have enough available power for the port.
- faulty—Port is unable to supply power.

show environment power

This example shows how to display the power status for modules and individual ports:

```
Console> (enable) show environment power help
Usage: show environment power [mod]
Console> (enable) show environment power 5
Feature not supported on module 5.
Console> (enable) show environment power 9
Module 9:
Default Inline Power allocation per port: 9.500 Watts (0.22 Amps @42V)
Total inline power drawn by module 9: 0 Watt

Slot power Requirement/Usage :

Slot Card Type           PowerRequested PowerAllocated CardStatus
                         Watts   A @42V Watts   A @42V
---- ------------------- ------- ------ ------- ------ ----------
9    WS-X6348            123.06   2.93  123.06   2.93  ok

Default Inline Power allocation per port: 9.500 Watts (0.22 Amps @42V)
Port      InlinePowered     PowerAllocated
      Admin Oper   Detected mWatt mA @42V
----- ----- ------ -------- ----- --------
 9/1  auto  off    no       0     0
 9/2  auto  off    no       0     0
 9/3  auto  off    no       0     0
 9/4  auto  off    no       0     0
 9/5  auto  off    no       0     0
 9/6  auto  off    no       0     0
 9/7  auto  off    no       0     0
 9/8  auto  off    no       0     0
.
(display text omitted)
.
 9/48 auto  off    no       0     0
Console> (enable)

Console> (enable) show environment power
PS1 Capacity: 1153.32 Watts (27.46 Amps @ 42V)
PS2 Capacity: none
PS Configuration : PS1 and PS2 in Redundant Configuration.
Total Power Available: 1153.32 Watts (27.46 Amps @ 42V)
Total Power Available for Line Card Usage: 1153.32 Watts (27.46 Amps @ 42V)
Total Power Drawn From the System: 289.80 Watts (6.90 Amps @ 42V)
Remaining Power in the System: 863.52 Watts (20.56 Amps @42V)
Default inline power allocation: 10.5 Watts/port (0.25 Amps @ 42V)

Slot power Requirement/Usage :

Slot  Card-Type            Power-Requested  Power-Allocated  Card-Status
                           Watts   A @ 42V  Watts   A @ 42V
----  -------------------  ------- -------  ------- -------  ------------
1                          0.00    0.00     126.42  3.01     none
2     WS-X6K-SUP1-2GE      138.60  3.30     138.60  3.30     ok
3     WS-X6348-RJ-45       114.24  2.72     151.20  3.60     ok
5     WS-X6348-RJ-45       109.20  2.60     100.88  2.40     partial-deny
6     Unknown              112.98  2.69     0       0        unknown
7     WS-X6248-RJ-45       84.84   2.02     0       0        power-bad
9     WS-X6416-GE-MT       105.00  2.50     0       0        power-deny
Console> (enable)
```

A partial-deny status indicates that some module ports are inline powered but not all the ports on the module are inline powered.

Configuring Auxiliary VLANs on Catalyst LAN Switches

These sections describe how to configure auxiliary VLANs:

Understanding Auxiliary VLANs

You can configure switch ports to send CDP packets that instruct an attached Cisco IP Phone 7960 to transmit voice traffic to the switch in these frame types:

- 802.1Q frames carrying the auxiliary VLAN ID and Layer 2 CoS set to 5
  - The switch port drops all 802.1Q frames except those carrying the auxiliary VLAN ID.
  - Reset the Cisco IP Phone 7960 if the auxiliary VLAN ID changes.
  - Use the **set port auxiliaryvlan** *mod*[/*port*] *aux_vlan_id* command.

Note Cisco recommends use of 802.1Q frames and a separate VLAN.

- 802.1p frames, which are 802.1Q frames carrying VLAN ID 0 and Layer 2 CoS set to 5 (use the **set port auxiliaryvlan** *mod*[/*port*] **dot1p** command)
- 802.3 frames, which are untagged and carry no VLAN ID and no Layer 2 CoS value (use the **set port auxiliaryvlan** *mod*[/*port*] **untagged** command)

Note The Cisco IP Phone 7960 always sets Layer 3 IP precedence to 5 in voice traffic.

Auxiliary VLAN Configuration Guidelines

Observe the following guidelines when configuring auxiliary VLANs:

- The IP phone and a device attached to the phone are in the same VLAN and must be in the same IP subnet:
  - If they use the same frame type
  - If the phone uses 802.1p frames and the device uses untagged frames
  - If the phone uses untagged frames and the device uses 802.1p frames
  - If the phone uses 802.1Q frames and the auxiliary VLAN equals the native VLAN
- The IP phone and a device attached to the phone cannot communicate if they are in the same VLAN and subnet but use a different frame type, because traffic between devices in the same subnet is not routed (routing would eliminate the frame type difference).
- You cannot use switch commands to configure the frame type used by traffic received from a device attached to the phone’s access port.

Configuring Auxiliary VLANs

This example shows how to add voice ports to auxiliary VLANs, specify an encapsulation type, or specify that the VLAN will not send or receive CDP messages with voice-related information:

```
Console> (enable) set port auxiliaryvlan 2/1-3 222
Auxiliaryvlan 222 configuration successful.
AuxiliaryVlan AuxVlanStatus Mod/Ports
------------- ------------- -------------------------
222           active        1/2,2/1-3
Console> (enable) set port auxiliaryvlan 5/7 untagged
Port 5/7 allows the connected device send and receive untagged packets and without 802.1p
priority.
Console> (enable) set port auxiliaryvlan 5/9 dot1p
Port 5/9 allows the connected device send and receive packets with 802.1p priority.
Console> (enable) set port auxiliaryvlan 5/12 none
Port 5/12 will not allow sending CDP packets with Voice VLAN information.
Console> (enable)
```

The default setting is **none**. Table 36-4 lists the **set port auxiliaryvlan** command keywords and their descriptions.

Table 36-4 Keyword Descriptions

Keyword	Action of the Phone
dot1p	Specify the phone send packets with 802.1p priority 5.
untagged	Specify the phone send untagged packets.
none	Specify that the switch does not send any auxiliary VLAN information in the CDP packets from that port.

Verifying Auxiliary VLAN Configuration

This example shows how to display auxiliary VLAN status:

```
Console> show port auxiliaryvlan 123
AuxiliaryVlan AuxVlanStatus Mod/Ports
------------- ------------- -------------------------
222           active        1/2,2/1-3
Console>
```

Configuring the Access Gateways

This section describes the commands used to configure the following Catalyst 6000 family access gateway modules:

- Analog station gateway—24-port FXS analog interface module
- Digital trunk gateway—8-port T1/E1 PSTN interface module

set port voice interface

If DHCP is enabled for a port, the port obtains all other configuration information from the TFTP server. When disabling DHCP on a port, some mandatory parameters must be specified as follows:

- If DNS parameters are not specified, the software uses the system DNS configuration on the supervisor engine to configure the port.
- 8-port T1/E1 PSTN interface module only: You cannot specify more than one port at a time because a unique IP address must be set for each port.

These examples show how to configure the port voice interface for DHCP, TFTP, and DNS servers:

```
Console> (enable)  set port voice interface help
Usage: set port voice interface <mod/port> dhcp enable [vlan <vlan>]
       set port voice interface <mod/port> dhcp disable <ipaddrspec>
                           tftp <ipaddr> [vlan <vlan>]
                           [gateway <ipaddr>] [dns [ipaddr] [domain_name]]
     (ipaddr_spec: <ipaddr> <mask>, or <ipaddr>/<mask>
           <mask>: dotted format (255.255.255.0) or number of bits (0..31)
             vlan: 0..1000
      System DNS may be used if disabling DHCP without DNS parameters)

Console> (enable) set port voice interface 7/1 dhcp enable
Port 7/1 DHCP enabled.

Console> (enable) set port voice interface 7/3 dhcp disable 171.68.111.41/24 tftp
173.32.43.11 dns 172.20.34.204 cisco.com
Port 7/3 dhcp disabled.
System DNS configurations applied.

Console> (enable) set port voice interface 7/4-6 dhcp enable vlan 3
Vlan 3 configuration successful
Ports 7/4-6 DHCP enabled.
Console> (enable)
```

show port voice interface

This example shows how to display the port voice interface configuration (this display is from the 24-port FXS analog interface module):

```
Console> (enable) show port voice interface help
Usage: show port voice interface <mod/port>
Console> show port voice interface 5
Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 5/1-24  disable 00-10-7b-00-13-ea 10.6.15.158     255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 5/1-24  10.6.15.155       -               10.6.15.155     -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 5/1-24  12.2.2.1*         cisco.cisco.com
         7.7.7.7
(*): Primary
Console> (enable)
```

show port voice fdl

This example shows how to display Facilities Data Link (FDL) statistics for the specified ports:

Note FDL is a link management protocol used to help diagnose problems and gather statistics.

```
Console> (enable) show port voice fdl 7/1-3
Port  ErrorEvents       ErroredSecond     SeverlyErroredSecond
      Last 15' Last 24h Last 15' Last 24h Last 15' Last 24h
----- -------- -------- -------- -------- -------- -----------
 7/1  17       18       19       20       21       22
 7/2  17       18       19       20       21       22
 7/3  17       18       19       20       21       22

Port  FailedSignalState FailedSignalSecond
      Last 15' Last 24h Last 15' Last 24h
----- -------- -------- -------- ---------
 7/1  37       38       39       40
 7/2  37       38       39       40
 7/3  37       38       39       40

Port         LES               BES               LCV
      Last 15' Last 24h Last 15' Last 24h Last 15' Last 24h
----- -------- -------- -------- -------- -------- --------
 7/1  41       48       49       50       53       54
 7/2  41       48       49       50       53       54
 7/3  41       48       49       50       53       54
Console> (enable)
```

Table 36-5 describes the possible fields (depending on the port type queried) in the **show port voice fdl** command output.

Table 36-5 FDL Field Descriptions

Field	Description
ErrorEvents	Count of errored events.
ErroredSecond	Count of errored seconds.
SeverelyErroredSecond	Count of severely errored seconds.
FailedSignalState	Count of failed signal state errors.
FailedSignalSecond	Count of errored events.
LES	Line errored seconds detected.
BES	Bursty errored seconds detected.
LCV	Line code violation seconds detected.

show port

This example shows how to display the port configuration for individual ports. This section describes the **show port** command displays for the following gateway modules:

- Digital trunk gateway—8-port T1/E1 PSTN interface module
- Analog station gateway—24-port FXS analog interface module

8-Port T1/E1 PSTN Interface Module

The Status field shows Layer 2 status of the ports. Possible values are: notconnect, connected, disabled, and faulty. The following display is for the T1 module. The display would be the same for the E1 module except the port speed for the E1 module would be 2.048.

```
Console> show port 7
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 7/1                     connected  123        full   1.544 T1
 7/2                     connected  2          full   1.544 T1
 7/3                     disable    1          full   1.544 T1
 7/4                     connected  11         full   1.544 T1
 7/5                     connected  123        full   1.544 T1
 7/6                     connected  1          full   1.544 T1
 7/7                     faulty     2          full   1.544 T1
 7/8                     faulty     2          full   1.544 T1

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 7/1     enable  00-10-7b-00-0a-58 172.20.34.68    255.255.255.0
 7/2     enable  00-10-7b-00-0a-59 172.20.34.70    255.255.255.0
 7/3     enable  00-10-7b-00-0a-5a 172.20.34.64    255.255.255.0
 7/4     enable  00-10-7b-00-0a-5b 172.20.34.66    255.255.255.0
 7/5     enable  00-10-7b-00-0a-5c 172.20.34.59    255.255.255.0
 7/6     enable  00-10-7b-00-0a-5d 172.20.34.67    255.255.255.0
 7/7     enable  00-10-7b-00-0a-5e (Port host processor not online)
 7/8     enable  00-10-7b-00-0a-5f (Port host processor not online)

Port     Call-Manager(s)   DHCP-Server     TFTP-Sever      Gateway
-------- ----------------- --------------- --------------- ---------------
 7/1     172.20.34.207*    172.20.34.207   172.20.34.207   -
         callm.cisco.com
 7/2     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20
 7/3     172.20.34.207     172.20.34.207   172.20.34.207   -
 7/4     172.20.34.207     172.20.34.207   172.20.34.207   -
 7/5     172.20.34.207     172.20.34.207   172.20.34.207   -
 7/6     172.20.34.207     172.20.34.207   172.20.34.207   -
 7/7     (Port host processor not online)
 7/8     (Port host processor not online)

Port     DNS-Server(s)   Domain
-------- --------------- -------------------------------------------------
 7/1     172.20.34.207   cisco.com
 7/2     172.20.34.207*  int.cisco.com
         171.69.45.34
         172.78.111.132
 7/3     172.20.34.207   -
 7/4     172.20.34.207   -
 7/5     172.20.34.207   -
 7/6     172.20.34.207   -
 7/7     (Port host processor not online)
 7/8     (Port host processor not online)
```

```
Port     CallManagerState DSP-Type
-------- ---------------- --------
 7/1     registered       C549
 7/2     registered       C549
 7/3     registered       C549
 7/4     registered       C549
 7/5     registered       C549
 7/6     notregistered    C549
 7/7     (Port host processor not online)
 7/8     (Port host processor not online)

Port  NoiseRegen NonLinearProcessing
----- ---------- -------------------
 7/1  disabled   disabled
 7/2  disabled   disabled
 7/3  disabled   disabled
 7/4  disabled   disabled
 7/5  enabled    disabled
 7/6  disabled   enabled
 7/7  (Port host processor not online)
 7/8  (Port host processor not online)

(*): Primary
Console>
```

8-Port T1/E1 PSTN Interface Module Configured for Trancoding/Conferencing

MTP (media termination point) and Conf Bridge (conference bridge) are types of ports. Transcoding applies to a call on an MTP port.

In this example, a transcoding port shows as “MTP” and a conference port shows as “Conf Bridge.”

```
Console> (enable) show port 7
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 7/1                     notconnect 1            full 1.544 T1
 7/2                     notconnect 1            full 1.544 T1
 7/3                     connected  1            full 1.544 T1
 7/4                     connected  1            full 1.544 T1
 7/5                     connected  1            full 1.544 T1
 7/6                     connected  1            full 1.544 T1
 7/7                     enabled    1            full     - Conf Bridge
 7/8                     enabled    1            full     - MTP

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 7/1     enable  00-10-7b-00-12-08 10.6.15.165     255.255.255.0
 7/2     enable  00-10-7b-00-12-09 10.6.15.166     255.255.255.0
 7/3     enable  00-10-7b-00-12-0a 10.6.15.167     255.255.255.0
 7/4     enable  00-10-7b-00-12-0b 10.6.15.168     255.255.255.0
 7/5     enable  00-10-7b-00-12-0c 10.6.15.169     255.255.255.0
 7/6     enable  00-10-7b-00-12-0d 10.6.15.170     255.255.255.0
 7/7     enable  00-10-7b-00-12-0e 10.6.15.171     255.255.255.0
 7/8     enable  00-10-7b-00-12-0f 10.6.15.172     255.255.255.0
```

```
Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 7/1     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/2     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/3     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/4     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/5     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/6     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/7     10.6.15.155       10.6.15.155     10.6.15.155     -
 7/8     10.6.15.155       10.6.15.155     10.6.15.155     -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 7/1     -                 -
 7/2     -                 -
 7/3     -                 -
 7/4     -                 -
 7/5     -                 -
 7/6     -                 -
 7/7     -                 -
 7/8     -                 -

Port     CallManagerState DSP-Type
-------- ---------------- --------
 7/1     registered       C549
 7/2     registered       C549
 7/3     registered       C549
 7/4     registered       C549
 7/5     registered       C549
 7/6     registered       C549
 7/7     registered       C549
 7/8     registered       C549

Port  NoiseRegen NonLinearProcessing
----- ---------- -------------------
 7/1  enabled    enabled
 7/2  enabled    enabled
 7/3  enabled    enabled
 7/4  enabled    enabled
 7/5  enabled    enabled
 7/6  enabled    enabled
 7/7  disabled   disabled
 7/8  disabled   disabled
Console> (enable)
```

24-Port FXS Analog Interface Module

In this example, all ports should have a “Type” of FXS, and all ports in the same module should belong to one VLAN.

```
Console> (enable) show port 3
Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 3/1                     onhook     1            full   64k FXS
 3/2                     onhook     1            full   64k FXS
 3/3                     onhook     1            full   64k FXS
 3/4                     onhook     1            full   64k FXS
 3/5                     onhook     1            full   64k FXS
 3/6                     onhook     1            full   64k FXS
 3/7                     onhook     1            full   64k FXS
 3/8                     offhook    1            full   64k FXS
 3/9                     offhook    1            full   64k FXS
 3/10                    onhook     1            full   64k FXS
 3/11                    onhook     1            full   64k FXS
 3/12                    onhook     1            full   64k FXS
 3/13                    onhook     1            full   64k FXS
 3/14                    onhook     1            full   64k FXS
 3/15                    onhook     1            full   64k FXS
 3/16                    onhook     1            full   64k FXS
 3/17                    onhook     1            full   64k FXS
 3/18                    onhook     1            full   64k FXS
 3/19                    onhook     1            full   64k FXS
 3/20                    onhook     1            full   64k FXS
 3/21                    onhook     1            full   64k FXS
 3/22                    onhook     1            full   64k FXS
 3/23                    onhook     1            full   64k FXS
 3/24                    onhook     1            full   64k FXS

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 3/1-24  enable  00-10-7b-00-13-e4 172.20.34.50    255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Sever      Gateway
-------- ----------------- --------------- --------------- ---------------
 3/1-24  172.20.34.207     172.20.34.207   172.20.34.207   -

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 3/1-24  172.20.34.207*    cisco.com
         172.34.23.111

Port     CallManagerState DSP-Type
-------- ---------------- --------
 3/1-24  registered       C549

Port     ToneLocal     Impedance InputGain(dB) OutputAtten(dB)
-------- ------------- --------- ------------- ---------------
 3/1-24  northamerica  0         0             0

Port     RingFreq Timing    Timing         Timing    Timing
         (Hz)     Digit(ms) InterDigit(ms) Pulse(ms) PulseDigit(ms)
-------- -------- --------- -------------- --------- --------------
 3/1-24  20       100       100            0         0
(*): Primary
Console> (enable)
```

Displaying Active Call Information

Use the **show port voice active** command to display active call information on a port. There are up to 8 calls per port for the 8-port T1/E1 PSTN interface module but only one call per port for the 24-port FXS analog station interface module.

This example shows that the **ip-addr** option displays one specific call for the specified IP address. You can also use an IP alias.

```
Console> (enable) show port voice active help
Usage: show port voice active [mod/port] [all|call|conference|transcode] [ipaddr]
Console> (enable)
```

Entering the **show port voice active** command without any parameters shows all the calls in the system (regular calls, conference calls, and transcoding calls). Display field descriptions are as follows:

- Type—The “call” notation is for 24-port FXS analog interface module and 8-port PSTN interface module calls.

  When 8-port T1/E1 PSTN interfaces are configured for transcoding and/or conferencing, the Type field displays “conferencing” for conferencing calls and “transcoding” for transcoding calls.
- Conference-ID, Transcoding-ID, and Party-ID are only applicable to 8-port T1/E1 PSTN interfaces configured for transcoding and/or conferencing.

This example shows all active calls in the system:

```
Console> show port voice active
Port  Type         Total Conference-ID/ Party-ID IP-Address
                         Transcoding-ID
----- ------------ ----- -------------- -------- ---------------
 3/1  call         1     -              -        199.22.25.254
 3/2  call         1     -              -        172.225.25.54
 4/5  call         3     -              -        165.34.234.111
                                                 172.32.34.12
                                                 198.96.23.111
 3/8  conferencing 2     1              1        255.255.255.241
                                        2        173.23.13.42
                                        3        198.97.123.98
                                        5        182.34.54.26
                         2              1        199.22.25.25
                                        3        182.34.54.2
                                        6        121.43.23.43
 3/2  call         1     -              -        172.225.25.54
 3/8  transcoding  1     1              1        255.255.255.241
                                        2        183.32.43.3
```

This example shows how to display detailed call information for a port (specifying the module only, shows detailed call information for all ports on the module):

```
Console> show port voice active 3/2
Port 3/2:
Channel #1:
  Remote IP address                          : 165.34.234.111
  Remote UDP port                            : 124
  Call state                                 : Ringing
  Codec Type                                 : G.711
  Coder Type Rate                            : 35243
  Tx duration                                : 438543 sec
  Voice Tx duration                          : 34534 sec
  ACOM Level Current                         : 123213
  ERL Level                                  : 123 dB
  Fax Transmit Duration                      : 332433
  Hi Water Playout Delay                     : 23004 ms
  Logical If index                           : 4
  Low water playout delay                    : 234 ms
  Receive delay                              : 23423 ms
  Receive bytes                              : 2342342332423
  Receive packets                            : 23423423402384
  Transmit bytes                             : 23472377
  Transmit packets                           : 94540
Channel #2:
  Remote IP address                          : 165.34.234.112
  Remote UDP port                            : 125
  Call state                                 : Ringing
  Codec Type                                 : G.711
  Coder Type Rate                            : 35243
  Tx duration                                : 438543 sec
  Voice Tx duration                          : 34534 sec
  ACOM Level Current                         : 123213
  ERL Level                                  : 123 dB
  Fax Transmit Duration                      : 332433
  Hi Water Playout Delay                     : 23004 ms
  Logical If index                           : 4
  Low water playout delay                    : 234 ms
  Receive delay                              : 23423 ms
  Receive bytes                              : 2342342332423
  Receive packets                            : 23423423402384
  Transmit bytes                             : 23472377
  Transmit packets                           : 94540
Channel #3:
.
(display text omitted)
.
Console>
```

This example shows how to display a specific call at a specified IP address:

```
Console> show port voice active 3/2 171.69.67.91
  Remote IP address                          : 171.69.67.91
  Remote UDP port                            : 125
  Call state                                 : Ringing
  Codec Type                                 : G.711
  Coder Type Rate                            : 35243
  Tx duration                                : 438543 sec
  Voice Tx duration                          : 34534 sec
  ACOM Level Current                         : 123213
  ERL Level                                  : 123 dB
  Fax Transmit Duration                      : 332433
  Hi Water Playout Delay                     : 23004 ms
  Logical If index                           : 4
  Low water playout delay                    : 234 ms
  Receive delay                              : 23423 ms
  Receive bytes                              : 2342342332423
  Receive packets                            : 23423423402384
  Transmit bytes                             : 23472377
  Transmit packets                           : 94540
Console>
```

Configuring QoS in the Cisco IP Phone 7960

These sections describe QoS in the Cisco IP Phone 7960:

- Understanding How QoS Works in the Cisco IP Phone 7960, page 36-29
- Configuring QoS in the Cisco IP Phone 7960, page 36-30

Understanding How QoS Works in the Cisco IP Phone 7960

Note The Cisco IP Phone 7960 always sets Layer 3 IP precedence and Layer 2 CoS to 5 in voice traffic generated by the phone. The Layer 3 IP precedence and Layer 2 CoS values in voice traffic generated by the phone are not configurable.

You can configure the Cisco IP Phone 7960 access port (see Figure 36-5) to either *trusted* or *untrusted* mode.

- Untrusted mode means that all traffic in 802.1Q or 802.1p frames received through the access port is marked with a configured Layer 2 CoS value. The default Layer 2 CoS value is 0. Untrusted mode is the default when the phone is connected to a Cisco LAN switch.
- Trusted mode means that all traffic received through the access port passes through the phone switch unchanged. Trusted mode is the default when the phone is not connected to a Cisco LAN switch.
- Traffic in frame types other than 802.1Q or 802.1p passes through the phone switch unchanged, regardless of the access port trust state.

Figure 36-5 Configuring QoS on the IP Phone Ports

Configuring QoS in the Cisco IP Phone 7960

These sections describe how to configure QoS in the Cisco IP Phone 7960:

- Setting the Phone Access Port Trust Mode, page 36-30
- Setting the Phone Access Port CoS Value, page 36-30
- Verifying the Phone Access Port QoS Configuration, page 36-31

Setting the Phone Access Port Trust Mode

This example shows how to set the phone access port to the trusted mode:

```
Console> (enable) set port qos 3/7 trust-ext trusted
Port in the phone device connected to port 3/7 is configured to be trusted.
Console> (enable)
```

This example shows how to set the phone access port to the untrusted mode:

```
Console> (enable) set port qos 3/7 trust-ext untrusted
Port in the phone device connected to port 3/7 is configured to be untrusted.
Console> (enable)
```

Setting the Phone Access Port CoS Value

This example shows how to configure the Layer 2 CoS value used by a phone access port in untrusted mode:

```
Console> (enable) set port qos 2/1 cos-ext 3
Port 2/1 qos cos-ext set to 3.
Console> (enable)
```

Verifying the Phone Access Port QoS Configuration

This example shows how to display QoS configuration information:

```
Console> (enable) show port qos 3/4
<...Output Truncated...>
Port  Ext-Trust Ext-Cos
----- --------- -------
 3/4  untrusted       0
<...Output Truncated...>
```