Configuring a VoIP Network

This chapter describes how to configure a Voice-over-IP (VoIP) network on the Catalyst 6500 series switches.

Note 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 6500 series products into your VoIP network.

Note For complete syntax and usage information for the commands used in this chapter, refer to the *Catalyst 6500 Series Switch Command Reference* publication.

This chapter consists of these sections:

Hardware and Software Requirements

The hardware and software requirements for the Catalyst 6500 series switches and Cisco CallManager are as follows:

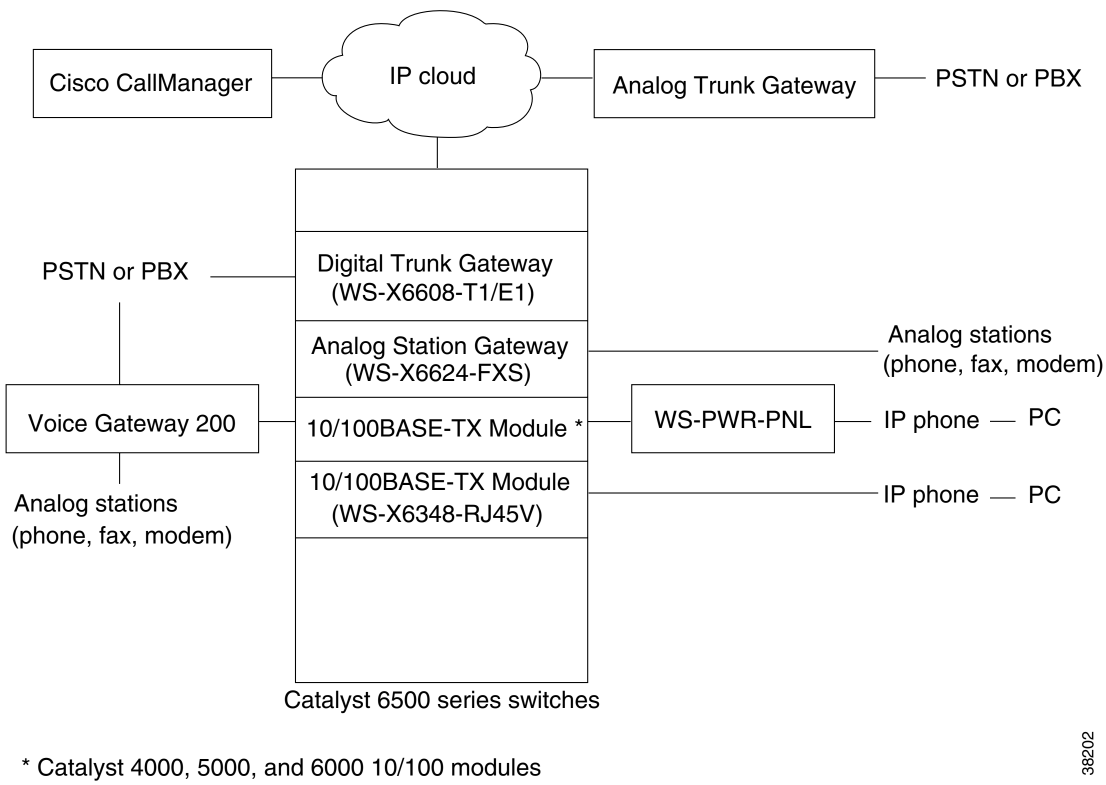
- Catalyst 4000, 5000, and 6500 series switches running supervisor engine software release 6.1(1) or later releases
- Cisco CallManager release 3.0 or later releases

Understanding How a VoIP Network Works

A telephony system built on an IP network instead of the traditional circuit-switched private branch exchange (PBX) network is called an IP PBX system. See Figure 47-1; the individual components of this system are described in these sections:

- Cisco IP Phone 7960, page 47-2
- Cisco CallManager, page 47-4
- Access Gateways, page 47-4
- How a Call Is Made, page 47-7

Figure 47-1 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.

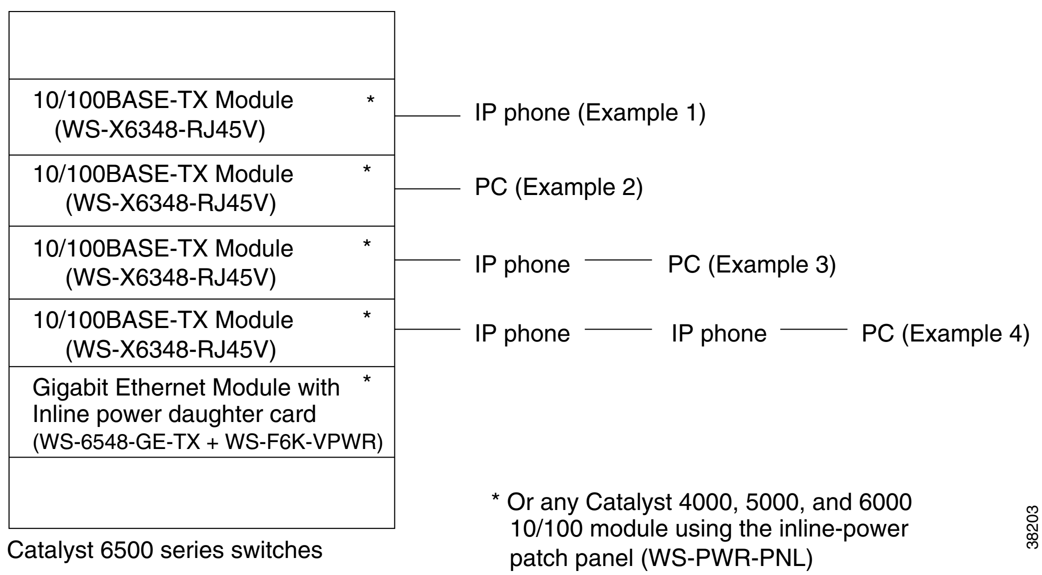
The IP phone is Dynamic Host Configuration Protocol (DHCP) capable. Optionally, the IP phone can be programmed with a static IP address.

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.
- Ethernet switching module with voice daughter card installed—Provides inline power to the IP phone.
- WS-PWR-PNL—Inline-power patch panel provides inline power to the IP phone. The inline patch panel allows the IP phone to be connected to existing Catalyst 4000, 5000, and 6500 series 10/100BASE-TX switching modules.

Examples 1 through 4 in Figure 47-2 show how to connect the Cisco IP Phone 7960 and PCs to the Catalyst 6500 series switch.

Figure 47-2 Connecting the Cisco IP Phone 7960 to the Catalyst 6500 Series Switch



Example 1: Single Cisco IP Phone 7960

Example 1 shows one IP phone that is connected to the 10/100 port on the Catalyst 6500 series 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 that is connected to the 10/100 port on the Catalyst 6500 series switch. The PC is wall powered.

Example 3: One Cisco IP Phone 7960 and One PC

Example 3 shows one IP phone that is connected to the 10/100 port on the Catalyst 6500 series switch and one PC that is 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 6500 series 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 that are connected to the 10/100 port on the Catalyst 6500 series switch and one PC that is 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 6500 series 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; its 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 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. 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 that are described in this chapter, refer to the *Cisco CallManager Administration Guide,* the *Configuration Notes for Cisco CallManager,* and the *Cisco CallManager 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.

These sections describe the gateways:

- Analog Station Gateway, page 47-4
- Analog Trunk Gateway, page 47-5
- Digital Trunk Gateway, page 47-5
- Converged Voice Gateway, page 47-7

Analog Station Gateway

The Catalyst 6500 series 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.

To configure the analog station interfaces, see the "Configuring VoIP on the Switch" section on page 47-9. The module features are listed in Table 47-1.

Table 47-1 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 and 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.

Digital Trunk Gateway

The Catalyst 6500 series 8-port T1/E1 PSTN interface module can support both digital T1/E1 connectivity to the PSTN or transcoding and conferencing. The module 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 support transcoding and conferencing. Transcoding and conferencing functions are mutually exclusive. For every transcoding port in use, one less conferencing port is available and vice versa.

To configure the 8-port T1/E1 PSTN interfaces, see the "Configuring VoIP on the Switch" section on page 47-9. The module features are listed in Table 47-2.

Table 47-2 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 pass-through
Silence suppression, voice activity detection
Line echo cancellation
Common channel signaling 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 common channel signaling
ISDN Primary Rate Interface 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.
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
FDL[1] is a link management protocol that is used to help diagnose problems and gather statistics on T1 lines

1. FDL = Facilities Data Link

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 provides a 10/100BASE-T 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 following:
  - PSTN using FXO emulation
  - T1 channel bank using FXS emulation
  - 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, refer to the documentation that shipped with the gateway.

How a Call Is Made

An IP phone connects 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 includes 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 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 the 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.

Understanding How VLANs Work

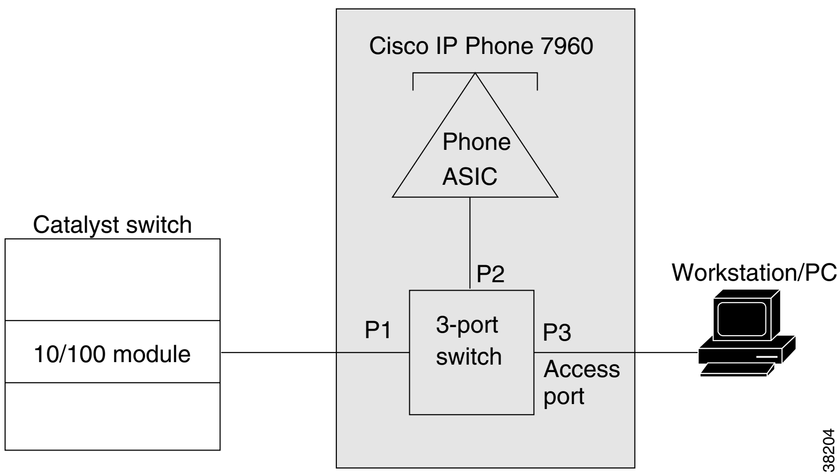
This section describes native VLANs and auxiliary VLANs. This section uses the following terminology:

- 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

Note For more information about VLANs, see Chapter 11, "Configuring VLANs."

Figure 47-3 shows how you can connect a Cisco IP Phone 7960 to a Catalyst 6500 series switch.

Figure 47-3 Switch-to-Phone Connections



When the IP phone connects to a 10/100 port on the Catalyst 6500 series 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 47-2).

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) that are connected to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.

You can resolve these issues by isolating the voice traffic onto a separate VLAN on each of the ports that are connected to a phone. The switch port that is configured for connecting a phone would have separate VLANs configured for carrying the following:

- Voice traffic to and from the IP phone (auxiliary VLAN)
- Data traffic to and from the PC that is 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.

Configuring VoIP on the Switch

This section describes the command-line interface (CLI) commands and the procedures that are used to configure the Catalyst 6500 series switch for VoIP operation:

- Voice-Related CLI Commands, page 47-9
- Configuring Per-Port Power Management, page 47-10
- Configuring Auxiliary VLANs on Catalyst LAN Switches, page 47-20
- Configuring the Access Gateways, page 47-22
- Displaying Active Call Information, page 47-28
- Configuring QoS in the Cisco IP Phone 7960, page 47-30
- Configuring a Trusted Boundary to Ensure Port Security, page 47-33

Note For information on using automatic voice configuration, see the "Using Automatic Voice Configuration" section on page 47-37.

Note You must enable Cisco Discovery Protocol (CDP) on the Catalyst 6500 series switch port that is connected to the IP phone in order to communicate the auxiliary VLAN ID, per-port power management details, and quality of service (QoS) configuration information.

Voice-Related CLI Commands

Table 47-3 lists the CLI commands that are described in the configuration procedures.

Table 47-3 Voice-Related CLI Command Module and Platform Support

CLI Commands	Ethernet Module[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

Table 47-3 Voice-Related CLI Command Module and Platform Support (continued)

CLI Commands	Ethernet Module[1]	WS-X6608-T1/E1[2]	WS-X6624-FXS[3]
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. Ethernet Module = Ethernet 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 6500 series switch only; XX = Command supported on Catalyst 4000, 5000, and 6500 series switches (note that all modules listed in Table 47-3 are supported only on Catalyst 6500 series switches).

Configuring Per-Port Power Management

This section describes per-port power management and the CLI commands that are 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 20-14.

Note This section applies to Ethernet switching modules with voice daughter cards only. For information on powering IP phones that are connected to other Ethernet switching modules, refer to the *Catalyst Family Inline-Power Patch Panel Installation Note* publication.

For each IP phone that is connected to an Ethernet switching module with the voice daugher card installed, 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 that is connected to the switch port, the second phone cannot be powered by the switch.

This section describes the following:

- Phone Detection Summary, page 47-15
- Error Detection and Handling, page 47-16
- Setting the Power Mode of a Port or Group of Ports, page 47-17
- Setting the Default Power Allocation, page 47-18
- Displaying the Power Status for Modules and Individual Ports, page 47-18
- Displaying the Power Status for Modules and Individual Ports, page 47-19

Using show Commands to Display Module Type and Version Information

To determine if an Ethernet switching module has a voice daughter card installed, enter the **show module** command and look at the "Sub" field. For example, in the following display, the 10/100BASE-TX module in slot 8 does not have a voice daughter card, while the module in slot 9 does have a voice daughter card.

To display module status and information, perform this task in normal mode:

Task	Command
Display module status and information.	**show module** [*mod*]

This example shows that there is a submodule field that provides information about submodules. The EARL daughter card is treated as a submodule while the Multilayer Switch Feature Card (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)
```

To display the version of modules and submodules, perform this task in normal mode:

Task	Command
Display the version of modules and submodules.	**show version** [*mod*]

This example shows the version of modules and submodules:

```
Console> (enable) show version 2
Mod Port Model               Serial #    Versions
--- ---- ------------------- ----------- --------------------------------------
2   2    WS-X6K-SUP2-2GE     SAD04450LF1 Hw : 1.1
                                         Fw : 6.1(2)
                                         Fw1: 6.1(3)
                                         Sw : 6.3(0.62)PAN
                                         Sw1: 6.3(0.62)PAN
         WS-F6K-PFC2         SAD04440HVU Hw : 1.0
Console>
```

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 the **set port inlinepower** CLI command):

- **Auto**—The supervisor engine directs the switching module to power up the port *only* if the switching module discovers 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 one of the following:

- on—Power is supplied by the port.
- off—Power is not 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.

These sections provide information that is related to IP phone power requirements and management:

- Unpowered Phone, page 47-13
- Power Requirements, page 47-13
- Wall-Powered Phones, page 47-13
- Powering Off the Phone, page 47-13
- Phone Removal, page 47-14
- High-Availability Support, page 47-14

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* that is required by a phone from the total available system power and 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 off power. The entire cycle is 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 7 W (167 mA at 42 V) 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 7 W. A Cisco IP Phone requiring 6.3 W is plugged into a port. The supervisor engine allocates 7 W 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 a power outage occurs, and the mode is set to **Auto**, the phone loses power, but the switching module discovers the phone and informs 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 situation occurs only 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.

In addition, the switching module 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

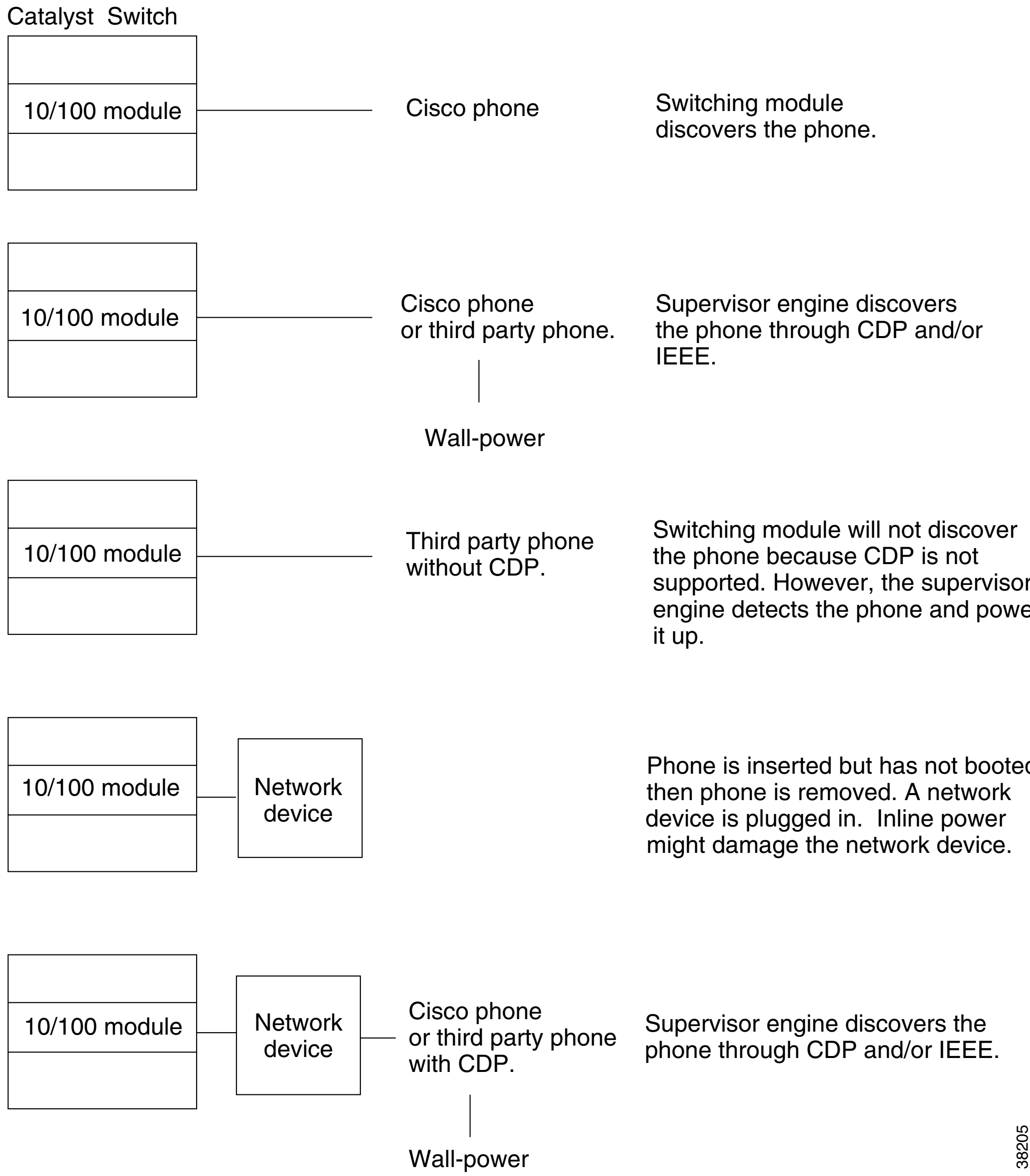
To support high availability during a failover from the active supervisor engine to the standby supervisor engine, the per-port power management and phone status information is synchronized between the active and standby supervisor engines.

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 that is 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 47-4 shows how the system detects a phone that is connected to a Catalyst 6500 series switch port.

Figure 47-4 Power Detection Summary



Error Detection and Handling

This section describes how the Catalyst 6500 series switch handles fault detection and errors that are related to per-port power management.

These sections discuss fault detection and power-management error scenarios:

- Device is Powered but Link is Not Up, page 47-16
- Port is Unable to Provide Inline Power to the Device, page 47-17
- Not Enough Available Power to Power the Device, page 47-17
- Power Supply Configured from Nonredundant to Redundant, page 47-17
- Power Supply Configured from Redundant to Nonredundant, page 47-17

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, this 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 if there is a problem providing inline power to the device and reports this problem to the supervisor engine. This 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 tracks the available power left in the system and does not power up any ports if no available power remains. This 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, you might need to power off some modules to prevent overdrawing power from the power supply. The supervisor engine first powers off and reallocates the power that is supplied by the ports and then starts powering off and reallocating the power that is 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.

Setting the Power Mode of a Port or Group of Ports

To set the power mode of a port or group of ports, perform this task in normal mode:

Task	Command
Set the power mode of a port or group of ports.	**set port inlinepower** *mod/port* {**off** | **auto**}

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

```
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)
```

Setting the Default Power Allocation

The **set inlinepower defaultallocation** command is global and only affects Cisco IP phones. The inline power threshold notification generates a syslog message when inline power usage exceeds the specified threshold. To set the default power allocation, perform this task in privileged mode (the default allocation value is 15400 milliwatts):

Caution The **set inlinepower defaultallocation** command can be harmful when there is not enough power in the system to bring up all connected inline power devices. If you set a small value for the power allocation, all connected inline power devices initially will be powered up. However, after receiving CDP messages, the system will learn that devices are consuming more power and deny power to some of the ports. Setting a small value might also result in the overdrawing of power for some time with unanticipated results, such as hardware failures and unexpected resets.

Note 7000 milliwatts is the maximum power supported for these modules: WS-X6348-RJ21V, WS-X6348-RJ-45V, WS-X6148-RJ-45V, and WS-X6148-RJ21V.

Task	Command
Set the default power allocation.	**set inlinepower defaultallocation** *value*

This example shows how to set the default power allocation:

```
Console> (enable) set inlinepower defaultallocation 9500
Default inline power allocation set to 9500 mWatt per applicable port.
Console> (enable)
```

Displaying the Power Status for Modules and Individual Ports

To display the power status for modules and individual ports, perform this task in normal mode:

Task	Command
Display the power status for modules and individual ports.	**show port inlinepower** [*mod*[*/port*]]

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

```
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)
```

The Operational (Oper) status field descriptions in the display are as follows:

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

Displaying the Power Status for Modules and Individual Ports

To display the power status for modules and individual ports, perform this task in privileged mode:

Task	Command
Display the power status for modules and individual ports.	**show environment power** [*mod*]

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

```
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.
  - Enter the **set port auxiliaryvlan** *mod*[/*port*] *aux_vlan_id* command.

  Note We recommend that you use 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 (enter 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 (enter 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

This section describes the guidelines for configuring auxiliary VLANs:

- An auxiliary VLAN port is operationally a trunk, even though it is not treated like a “normal” trunk port. When an auxiliary VLAN is added to a port and the **set dot1q-all-tagged** command is enabled, the **set dot1q-all-tagged** command tags the native VLAN on the port where the auxiliary VLAN is configured. A port with an auxiliary VLAN configured is not viewed as a dot1q trunk in the **show trunk** command output, but the port acts like a dot1q trunk if the **set dot1q-all-tagged** command is enabled.
- The IP phone and a device that is attached to the phone are in the same VLAN and must be in the same IP subnet if one of the following occurs:
  - They use the same frame type.
  - The phone uses 802.1p frames and the device uses untagged frames.
  - The phone uses untagged frames and the device uses 802.1p frames.
  - The phone uses 802.1Q frames and the auxiliary VLAN equals the native VLAN.
- The IP phone and a device that is attached to the phone cannot communicate if they are in the same VLAN and subnet but use different frame types, 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 a frame type that is used by traffic received from a device attached to the phone’s access port.
- With software release 6.2(1) and later releases, dynamic ports can belong to two VLANs—a native VLAN and an auxiliary VLAN. See Chapter 18, “Configuring Dynamic Port VLAN Membership with VMPS,” for configuration details for auxiliary VLANs.

Configuring Auxiliary VLANs

To configure auxiliary VLANs, perform this task in privileged mode:

Task	Command
Configure auxiliary VLANs.	**set port auxiliaryvlan** *mod*[*/ports*] {*vlan* | **untagged** | **dot1p** | **none**}

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 47-4 lists the **set port auxiliaryvlan** command keywords and their descriptions.

Table 47-4 Keyword Descriptions

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

Verifying Auxiliary VLAN Configuration

To verify auxiliary VLAN configuration status, perform this task in privileged mode:

Task	Command
Verify auxiliary VLAN configuration status.	**show port auxiliaryvlan** {*vlan* | **untagged** | **dot1p** | **none**}

This example shows how to verify auxiliary VLAN configuration 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 that are used to configure the following Catalyst 6500 series access gateway modules:

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

Configuring a 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, you must specify some mandatory parameters as follows:

- If you do not specify DNS parameters, 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.

To configure a port voice interface for DHCP, TFTP, and DNS servers, perform this task in privileged mode:

Task	Command
Configure a port voice interface for DHCP, TFTP, and DNS servers.	**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*]]

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

```
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)
```

Displaying a Port Voice Interface Configuration

To display a port voice interface configuration, perform this task in privileged mode:

Task	Command
Display a port voice interface configuration.	**show port voice interface** [*mod*[*/port*]]

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

```
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)
```

Displaying FDL Statistics

Note Facilities Data Link (FDL) is a link management protocol that is used to help diagnose problems and gather statistics.

To display FDL statistics for the specified ports, perform this task in privileged mode:

Task	Command
Display FDL statistics for the specified ports.	**show port voice fdl** [*mod*[/*port*]]

This example shows how to display FDL statistics for the specified ports:

```
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 47-5 describes the possible fields (depending on the port type queried) in the **show port voice fdl** command output.

Table 47-5 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.

Displaying the Port Configuration for Individual Ports

To display the port configuration for individual ports, perform this task in normal mode:

Task	Command
Display the port configuration for individual ports.	**show port** [*mod*[/*port*]]

This section provides the **show port** command displays for the following gateway modules:

- 8-Port T1/E1 PSTN Interface Module, page 47-25
- 8-Port T1/E1 PSTN Interface Module Configured for Trancoding/Conferencing, page 47-26
- 24-Port FXS Analog Interface Module, page 47-27

8-Port T1/E1 PSTN Interface Module

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

This example shows a transcoding port as MTP and a conference port 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

This example shows that 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

Enter 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.

To display active call information, perform this task in normal mode:

Task	Command
Display active call information.	**show port voice active** [*mod/port*] [**all** | **call** | **conference** | **transcode**] [*ipaddr*]

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 you configure 8-port T1/E1 PSTN interfaces 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 that are 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, this example 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 47-31
- Configuring QoS in the Cisco IP Phone 7960, page 47-31

Note For information on using automatic QoS, see Chapter 44, "Using Automatic QoS."

Note For information on using automatic voice configuration, see the "Using Automatic Voice Configuration" section on page 47-37.

Understanding How QoS Works in the Cisco IP Phone 7960

Note The Cisco IP Phone 7960 always sets the 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.

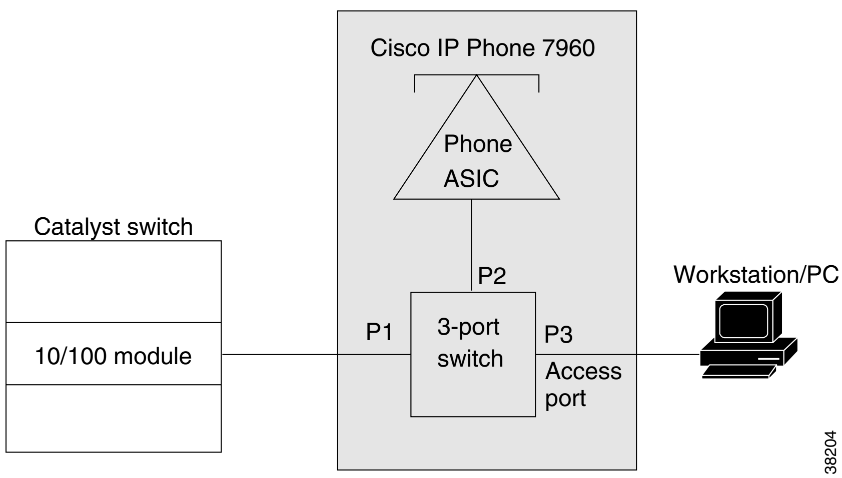
You can configure the Cisco IP Phone 7960 access port (see Figure 47-5) to either *trusted* or *untrusted* mode.

In untrusted mode, all the traffic in 802.1Q or 802.1p frames that are 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.

In trusted mode, all traffic that is 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 47-5 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 47-32
- Setting the Phone Access Port CoS Value, page 47-32
- Verifying the Phone Access Port QoS Configuration, page 47-32

Setting the Phone Access Port Trust Mode

To set the phone access port trust mode, perform this task in privileged mode:

Task	Command
Set the phone access port trust mode.	**set port qos** *mod/ports...***trust-ext** {**trusted** | **untrusted**}

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

To set the phone access port CoS value, perform this task in privileged mode:

Task	Command
Set the phone access port CoS value.	**set port qos** *mod/ports* **cos-ext** *cos_value*

This example shows how to set the Layer 2 CoS value that is 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

To verify the phone access port QoS configuration, perform this task in normal mode:

Task	Command
Verify the phone access port QoS configuration.	**show port qos** [*mod*[*/port*]]

This example shows how to verify the phone access port QoS configuration:

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

Configuring a Trusted Boundary to Ensure Port Security

This section describes the trusted boundary feature that is used to prevent security problems if users disconnect their PCs from networked Cisco IP Phones and plug them directly into the switch port to take advantage of the QoS **trust-cos** switch port settings.

These sections describe the trusted boundary feature:

Supported Cisco IP Phones

These Cisco IP phones are supported with the trusted boundary feature:

- Cisco IP Phone 7910
- Cisco IP Phone 7935
- Cisco IP Phone 7940
- Cisco IP Phone 7960

QoS and Cisco IP Phone Configuration

Cisco IP Phones are directly attached to the Catalyst 6500 series switch ports. Traffic coming from the phone entering the switch is typically marked with a tag using the 802.1Q header. The header contains the VLAN information as well as the class of service (CoS) 3-bit field. The CoS determines the priority of the packet. For most Cisco IP Phone configurations, the traffic coming from the phone and entering the switch is trusted to ensure that the voice traffic is properly prioritized over other types of traffic in the network. The port on the switch where the phone is attached is configured to **trust-cos**, which means that the port trusts the CoS labeling of all packets arriving on that port.

QoS, Cisco IP Phone, and PC Configuration

A PC or workstation can be attached to the Cisco IP Phone. The phone has a built-in hub that mixes the traffic coming from the PC, the phone, and the switch port. To distinguish traffic coming from the PC from traffic coming from the phone, use the 3-bit CoS labels.

You need to configure QoS features on the phone for proper labeling to occur. QoS configuration information is sent to the phone using the Cisco Discovery Protocol (CDP) from the switch. The QoS configuration determines the trust state of the phone and the classification information (Ext-Cos). The phone supports two trust states:

- Trusted
- Untrusted and marked with a new COS value (Ext-Cos)

If the phone is in trusted mode, all the labels that are produced by the PC are sent directly through the phone toward the switch, untouched. If the phone is in untrusted mode, all traffic coming from the PC is marked with the Ext-Cos value before it is sent to the switch.

For most setups, the PC or workstation that is attached to the phone is unable to tag its packets. In these cases, all the traffic that comes from the PC and enters the switch through the phone, is marked with the “default ext-cos” configured on the phone.

In some cases, the PC can tag its own packets. A PC running Windows 2000 can be configured to send dot1q frames of any priority. To solve this problem, the phones should be configured to be untrusted, which marks all the traffic coming from the PC to the appropriate priority.

The trusted boundary feature prevents users from taking advantage of the trust-cos setting on the switch by disconnecting their phone from the network and plugging their PC directly into the switch port. It uses CDP to detect the phone’s presence on a port. If the phone leaves the port, the feature automatically configures the port to be untrusted, which solves the security issue.

The trusted boundary feature is implemented using a configuration command to create a new type of trust. The command allows you to configure port trust based on the presence of a given device on a port. For Cisco IP Phones, you configure the trust as “**trust-device ciscoipphone**.”

Configuration Guidelines

This section describes the guidelines for configuring the trusted boundary feature:

- Common Open Policy Service (COPS) considerations

  COPS directly affects how QoS parameters are applied. A port may have either a local policy or a COPS policy. This setting specifies whether the port should get its QoS configuration information from the local configuration or through a COPS server. If COPS is enabled on a port and is also globally enabled, the policy that is specified by the COPS server applies. If COPS is disabled and/or the run-time policy is local, the local configuration QoS policy applies. The extended trust boundary feature overrides the “local” policy on a port.

- QoS configuration support

  All QoS port trust configuration settings are supported (**trust-cos**, **trust-ipprec**, **trust-dscp**) but you should use **trust-cos** for the Cisco IP Phone networks.

- System log messaging

  New QoS syslogs were added for the trusted boundary feature to notify you of changes to a port’s trust state and to warn of improper configuration. To see these syslogs, set the QoS logging level to 5 (**set logging level qos 5**). The default is 3. Refer to the *Catalyst 6500 Series, Catalyst 4000 Family, Catalyst 2948G, and Catalyst 2980G Switches System Message Guide* publication for descriptions of the syslogs.

- Final run-time port trust value

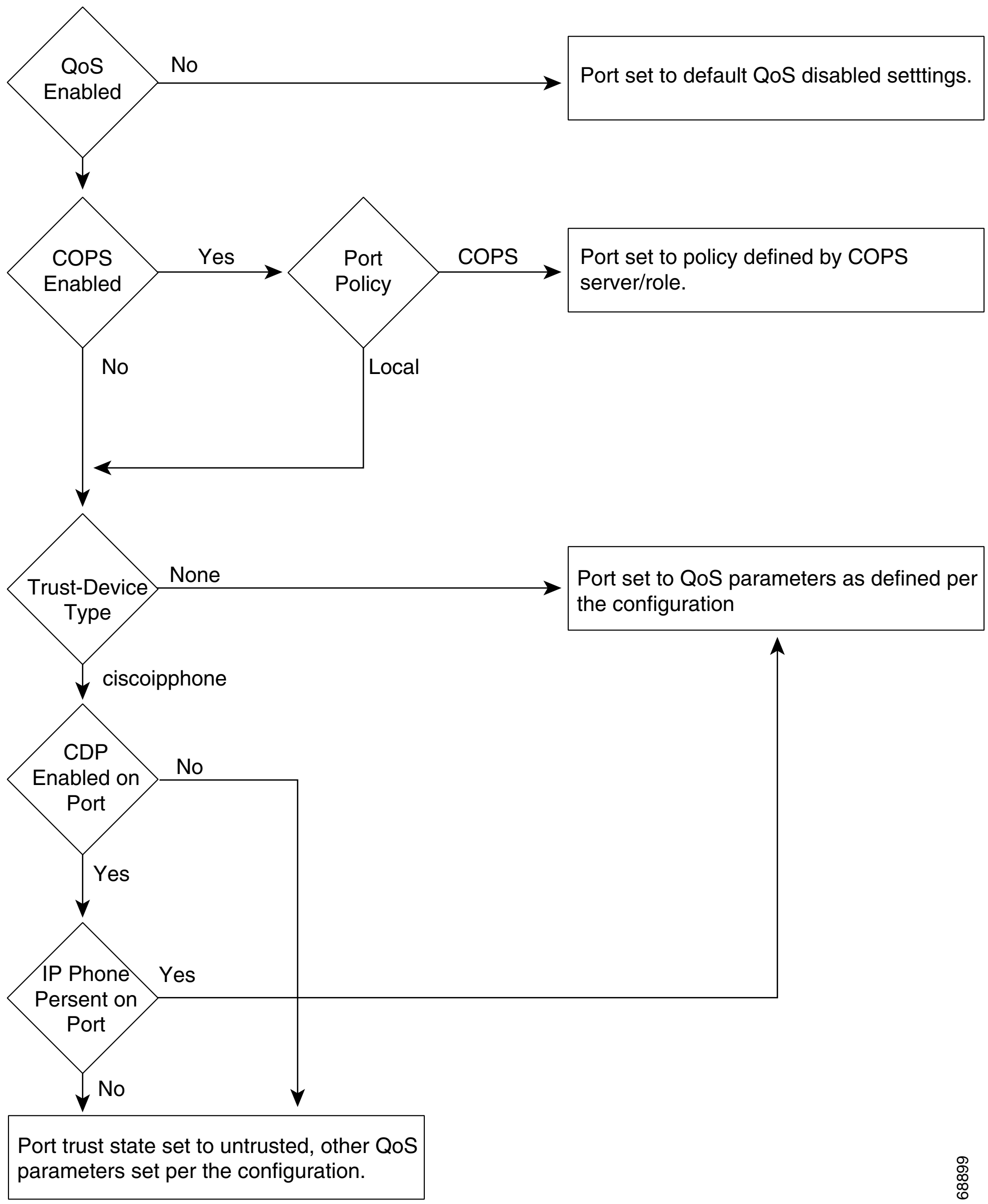
  The final run-time port trust on any port is dependent on the following:

  - Trusted boundary feature configuration
  - Phone’s presence on the port
  - QoS configuration
  - COPS configuration

  To enable the trusted boundary feature, you must enable QoS and you must enable CDP globally and on the port, running in version 2 mode. You must set COPS to local policy (the COPS default) or to disabled (the COPS default). When **ciscoipphone** is configured as the trust-device on the port, the feature is enabled and detects the presence of a Cisco IP Phone and sets the trust values.

  See Figure 47-6 to determine the final trust value on a port.

Figure 47-6 Determining the Final Trust Value of a Port



Configuring a Trusted Boundary

These sections describe how to configure the trusted boundary feature:

- Default Configuration, page 47-36
- Specifying a Cisco IP Phone as the Trust Device, page 47-36
- Verifying a Port's Trust-Device State, page 47-36

Default Configuration

The default setting for all ports is **trust-device none**.

Specifying a Cisco IP Phone as the Trust Device

To specify a Cisco IP Phone as the trust device, perform this task in privileged mode:

Task	Command
Specify a Cisco IP Phone as the trust device.	**set port qos** *mod/ports...***trust-device** [**ciscoipphone** | **none**]

This example shows how to trust only Cisco IP phones on port 4/1:

```
Console> (enable) set port qos 4/1 trust-device ciscoipphone
Port 4/1 set to only trust device of type ciscoIPPhone.
Console> (enable)
```

This example shows how to disable the device trust on port 4/1:

```
Console> (enable) set port qos 4/1 trust-device none
Port 4/1 trust device feature disabled.
Console> (enable)
```

Verifying a Port's Trust-Device State

To verify a port's trust-device state, perform this task in normal mode:

Task	Command
Verify a port's trust-device state.	**show port qos** [*mod*[/*port*]]

When the trusted boundary feature is active, the run-time trust state of the port changes depending on the presence of the phone.

Note The moment the phone leaves the switch port, there is a slight convergence time for the port to change to the untrusted state (a maximum time of 15 seconds).

This example shows how to verify the trust-device state and trust state on port 4/1:

```
Console> (enable) show port qos 4/1

<truncated ...>

Port  TxPort Type  RxPort Type  Trust Type   Trust Type    Def CoS Def CoS
                                config       runtime       config  runtime
----- ------------ ------------ ------------ ------------- ------- -------
 4/1        1p3q1t       1p1q0t    trust-cos    trust-cos*       0       0

Port  Ext-Trust Ext-Cos Trust-Device

----- --------- ------- ------------

 4/1  untrusted       0 ciscoIPPhone


(*)Runtime trust type set to untrusted.

Config:
Port  ACL name                         Type
----- -------------------------------- ----
No ACL is mapped to port 4/1.

Runtime:
Port  ACL name                         Type
----- -------------------------------- ----
No ACL is mapped to port 4/1.
Console> (enable)
```

Using Automatic Voice Configuration

Automatic voice configuration consists of two macros that simplify voice configuration on the Catalyst 6500 series switches. The automatic voice configuration macros cover all the voice configuration tasks required for implementing the recommended Architecture for Voice, Video, and Integrated Data (AVVID) settings for a voice port.

Automatic voice configuration focuses on the voice networks that are built using the Cisco IP Phone 79xx series and the Cisco SoftPhone. With automatic voice configuration, you use the **ciscoipphone** or **ciscosoftphone** keywords to initiate macros that specify the type of voice parameters that you desire on a particular port.

Automatic voice configuration is described in these sections:

Understanding Automatic Voice Configuration Macros

When you execute the automatic voice configuration macros on a port using the **ciscoipphone** or **ciscosoftphone** keywords, the following features are implemented:

- The port is enabled.
- The Layer 2 protocol is disabled for CDP, STP, and VTP.
- The port membership is set to "static."
- The **set port host** command is executed on the port.
- The specified data VLAN is associated with the port.
- The global automatic QoS command is executed.

When you execute the **ciscoipphone** keyword on a port, in addition to the previous features, the following features are also implemented:

- The specified auxiliary VLAN is associated with the port.
- Inline power is enabled.
- CDP is enabled globally and on the port.
- CDP is configured to version v2.
- The port-based automatic QoS command for the Cisco IP phone is executed.

When you execute the **ciscosoftphone** keyword on a port, in addition to the previous features, the following features are also implemented:

- The auxiliary VLAN for the port is set to "none."
- The port-based automatic QoS command for the Cisco SoftPhone is executed.

Automatic Voice Configuration—Cisco IP Phone

In most configurations, the Cisco IP Phone 79xx is connected directly to the Catalyst switch port. Optionally, you can attach a PC to the phone and use the phone as a hop to the switch.

Typically, the traffic that comes from the phone and enters the switch is marked with a tag using the 802.1Q/p header. The header contains the VLAN information and the CoS 3-bit field. The CoS determines the priority of the packet. The switch uses the CoS field to distinguish the PC traffic from the phone traffic. The switch can also use the DSCP field for the same purpose.

In most Cisco IP Phone 79xx configurations, the traffic that comes from the phone and enters the switch is trusted. You set the port trust to trust-cos to properly prioritize the voice traffic over other types of traffic in the network.

The Cisco IP Phone 79xx has a built-in switch that mixes the traffic that comes from the PC, the phone, and the switch port. The Cisco IP Phone 79xx has trust and classification capabilities that you need to configure.

Ports that connect the IP phones need to have several features enabled or disabled. Automatic voice configuration ensures that the necessary features are enabled. Most of these features are implemented when you execute the **set port host** command (such as disabling channels, enabling PortFast, and so on). A VLAN and an auxiliary VLAN must be configured on the port for QoS to work. Inline power needs to be enabled (if available) and CDP must be enabled for the trusted boundary feature to work. QoS configuration is handled by the automatic QoS feature (see Chapter 44, "Using Automatic QoS").

Automatic Voice Configuration—Cisco Softphone

The Cisco SoftPhone is a software product that runs on a standard PC and emulates an IP phone. The main difference between the Cisco SoftPhone and the Cisco IP Phone 79xx is that the Cisco SoftPhone marks its voice traffic through a DSCP, while the Cisco IP Phone 79xx marks its traffic through a CoS. The QoS settings on the switch accommodate this behavior by trusting the Layer 3 marking of the traffic entering the port. All other behavior is similar to the Cisco IP Phone 79xx. Features such as CDP do not need to be enabled because the trusted boundary feature does not support Cisco SoftPhone.

Automatic Voice Configuration Guidelines and Restrictions

These sections provide the configuration guidelines and restrictions for automatic voice configuration:

Supported Phones

When you use automatic voice configuration with the **ciscoipphone** keyword, some of the QoS configuration requires phone-specific configuration (trust-ext, ext-cos) which is supported only on the following phones: Cisco IP Phone 7910, Cisco IP Phone 7940, Cisco IP Phone 7960, and Cisco IP Phone 7935. However, the **ciscoipphone** keyword is not exclusive to these models only; any phone can benefit from all the other QoS settings that are configured on the switch.

Cisco SoftPhone is supported through the **ciscoipsoftphone** keyword.

CDP Dependencies

To configure the QoS settings and the trusted boundary feature on the Cisco IP Phone, you must enable Cisco Discovery Protocol (CDP) version 2 or later on the port.

You need to enable CDP only for the **ciscoipphone** QoS configuration; CDP does not affect the other components of the automatic voice configuration feature.

EtherChannel Considerations

The automatic voice configuration commands do not support channeling.

PFC/PFC2 Support

No PFC or PFC2 is required for the **ciscoipphone** keyword. A PFC or PFC2 is required for the **ciscosoftphone** keyword.

Module Support

The **ciscoipphone** keyword is only supported on 10/100- and 10/100/1000-Ethernet ports.

The **ciscosoftphone** keyword is supported on all Ethernet ports.

CLI Interface for Automatic Voice Configuration

These sections describe the CLI interface for automatic voice configuration:

- Command Description, page 47-40
- ciscoipphone Command Output, page 47-40
- ciscosoftphone Command Output, page 47-41

Command Description

You must specify either the **ciscoipphone** or **ciscosoftphone** keywords and a data VLAN. Specifying an auxiliary VLAN is optional for the **ciscoipphone** keyword. RSPAN and private VLANs are not supported. The command syntax for automatic voice configuration is as follows:

```
Console> (enable) set port macro
Usage: set port macro <mod/ports..> ciscoipphone vlan <vlan> [auxvlan <auxvlan>]
       set port macro <mod/ports..> ciscosoftphone vlan <vlan>
Console> (enable)
```

ciscoipphone Command Output

When you execute the **ciscoipphone** keyword, the following displays (specifying the auxiliary VLAN is optional):

```
Console> (enable) set port macro 3/1 ciscoipphone vlan 2 auxvlan 3
Port 3/1 enabled.
Layer 2 protocol tunneling disabled for CDP STP VTP on port(s) 3/1.
Port 3/1 vlan assignment set to static.
Spantree port fast start option set to default for ports 3/1.
Port(s) 3/1 channel mode set to off.

Warning: Connecting Layer 2 devices to a fast start port can cause
temporary spanning tree loops. Use with caution.

Spantree port  3/1 fast start enabled.
Dot1q tunnel feature disabled on port(s)  3/1.
Port(s)  3/1 trunk mode set to off.
VLAN  Mod/Ports
---- -----------------------
2     2/1
      3/1
      16/1
AuxiliaryVlan Status   Mod/Ports
------------- --------
------------------------------------------------------
3             inactive 3/1

Vlan 3 is not active.
Inline power for port 3/1 set to auto.
```

```
CDP enabled globally
CDP enabled on port 3/1.
CDP version set to v2
........
All ingress and egress QoS scheduling parameters configured on all ports.
CoS to DSCP, DSCP to COS, IP Precedence to DSCP and policed dscp maps
configured.  Global QoS configured.
Port 3/1 ingress QoS configured for Cisco IP Phone.
Macro completed on port 3/1.
Console> (enable)
```

If you do not specify an auxiliary VLAN, the following warning message displays:

```
Console> (enable) set port macro 3/1 ciscoipphone vlan 2
Warning:  All inbound QoS tagging information will be lost as no auxiliary
vlan was specified.
Do you want to continue (y/n) [n]?
```

ciscosoftphone Command Output

When you execute the **ciscosoftphone** keyword, the following displays:

```
Console> (enable) set port macro 3/1 ciscosoftphone vlan 32
Port 3/1 enabled.
Layer 2 protocol tunneling disabled for CDP STP VTP on port(s) 3/1.
Port 3/1 vlan assignment set to static.
Spantree port fast start option set to default for ports 3/1.
Port(s) 3/1 channel mode set to off.

Warning: Connecting Layer 2 devices to a fast start port can cause
temporary spanning tree loops. Use with caution.

Spantree port  3/1 fast start enabled.
Dot1q tunnel feature disabled on port(s)  3/1.
Port(s)  3/1 trunk mode set to off.
Vlan 32 configuration successful
VLAN 32 modified.
VLAN 2 modified.
VLAN  Mod/Ports
---- -----------------------
32    3/1
      16/1
Port 3/1 will not send out CDP packets with AuxiliaryVlan information.
Executing autoqos........
All ingress and egress QoS scheduling parameters configured on all ports.
CoS to DSCP, DSCP to COS, IP Precedence to DSCP and policed dscp maps
configured.  Global QoS configured.
Port 3/1 ingress QoS configured for Cisco Softphone.
Macro completed on port 3/1.
Console>> (enable)
```

Detailed Automatic Voice Configuration Statements

These sections provide detailed automatic voice configuration statements:

- ciscoipphone Configuration Statement, page 47-42
- ciscosoftphone Configuration Statement, page 47-42

ciscoipphone Configuration Statement

The **ciscoipphone** automatic voice configuration command results in the following configuration:

```
set port macro mod/port ciscoipphone vlan vlan [auxvlan auxvlan]
----------------------------------------------------------
set port enable mod/port
set port l2protocol-tunnel mod/port cdp stp vtp disable
set port membership mod/port static
set port host mod/port
set vlan mod/port vlan
set port auxiliaryvlan mod/port auxvlan  (set to none if not specified)
set port inlinepower mod/port auto (if supported by module)
set cdp enable
set cdp enable mod/port
set cdp version v2
set qos autoqos
set port qos mod/port autoqos voip ciscoipphone
```

ciscosoftphone Configuration Statement

The **ciscosoftphone** automatic voice configuration command results in the following configuration:

```
set port macro mod/port ciscosoftphone vlan vlan
----------------------------------------------------------
set port enable mod/port
set port l2protocol-tunnel mod/port cdp stp vtp disable
set port membership mod/port static
set port host mod/port
set vlan mod/port vlan
set port auxiliaryvlan mod/port none
set qos autoqos
set port qos mod/port autoqos voip ciscosoftphone
```

How to Use Automatic Voice Configuration in Your Network

Depending on the interface and what is connected to it, you will need to execute different automatic voice macros. For each port, execute the port-based macro command with the appropriate keyword as shown in Table 47-6:

Table 47-6 Using Automatic Voice Configuration Keywords

Keyword	Port Type
ciscoipphone	Ports that connect only a Cisco IP Phone 79xx.
ciscoipphone	Ports that connect a Cisco IP Phone 79xx with a PC connected to the 79xx.
ciscoipphone	Ports that connect a Cisco IP Phone 79xx with a PC connected to the 79xx running Cisco SoftPhone[1].
ciscosoftphone	Ports that connect a PC running Cisco SoftPhone without a Cisco IP Phone 79xx.

1. For cases where ports connect a Cisco IP Phone 79xx with a PC running Cisco SoftPhone, the control traffic through CTI communication with the Cisco CallManager is tagged but is remarked to DSCP 0.