CHAPTER 55

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 that are related to VoIP, the primary focus of the chapter is to provide configuration information for integrating the Catalyst 6500 series products into your VoIP network.

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

This chapter consists of these sections:

Hardware and Software Requirements

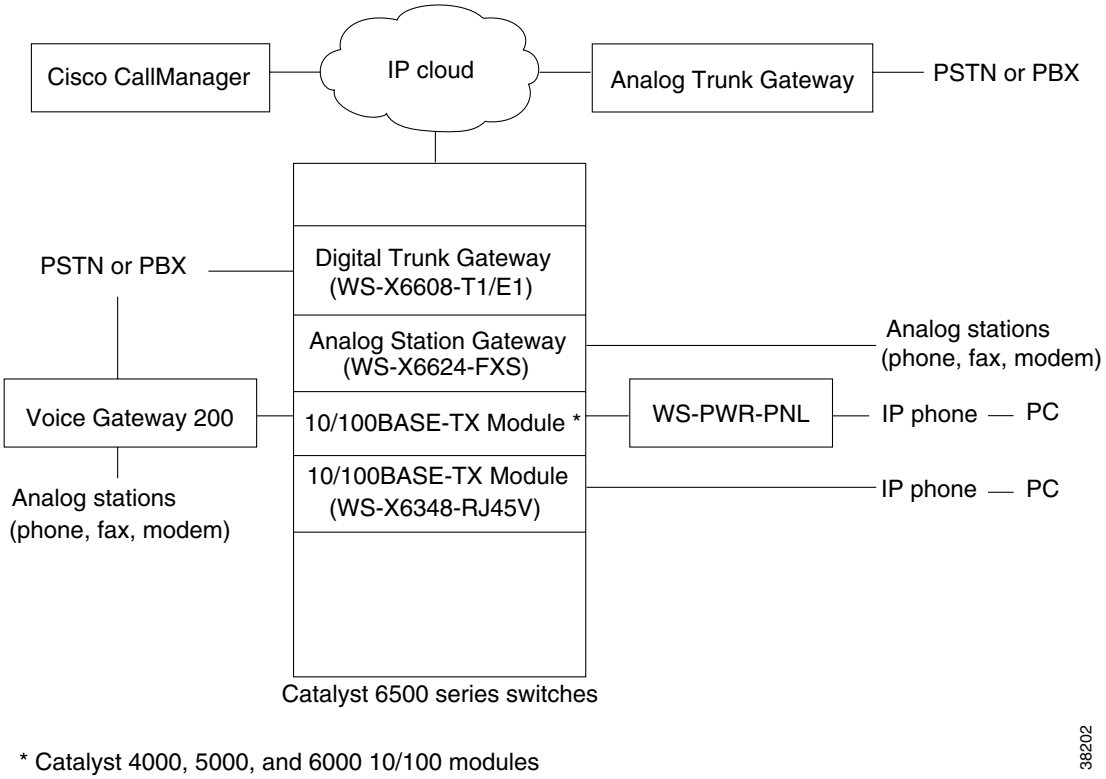
The hardware and software requirements for the Catalyst 6500 series switches and Cisco CallManager are as follows:

- Catalyst 4500 series, 5000 family, and Catalyst 6500 series switches running supervisor engine software release 6.1(1) or later releases
- Catalyst 4500 series and Catalyst 6500 series switches running supervisor engine software release 8.2(1) or later releases for IEEE 802.3af compliance
- 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 55-1.) The system's components are described in these sections:

Figure 55-1 IP PBX System



Cisco IP Phone 7960

The Cisco IP Phone 7960 provides the connectivity to the IP PBX system. The IP phone has two RJ-45 jacks for connecting to the 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 inline power is designed to work in cables from Category 3, Category 4, Category 5, and later up to 100 meters. The inline power works with IBM Token Ring STP cable of 100 meters when used with a Token Ring to Fast Ethernet adapter (LanTel Silver Bullet SB-LN/VIP-DATA adapter).

The IP phone is Dynamic Host Configuration Protocol (DHCP) capable. Optionally, you can program the IP phone 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 modules with the voice daughter card installed—Provides the inline power to the IP phone.

WS-PWR-PNL (inline-power patch panel)—Provides the inline power to the IP phone. The inline patch panel allows the IP phone to connect to existing Catalyst 4500 series, 5000 family, and 6500 series 10/100BASE-TX switching modules.

WS-PWR-PNL (inline-power patch panel)—Provides the inline power to the IP phone. The inline patch panel allows the IP phone to connect to existing Catalyst 4500 series, 5000 family, and 6500 series 10/100BASE-TX switching modules.

WS-X6148-RJ-45 10/100 switching module with either the WS-F6K-VPWR inline-power field-upgrade module or the WS-F6K-FE48-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

WS-X6148-RJ-21 10/100 switching module with either the WS-F6K-VPWR inline-power field-upgrade module or the WS-F6K-FE48-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

WS-X6148X2-RJ-45 10/100 switching module with the WS-F6K-FE96-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

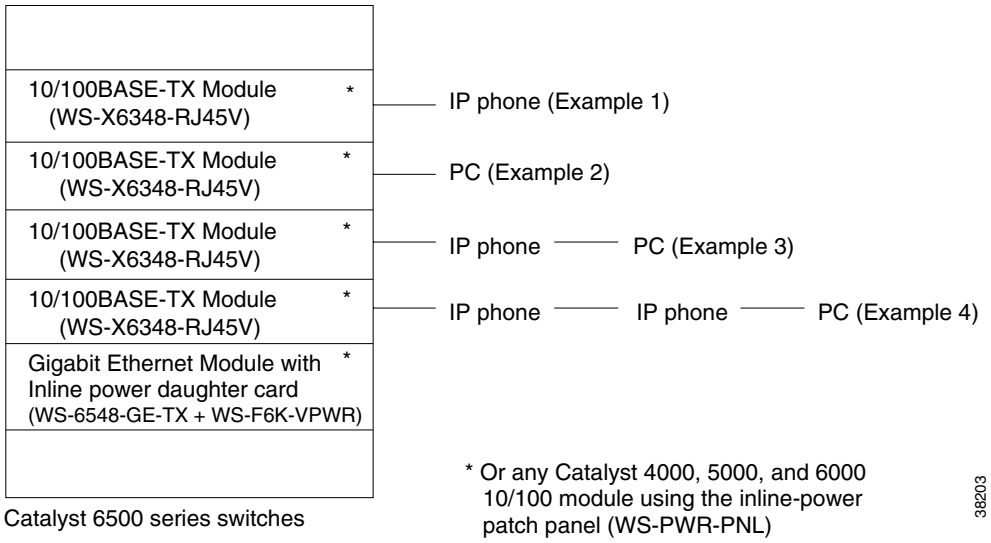
WS-X6148X2-RJ-21 10/100 switching module with the WS-F6K-FE96-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

WS-6548-GE-TX Gigabit Ethernet switching module with either the WS-F6K-VPWR-GE inline-power field-upgrade module or the WS-F6K-GE48-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

WS-6148-GE-TX Gigabit Ethernet switching module with either the WS-F6K-VPWR-GE inline-power field-upgrade module or the WS-F6K-GE48-AF inline-power field-upgrade module—Provides the inline power to the IP phone.

Figure 55-2 shows how to connect the Cisco IP Phone 7960 and PCs to the Catalyst 6500 series switch.

Figure 55-2 Connecting the Cisco IP Phone 7960 to the Catalyst 6500 Series Switch



The examples shown in Figure 55-2 are described in detail as follows:

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 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.

For more information on configuring the 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 the calls between the phones, integrating traditional PBX functionality with the corporate IP network. Cisco CallManager manages the components of the IP PBX system, the phones, the 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 the calls possible across multiple zones. Cisco CallManager can work with the 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

The access gateways allow the IP PBX system to talk to the existing PSTN or PBX systems. The 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 55-5
- Analog Trunk Gateway, page 55-6
- Digital Trunk Gateway, page 55-6
- Converged Voice Gateway, page 55-7

Analog Station Gateway

The Catalyst 6500 series 24-port Foreign Exchange Station (FXS) analog interface module allows the 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 a Switch" section on page 55-10. The module features are listed in Table 55-1.

Table 55-1 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

Table 55-1 24-Port FXS Analog Interface Module Features (continued)

Digital Signal Processing Per Port
Line echo cancellation (32 ms)
Impedance (600 ohms)
Programmable analog gain, signaling timers
Fax pass-through
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

The 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 the ring voltage as it is not connected to the POTS end devices such as the 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 the T1/E1 interfaces or the ports support transcoding and conferencing. The 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 a Switch" section on page 55-10. The module features are listed in Table 55-2.

Table 55-2 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 the 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 the 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 the 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 the voice packets are sensitive to delays, TCP is not suitable for voice transmission because the timeouts and retries increase the delay between the 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 routes it to the PSTN.

Understanding How VLANs Work

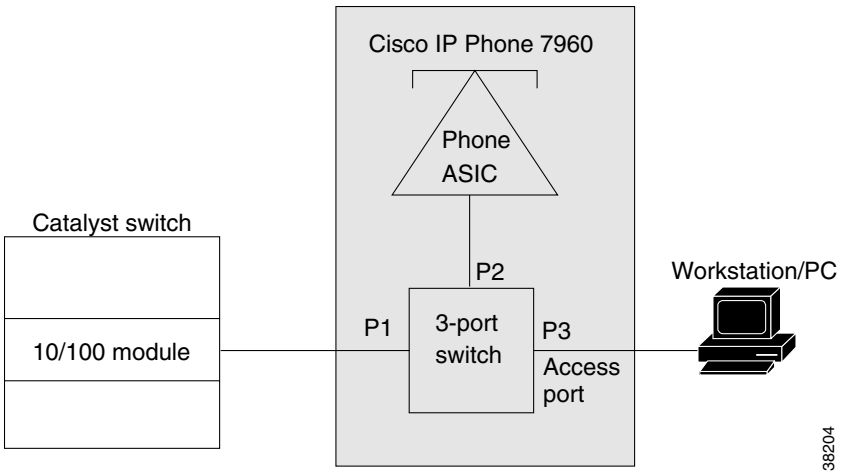
This section describes the native VLANs and the auxiliary VLANs. This section uses the following terminology:

- 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 the VLANs, see Chapter 11, "Configuring VLANs."

Figure 55-3 shows how to connect a Cisco IP Phone 7960 to a Catalyst 6500 series switch.

Figure 55-3 Switch-to-Phone Connections



When the IP phone connects to a 10/100 port on the Catalyst 6500 series switch, the *access port*

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-

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 that are 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.

Understanding How CDP and VoIP Work

Cisco Discovery Protocol (CDP) was enhanced in software release 8.1(1) to facilitate backward compatibility with the newer, higher-powered Cisco IP phones. With this enhanced CDP, a Cisco IP phone can negotiate its power requirements to the switch within the CDP packet. The switch uses this information to ensure that it does not oversubscribe the available power.

We recommend that you enable CDP on the switch so that the switch can correctly detect and supply power to the IP phones that are connected to it. CDP is enabled on the Catalyst 6500 series switches by default; however, you should confirm that CDP is enabled when setting up your VoIP network. For more information on CDP, see Chapter 31, "Configuring CDP."

Configuring VoIP on a 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:

Note For information on using automatic voice configuration, see the "Using SmartPorts" section on page 55-39.

Note You must enable 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 55-3 lists the CLI commands that are described in the configuration procedures.

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	[4]		
set inlinepower defaultallocation			
show port inlinepower			

show environment power			
Voice-related commands			
set port auxiliaryvlan			
show port auxiliaryvlan			
set port voice interface			
show port voice interface			
show port voice			
show port voice fdl			
show port voice active			
QoS commands related to voice			
set port qos *mod/port* **cos-ext** **set port qos** *mod/port* **trust-ext**			
show port qos			

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 4500 series, 5000 family, and 6500 series switches. All modules that are listed in Table 55-3 are supported only on Catalyst 6500 series switches.

Configuring Per-Port Power Management

Note

Note

Catalyst Family Inline-Power Patch Panel Installation Note

This section describes the following topics:

Using show Commands to Display Module Type and Version Information

To determine if the module has a voice daughter card installed, enter the command and look at the “Sub” field. For example, in the following display, the 10/100BASE-TX module in slot 3 has a voice daughter card.

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

	[ ]

This example shows a submodule field that provides information about the submodules. The inline power daughter card that is installed on module 3, as shown in the display, is WS-F6K-SVDB-FE, and the inline power daughter card that is installed on module 6, as shown in the display, is WS-F6K-VPWR-GE-TX.

```
Console> (enable) show module
Mod Slot Ports Module-Type                Model               Sub Status
--- ---- ----- ------------------------- ------------------- --- --------
1   1    2     1000BaseX Supervisor      WS-X6K-SUP2-2GE     yes ok
3   3    48    10/100BaseTX Ethernet     WS-X6548-RJ-45      yes ok
4   4    48    10/100BaseTX Ethernet     WS-X6148-RJ45V      no  ok
6   6    48    10/100/1000BaseT Ethernet WS-X6148-GE-TX      yes ok

Mod Module-Name          Serial-Num
--- -------------------- -----------
1                        SAD04460M9G
3                        SAD0447099V
4                        SAD061901FL
6                        SAD0706025A

Mod MAC-Address(es)                         Hw     Fw          Sw
--- --------------------------------------- ------ ----------- -----------------
1   00-d0-c0-d4-04-4e to 00-d0-c0-d4-04-4f 1.1    6.1(2)      7.7(0.82-Eng)
    00-d0-c0-d4-04-4c to 00-d0-c0-d4-04-4d
    00-02-4a-30-88-00 to 00-02-4a-30-8b-ff
3   00-02-b9-ff-eb-70 to 00-02-b9-ff-eb-9f 0.203  6.3(1)      8.2(1)
4   00-00-00-00-00-00 to 00-00-00-00-00-2f 1.3    5.4(2)      7.7(0.81)
6   00-40-0b-ff-00-00 to 00-40-0b-ff-00-2f 0.304  7.2(1)      8.2(1)
```

```
1   L3 Switching Engine II  WS-F6K-PFC2         SAD044302EA 1.0
3   IEEE InlinePower Module WS-F6K-FE48-AF      sasdfasdf   0.1    8.1(0)
6   Inline Power Module     WS-F6K-VPWR-GE      SAD070700GV 0.201  8.1(0)
Console> (enable)
```

	mod

```
Console> (enable) show version 6
Mod Port Model               Serial #    Versions
--- ---- ------------------- ----------- -------------------------------------
6   48   WS-X6148-GE-TX      SAD0706025A Hw :0.304
                                         Fw :7.2(1)
                                         Sw :8.1(0)
         WS-F6K-VPWR-GE      SAD070700GV Hw :0.201
                                         Sw :8.1(0)
Console>
```

Power Management Modes

{{ | | } [ ] | }.

- —Discovery is enabled and the supervisor engine directs the switching module to power up the port if the switching module discovers the phone. You can specify the maximum wattage that is allowed on the port. If you do not specify a wattage, then the switch will deliver no more than the hardware-supported maximum value.
- —Discovery is enabled and the supervisor engine directs the switching module to power up the port to the wattage that you specify if the switching module discovers the phone. You can specify the maximum wattage that is allowed on the port. If you do not specify a wattage, then the switch allows the hardware-supported maximum value. The maximum wattage, whether determined by the switch or specified by you, is preallocated to the port. If the switch does not have enough power for the allocation, the command will fail.
- —Discovery is disabled which prevents the port from providing power to an external device. If the external device is wall-powered and the inline power is off, the port should still link up, join the bridge group, and go to the STP forwarding state.
- —Discovery is enabled. This mode provides you with the option to limit the power allocated for an external device. If the wattage value that you specify with the keyword is less than the power determined through IEEE classification, instead of denying power, the minimum of these two values is allocated. If the device consumes more than the configured value, the port is shut down and an appropriate syslog message is displayed. The keyword is not supported on all modules. To check if the keyword is supported on a module, enter the command. If the output of the command indicates support for per-port power monitoring, the mode is supported.
- *max-wattage*

-
-
-
-
-
-
-
-
-
-

Power Requirements

Power Requirements for IP Phones

Phone Class	Required Power (W)
Cisco + IEEE	7
Cisco High Power	15.4
Class 0 IEEE	15.4
Class 1 IEEE	4
Class 2 IEEE	7.0
Class 3	15.4
Class 4 Refer to Class 0	Reserved

Available Power

Efficiency of Voice Daughter Cards

Daughter Card	Maximum Power Per Port (W)	Efficiency
		100%
WS-F6K-VPWR-GE	6.3	89%
WS-F6K-GE48-AF	15	89%
WS-F6K-FE48-AF	15	89%
WS-F6K-FE96-AF	15	89%

card with 89 percent efficiency must be 6.3/(0.89) = 7 W. If you are using a voice daughter card with 100 percent efficiency, then the allotted power is 6.3 W.

Wall-Powered Phones

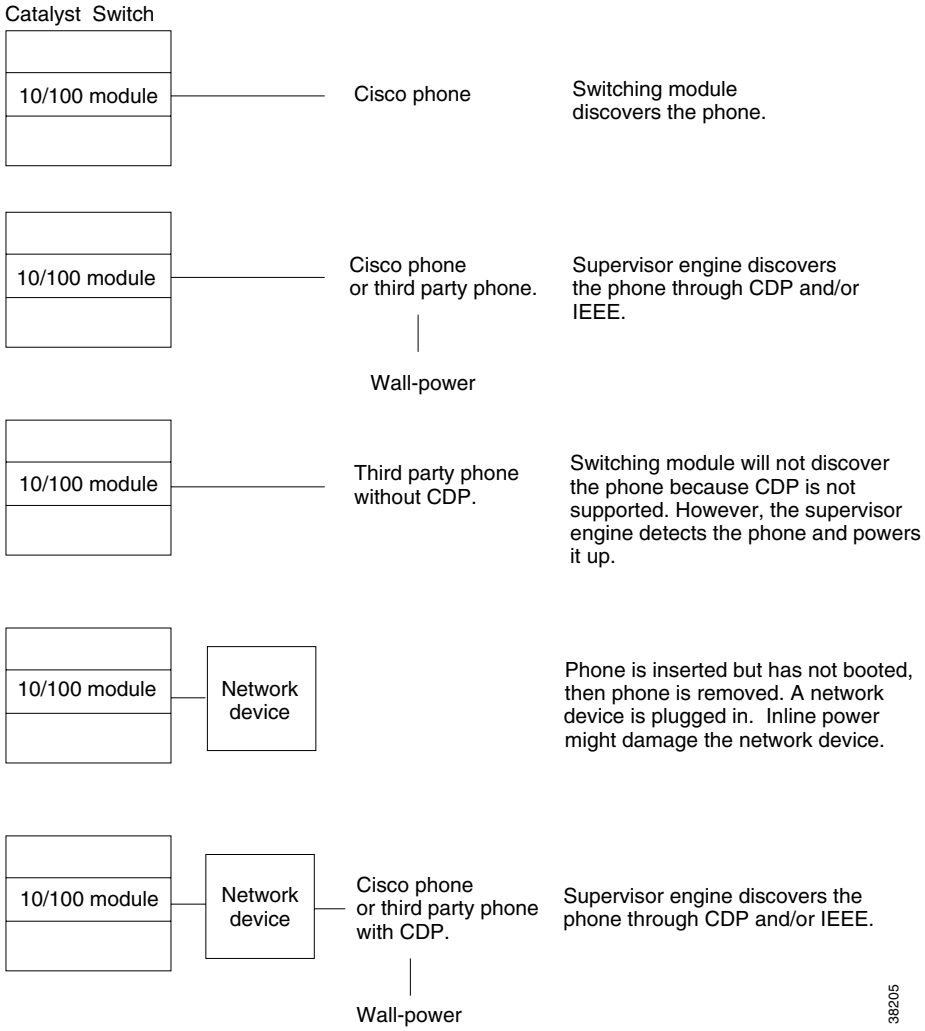
Powering Off the Phone

Phone Removal

Caution

High-Availability Support

Phone Detection Summary



max-wattage

mod/port *max-wattage*

set port inlinepower 2/5 off

set port inlinepower 2/3-9 auto 800

```
Inline power for ports 2/3-9 set to auto and max-wattage to 800.
Console> (enable)
```


```
Console> (enable) set inlinepower defaultallocation 9500
```

Setting the Inline Power Notification Threshold for a Module

	mod_num

```
                    set inlinepower notify-threshold 50 mod 4
Module 4 inlinepower notify-threshold is set to 50%.
Console> (enable)
```

	mod /port

```
Console> show port inlinepower 6/1

Amps @42V)
Total inline power drawn by module 4:  33.934 Watts ( 0.807 Amps @42V)

Port  InlinePowered  PowerAllocated   Device     IEEE class
                     From PS   To PD
      Admin  Oper    mWatts    mWatts
----- ------ ------ -------   ------- ---------- ----------
 6/1  auto   on     7079      6300    cisco      none

Port  MaximumPower  ActualConsumption
      mWatts        mWatts
----- ------------  -----------------
 6/1  15400         6300

Console>
```

```
Console>
Configured Default Inline Power allocation per port: 15.400 Watts (0.36
Amps @42V)
Total inline power drawn by module 4:  33.934 Watts ( 0.807 Amps @42V)

Port      InlinePowered      PowerAllocated  Device     IEEE class DiscoverMode
                             From PS To PD
      Admin  Oper   Detected mWatts  mWatts
----- ------ ------ -------- ------- ------- ---------- ---------- ------------
 4/1  auto   on     yes      7079    6300    cisco      none        cisco

Port  MaximumPower  ActualConsumption  absentCounter  OverCurrent
      mWatts        mWatts
----- ------------  -----------------  -------------  -----------
 4/1  15400         6300               0              0
Console>
```

Displaying the Switch Power Environment for Modules


```
Slot power Requirement/Usage :

Slot Card Type           PowerRequested PowerAllocated CardStatus
                         Watts   A @42V Watts   A @42V
---- ------------------- ------- ------ ------- ------ ----------
1    WS-X6K-SUP2-2GE     128.52   3.06  128.52   3.06  ok
2                          0.00   0.00  128.52   3.06  none
3    WS-X6548-RJ-45      123.06   2.93  123.06   2.93  ok
4    WS-X6148-RJ45V      100.38   2.39  100.38   2.39  ok
6    WS-X6148-GE-TX      145.74   3.47  145.74   3.47  ok
```

```
Slot Inline Power Requirement/Usage :

Slot CardType            Total Allocated   Max H/W Supported  Max H/W
Supported
                         To Module (Watts) Per Module (Watts) Per Port (Watts)
---- ------------------- ----------------- ------------------ ----------------
3    WS-X6548-RJ-45      31.08             315.84             15.400
6    WS-X6148-GE-TX      26.46             315.84             7.000
Console> (enable)
```

Configuring the Auxiliary VLANs on Catalyst LAN Switches

-
-
-
-
-

Understanding the Auxiliary VLANs

-

Note

- **dot1p**
- **set port auxiliaryvlan untagged**

set dot1q-all-tagged

set dot1q-all-tagged

show

trunk **set dot1q-all-tagged**

set port auxiliaryvlan 2/1-3 222

```
222           active        1/2,2/1-3
Console> (enable) set port auxiliaryvlan 5/7 untagged
```

set port auxiliaryvlan 5/9 dot1p

set port auxiliaryvlan 5/12 none

Keyword Descriptions

Keyword	Action

Task	Command

```
show port auxiliaryvlan 123
```

Disabling the Auxiliary VLANs Until an IP Phone is Detected

Note

displayed: “cdpverify feature on port <mod>/<port> is disabled.”

To enable or disable the auxiliary VLAN IP phone detection, perform this task in privileged mode (the default is disabled):

Task	Command
	enable disable

```
set port auxiliaryvlan 3/1 50 cdpverify enable
```

```
show config
```

```
Use 'show config all' to show both default and non-default configurations.
.
.
.
!
#module 3 : 48-port 10/100BaseTX Ethernet
set port auxiliaryvlan 3/1 50 cdpverify enable
!
Console> (enable)
```


```
Console> (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)
```

Task	Command

```
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 the FDL Statistics

Note

Task	Command

Field	Description

Displaying the Port Configuration for the Individual Ports

Task	Command

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-
-

8-Port T1/E1 PSTN Interface Module

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

24-Port FXS Analog Interface Module

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

Displaying the Active Call Information

Task	Command

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Configuring QoS in the Cisco IP Phone 7960

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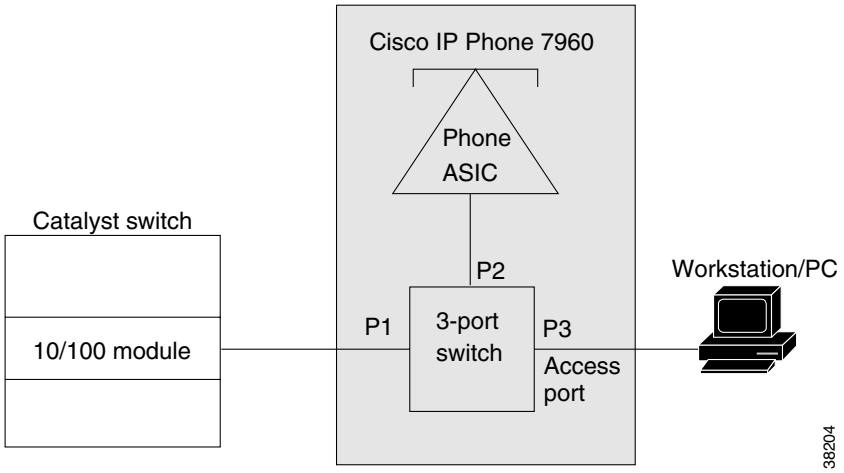
Note

Note

Understanding How QoS Works in the Cisco IP Phone 7960

Note

Configuring QoS on the IP Phone Ports



Configuring QoS in the Cisco IP Phone 7960

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Setting the Phone Access Port Trust Mode


```
set port qos 3/7 trust-ext trusted
```

```
set port qos 3/7 trust-ext untrusted
```

Setting the Phone Access Port CoS Value

Task	Command

Verifying the Phone Access Port QoS Configuration

Task	Command

```
<...Output Truncated...>
Port  Ext-Trust Ext-Cos
----- --------- -------
 3/4  untrusted       0
<...Output Truncated...>
```

Configuring a Trusted Boundary to Ensure Port Security

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Supported Cisco IP Phones

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QoS and Cisco IP Phone Configuration

QoS, Cisco IP Phone, and PC Configuration

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Trusted Boundary Configuration Guidelines

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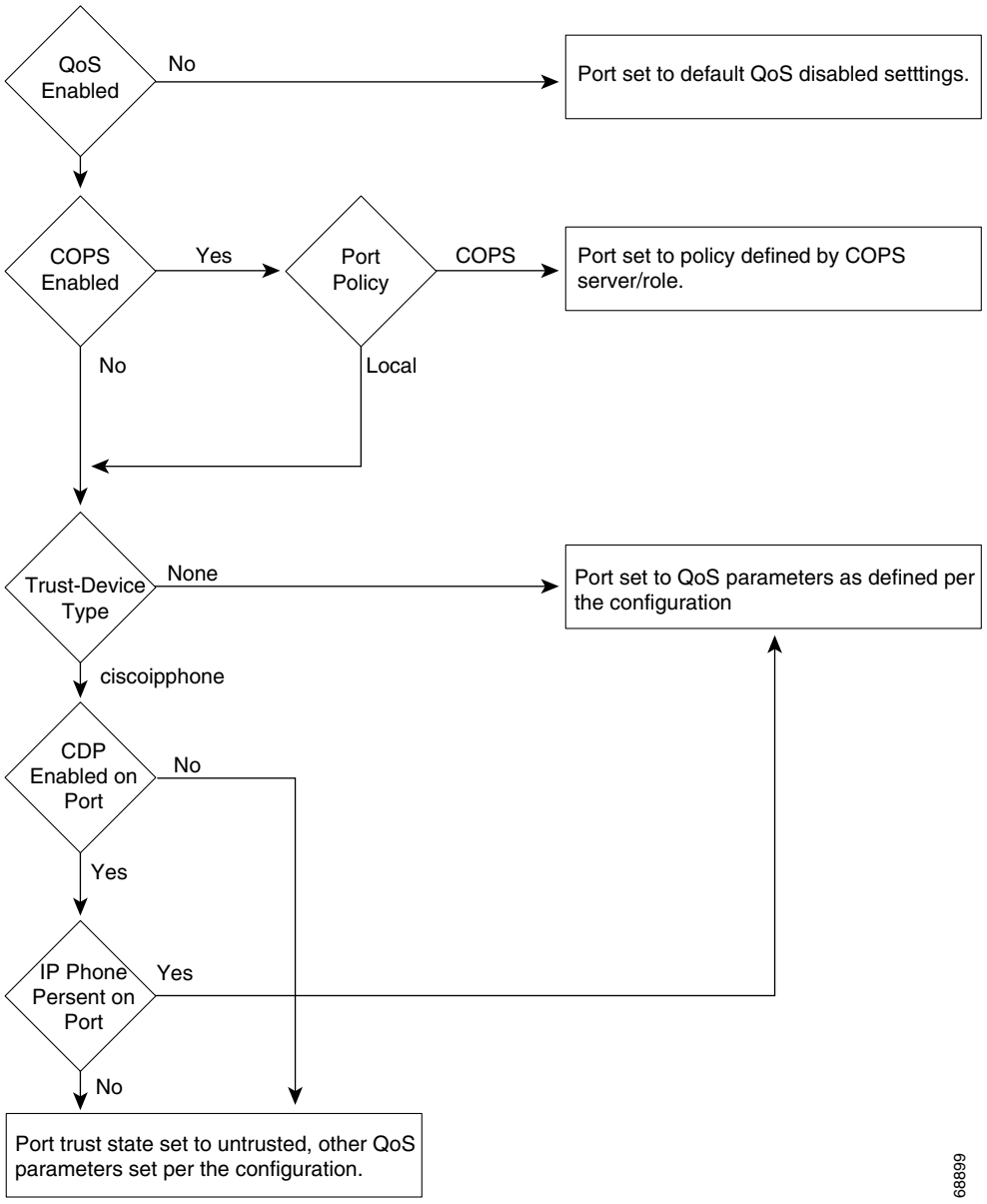
-
  level qos 5
-

ciscoipphone



Configuring a Trusted Boundary

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Default Configuration

Specifying a Cisco IP Phone as the Trust Device

Task	Command

Verifying a Port's Trust-Device State

Task	Command

Note

Using SmartPorts

Understanding SmartPorts Macros

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SmartPorts—Cisco IP Phone

SmartPorts—Cisco Softphone

SmartPorts Guidelines and Restrictions

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Supported Phones

CDP Dependencies

EtherChannel Considerations

PFC/PFC2 Support

Module Support

CLI Interface for SmartPorts

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Command Description

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

ciscoipphone Command Output

```
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)



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




Console> (enable)
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)
```

mod/port *vlan* *auxvlan*

mod/port *vlan*

How to Use SmartPorts in Your Network

Using Automatic Voice Configuration Keywords

	Ports that connect a PC running Cisco SoftPhone without a Cisco IP Phone 79xx.

For cases where the 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.

SmartPorts Enhancements in Software Release 8.4(1)

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Ciscorouter SmartPorts Template

Note

nativevlan *vlans*

mod/port *nativevlan* *vlans*

mod/port **ciscodesktop vlan** *vlan*

nativevlan)

nativevlan)

```
set spantree mode rapid-pvst+
set spantree macreduction enable
set spantree portfast bpdu-guard enable
set spantree global-default loop-guard enable
set qos autoqos
```

Configuring User-Definable SmartPorts Macros

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Overview

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ACLs). The variables are defined as “'keyword-value” pairs, where the first parameter must be the name of the variable and the second parameter is its value. Each variable can be defined on a per-port or global basis. The variables are created using the command. The variables and their values are stored in the switch in a table/database. When a macro with a variable in its definition is applied to a port, the macro takes the values from the table/database and executes the commands in the macro.

Displaying macros and variable definitions—To display macros and their variable definitions, enter the command and the [ ] [ ] [ ] command.

Applying a macro—After you create a macro, it needs to be applied to a port. When the macro is applied to a port, if the macro contains any variables, the variables are replaced with the respective values that are predefined in the table/database, and then the commands in the macro definition are executed. To apply a macro to a port, enter the command.

Clearing (deleting) a macro—You can clear a macro when it is no longer needed. When you clear a macro, only the macro and its definition are cleared from the system; the configuration on the ports that the macro was applied to is not cleared. To clear a macro, enter the command.

Types of macros—The two types of macros are the global macros and the port-based macros.

Using the CLI to Configure User-Definable SmartPorts Macros

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Creating User-Defined Macros

@

```
Enter macro commands one per line. End with character '@'.
set port enable #MODPORT
set vlan $DATAVLAN #MODPORT
set port auxiliaryvlan #MODPORT $AUXVLAN
set qos autoqos
@
```

In the above example, #MODPORT is a variable that specifies the port to which the macro is applied. If the macro is applied on port 3/2, then #MODPORT is replaced by 3/2 when the macro is applied to a port.

In the above example, $DATAVLAN and $AUXVLAN are variables and are substituted with appropriate values when the macro is applied to a port.

After the macro is defined, it is stored in NVRAM.

Modifying Existing User-Defined Macros

```
Warning: The macro fileserver has been modified; Do you want to modify (y/n)
Console> (enable)
```

```
Console> (enable)

Variable DATAVLAN successfully created
Console> (enable)
Console> (enable)

Variable AUXVLAN successfully created
Console> (enable)
```

```
Console> (enable) set macro variable $CDPVER v2
```

```
set macro variable $DATAVLAN 77
```

set macro name videophone

```
set vlan $DATAVLAN #MODPORT
set port auxiliaryvlan #MODPORT $AUXVLAN
@
Console> (enable)
```

Step 1

Step 2

Step 3

Step 4

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Displaying Macros

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Displaying Macro Variables

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Clearing Macros and Macro Variables

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Displaying Macro Port Mappings

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Displaying the User-Definable SmartPorts Macro Configuration

Configuring a Macro within a Macro

Note