# Configuring the Catalyst 6000/6500 WS-X6624 FXS Blade with Cisco CallManager 3.x

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#### Introduction

This document explains how to configure the Cisco CallManager server and the Catalyst 6000/6500 WS-X6624 Foreign Exchange Station (FXS) blade.

Most of the configuration parameters are entered on the Cisco CallManager server. The WS-X6624 FXS blade in the Catalyst 6000/6500 Switch receives its configuration from the Cisco CallManager server via TFTP.

After the WS-X6624 FXS blade receives its configuration via TFTP, it uses the Skinny (SCCP) protocol (3.0) / Media Gateway Control Protocol (MGCP) (3.1, 3.2, 3.3) in order to communicate with the Cisco CallManager server to set up and tear down calls. SCCP is a subset of the H.323 protocol.

## **Prerequisites**

## Requirements

There are no specific requirements for this document.

#### **Components Used**

The information in this document is based on these software and hardware versions:

- Cisco MCS7835 that runs Cisco CallManager 3.2
- Catalyst 6000/6500 Switch that runs CatOS 7.4(2)
- WS-X6624 FXS

The information in this document was created from the devices in a specific lab environment. All of the devices used in this

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document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

#### **Conventions**

For more information on document conventions, refer to the Cisco Technical Tips Conventions.

# Configure the IP Settings on the WS-X6624 Blade (Optional)

The WS-X6624 has a single MAC address and a single IP address. The IP address, IP default gateway, and TFTP server address parameters can be configured manually or they can be configured dynamically from a Dynamic Host Control Protocol (DHCP) server. This example uses manually configured (static) parameters.

**Note:** This document can also be used as a guide for systems that use DHCP in order to set IP parameters. However, this document does not explain how to configure a DHCP server.

If you plan to use DHCP, but you are not sure that your WS-X6624 is configured properly, step 2 provides the syntax to enable DHCP.

If you plan to set your IP parameters manually, step 3 provides an example of how to do this.

1. Use the **set port voice interface help** command in order to view the syntax for setting the IP parameters on a port.

The output from the Catalyst 6000/6500 switch shows this.

2. Use the **set port voice interface 7 dhcp enable** command in order to enable DHCP on a port.

The output from the Catalyst 6000/6500 switch shows this.

```
Greece (enable) set port voice interface 7 dhcp enable Port 7 DHCP enabled.

Greece (enable)
```

Refer to Configure Windows 2000 DHCP Server for Cisco CallManager for more information.

3. Use the **set port voice interface** < mod\_num> **dhcp disable** < ip\_address/mask> tftp < tftp-server-ip-address> gateway < gateway-ip-address> command in order to disable DHCP on a port and assign IP parameters manually.

The output from the Catalyst 6000/6500 shows this.

**Note:** In this example the IP address/mask is 10.48.79.205 255.255.25.0. The TFTP server address is 10.48.80.27. The gateway address is 10.48.79.1.

```
Greece (enable) set port voice interface 4 dhcp disable
10.48.79.205 255.255.255.0 tftp 10.48.80.27 gateway 10.48.79.1
Ports 7/1-24 DHCP disabled.
System DNS configurations used.
Greece (enable)
```

**Note:** The WS-X6624 gateway module does not register with the Cisco CallManager until it is configured on the Cisco

CallManager server. The <u>Create the Catalyst 6000 FXS Gateway in Cisco CallManager</u> explains how to add the new gateway.

## Create the Catalyst 6000 FXS Gateway in Cisco CallManager

Use this procedure in order to create the Catalyst 6000/6500 FXS gateway in Cisco CallManager.

1. From the Device menu, choose **Gateway**.



2. Click Add a New Gateway.



3. Select the Gateway type\* as **Cisco Catalyst 6000/6500 24 port FXS Gateway**, the Device Protocol\* as **Analog Access**, and click **Next**.

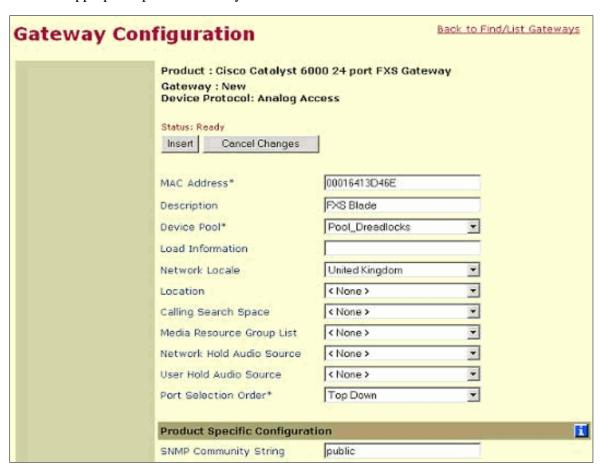


4. Fill in the MAC address of the WS-X6624 blade. This is determined from the Catalyst switch when you issue the **show module** command.

In this example, the **show module 7** command is used and the MAC address is **00-01-64-13-d4-6e**.

```
Greece (enable) show module 7
Mod Slot Ports Module-Type
                             Model
                                           Sub Status
 7 24
                             WS-X6624-FXS
         FXS
                                        no ok
Mod Module-Name Serial-Num
                 SAD043903HW
                              Hw Fw
Mod MAC-Address(es)
                                           Sw
--- ------
                              _____
                              2.0 5.4(2) 7.4(2)
  00-01-64-13-d4-6e
```

5. Enter the appropriate parameters for your environment.



This table explains the parameters used. Use the MAC address shown in the screen in this step.

Field	Description	Usage Notes
MAC Address	Identifies hardware-based devices.	Value must be 12 hexadecimal characters.
Description	Clarifies the purpose of the device.	No usage notes.

Device Pool	Specifies the collection of properties for this device including CallManager Group, Date/Time Group, and Region.	No usage notes.	
Load Information	Specifies the custom software for the gateway.	Values entered here override the default values for this gateway.	
Network Locale	The network locale identifies a set of detailed information to support the hardware in a specific location. It includes a definition of the tones and cadences used by the device in a specific geographic area.	No usage notes.	
Location	Specifies the location this device is to be associated with when using Call Admission Control (CAC).	No usage notes.	
Calling Search Space	Specifies the collection of partitions searched to determine how a collected (originating) number should be routed.	No usage notes.	
Media Resource Group List	This list provides a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music On Hold server, among the available media resources. This is based on the priority order defined in a Media Resource Group List.	No usage notes.	
Network Hold Audio Source	This audio source plays when the network initiates a hold action.	No usage notes.	
User Hold Audio Source	This audio source plays when a user initiates a hold action.	No usage notes.	

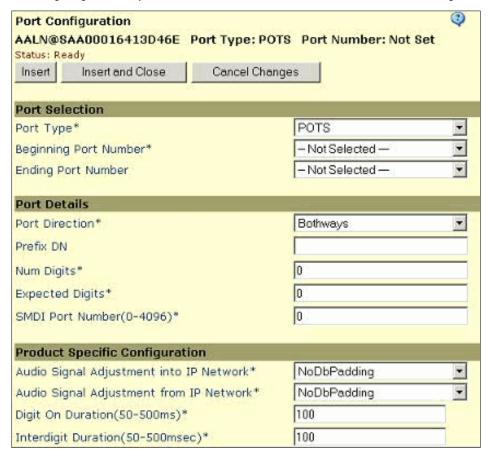
Port Selection Order	Specifies the order in which ports are selected.  TOP_DOWN selects ports in descending order, from port 1 to port 24.  BOTTOM_UP selects ports in ascending order, from port 24 to port 1.	Valid entries are TOP_DOWN or BOTTOM_UP. If you are not sure which port order to use, choose TOP_DOWN.
-------------------------	--	--

6. Click Add a New Port.

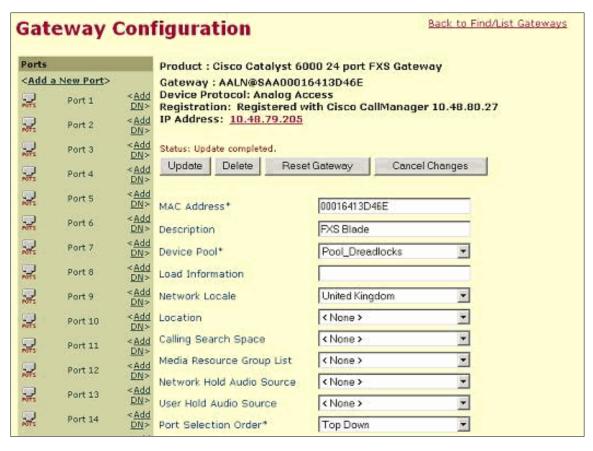


7. For this example the **All Ports** option is used under Beginning Port Number. Click **Insert and Close**.

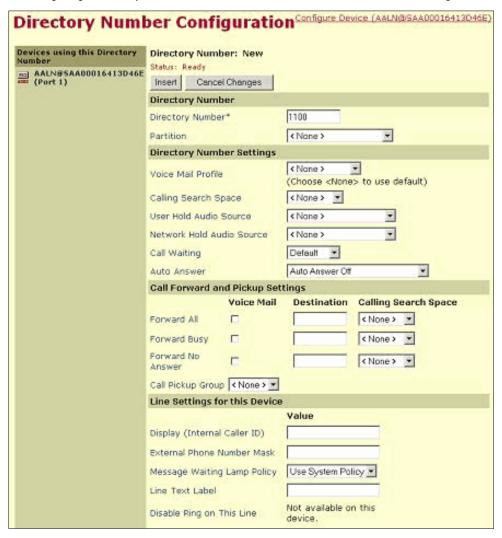
Note: The only port type supported when this document was written was Plain Old Telephone Service (POTS) (FXS).



8. Configure the Directory Number (DN) for each of the ports seen in this screen. Click **Add DN** for one of the ports shown on the left.



9. Enter your required parameters in this screen.



This table explains the parameters for this screen.

Field	Description	Usage Notes				
Directory Number						
Directory Number	Indicates a dialable phone number.  If the words 'Shared Line' appear in red next to the DN, the DN appears on more than one device in the same partition.	<ul> <li>Values can include a maximum of 50 numeric characters except for (.) and (@).</li> <li>Can appear in more than one partition as long as the DN/Partition pair forms a unique combination.</li> </ul>				
Partition	Indicates the partition to which the DN belongs.	Unique in combination with the DN.				
,	Directory Number	Settings				

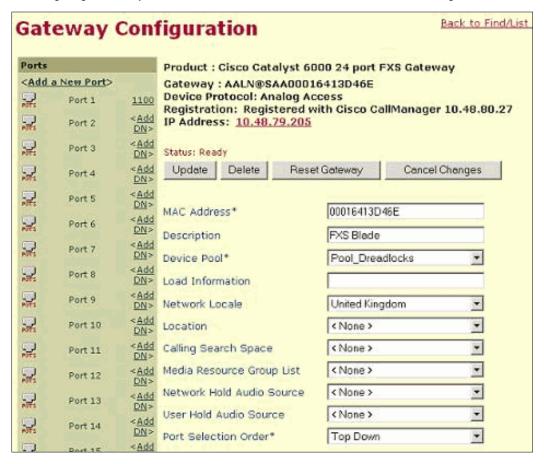
Voice Mail Profile	List of Voice Mail Profiles defined in the Voice Mail Profile Configuration.	The first option is <none>, which is the current default Voice Mail Profile configured in the Voice Mail Profile Configuration.</none>	
Calling Search Space	Collection of partitions that are searched for numbers called from this DN.	Applies to all devices that use this DN.	
User Hold Audio Source	The audio source played when a user initiates a hold action.	No usage notes.	
Network Hold Audio Source	This audio source plays when the network initiates a hold action.	No usage notes.	
Call Waiting	Specifies whether this DN uses Call Waiting when a line is busy (On), responds with a busy signal (Off), or uses the system-wide default setting (Default).	Applies to all devices that use this DN.	
Auto Answer	Activates the Auto-Answer feature for this DN.	O Auto Answer Off <default> O Auto Answer with headset O Auto Answer with speakerphone (Intercom)  Make sure the headset or speakerphone is not disabled when you select Auto Answer with headset or Auto Answer with speakerphone.</default>	
	Call Forward and Pic	kup Settings	
Call Pickup Group	Indicates a number that can be dialed to answer calls to this DN (in the specified partition).	No usage notes.	

Forward All	Indicates the DN to which all calls are forwarded.	<ul> <li>Any dialable phone number. This includes an outside destination.</li> <li>Applies to all devices that use this DN.</li> </ul>
Forward Busy	Indicates the DN that a call is forwarded to when the line is in use.	<ul> <li>Any dialable phone number.         This includes an outside destination.     </li> <li>Applies to all devices that use this DN.</li> </ul>
Forward No Answer	Indicates the DN that a call is forwarded to when no one answers after four rings.	<ul> <li>Any dialable phone number, includes an outside destination.</li> <li>Applies to all devices that use this DN.</li> </ul>
Calling Search Space	Indicates the Calling Search Space to use when forwarding to the specified destination.  Calling Search Space can be configured for Forward All, Forward Busy, and Forward No Answer DNs.	Applies to all devices that use this DN.
	Line Settings for th	is Phone
Display	Indicates text that appears on the phone of the called party when a call is placed from this line.	<ul> <li>Leave this field blank to have the system display the extension.</li> <li>Maximum of 30 alphanumeric characters.</li> <li>Typically use the name of the user or the DN.</li> <li>Applies only to</li> </ul>

		the current device.
External Phone Number Mask	Indicates the phone number (or mask) used to send caller ID information when you place a call from this line.	Maximum of 30 number and "X" characters. The X characters must appear at the end of the pattern.
	Configures the handset lamp illumination policy. Choose one of these options:  O Use System	
Message Waiting Lamp Policy	Policy. The DN refers to the service parameter "Message Waiting Lamp Policy" setting.	Applies to the current device.
	<ul><li>Always Light</li><li>Never Light</li></ul>	
	Indicates the text for the line button on this phone.	
Line Text Label	O Cisco IP Phone 7960/7940 - The text is displayed on the LCD.  Applies only to the current device.	
Label	Other Cisco IP Phones - The text is not displayed but could be used when you print button templates.	
Disable ring on this line	Stops the phone from ringing to indicate incoming calls.	Applies only to the current device.

- 10. Click **Insert** and repeat steps **9** and **10** for all of the ports that you use.
- 11. When you are finished, return to the **Gateway Configuration** menu.

**Note:** Only one DN is configured in this example.

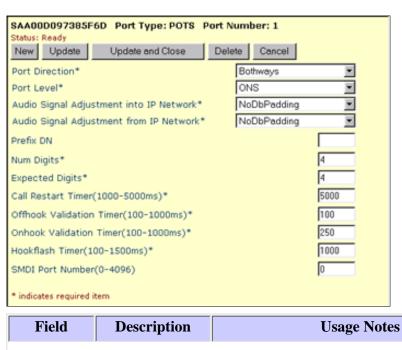


12. In order to finish the configuration of each port, at a minimum, configure the **Num Digits** and the **Expected Digits** parameters for each port that you use. Select the icon labeled with POTS for one of the ports that you assigned a DN to.

Enter your required parameters in this screen.

Click **Update and Close** when you are done.

Enter your required parameters in this screen. The table that follows this screen explains the different fields within this screen.

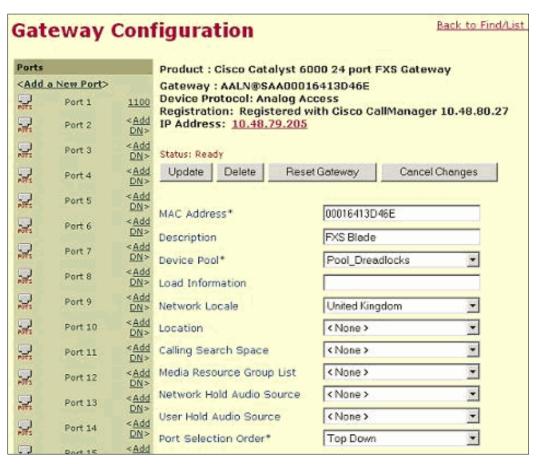


Port Direction	The direction of calls that pass through this port:  O Inbound - Use for incoming calls only.  O Outbound - Use for outgoing calls.  O Bothways - Inbound and outbound calls (default).	No usage notes.		
Port Level	Adjusts the gain of audio that enters or leaves the span.	Do not modify this parameter.		
Audio Signal Adjustment into IP Network	Specifies the gain or loss applied to the received audio signal relative to the port application type.	Select the gain or loss you want applied to the received audio signal relative to these port application types:  AnalogCOTrunk=Minus3db DigitalToAnalogCO=NoDbPadding AnalogTieTrunk=NoDbPadding DigitalToDigitalCO=NoDbPadding ISDNStation=NoDbPadding ISDN_DigitalTieTrunk=NoDbPadding ISDNTrunk=NoDbPadding ISDNTrunk=NoDbPadding OnPremisePOTSLine=Plus3db OffPremisePOTSLine=NoDbPadding SatelliteAnalogTieTrunk=NoDbPadding SatelliteDigitalTieTrunk=NoDbPadding AnalogTollTrunk=Plus3db Do not modify these parameters unless instructed by Cisco.		

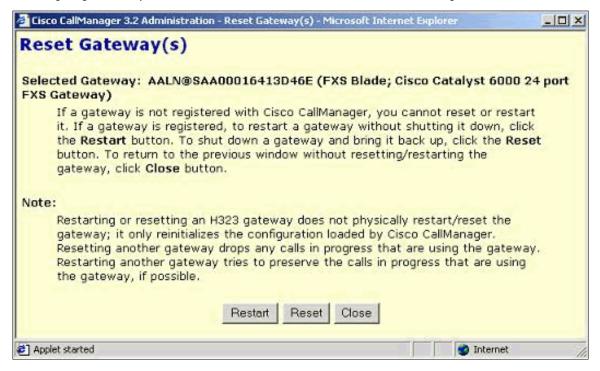
Specifies the gain or loss applied to the transmitted audio signal relative to the port application type.  Specifies the prefix digits that	<ul> <li>AnalogCOTrunk=Minus6db</li> <li>DigitalToAnalogCO=Minus3db</li> <li>AnalogTieTrunk=NoDbPadding</li> <li>DigitalToDigitalCO=NoDbPadding</li> <li>ISDNStation=NoDbPadding</li> <li>ISDN_DigitalTieTrunk=NoDbPadding</li> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless instructed by Cisco.</li> </ul>		
gain or loss applied to the transmitted audio signal relative to the port application type.  Specifies the	<ul> <li>AnalogTieTrunk=NoDbPadding</li> <li>DigitalToDigitalCO=NoDbPadding</li> <li>ISDNStation=NoDbPadding</li> <li>ISDN_DigitalTieTrunk=NoDbPadding</li> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
gain or loss applied to the transmitted audio signal relative to the port application type.  Specifies the	<ul> <li>DigitalToDigitalCO=NoDbPadding</li> <li>ISDNStation=NoDbPadding</li> <li>ISDN_DigitalTieTrunk=NoDbPadding</li> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
gain or loss applied to the transmitted audio signal relative to the port application type.  Specifies the	<ul> <li>ISDNStation=NoDbPadding</li> <li>ISDN_DigitalTieTrunk=NoDbPadding</li> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
applied to the transmitted audio signal relative to the port application type.  Specifies the	<ul> <li>ISDN_DigitalTieTrunk=NoDbPadding</li> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
transmitted audio signal relative to the port application type.	<ul> <li>ISDNTrunk=NoDbPadding</li> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
relative to the port application type.  Specifies the	<ul> <li>OnPremisePOTSLine=Plus3db</li> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
Specifies the	<ul> <li>OffPremisePOTSLine=Minus3db</li> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
Specifies the	<ul> <li>SatelliteAnalogTieTrunk=Minus3db</li> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
•	<ul> <li>SatelliteDigitalTieTrunk=Minus3db</li> <li>AnalogTollTrunk=NoDbPadding</li> <li>Do not modify these parameters unless</li> </ul>		
•	AnalogTollTrunk=NoDbPadding  Do not modify these parameters unless		
•	Do not modify these parameters unless		
•	_ =		
•			
are appended to the digits this trunk receives on incoming calls.	The Cisco CallManager adds prefix digits after truncating the number in accordance with the Num Digits setting.		
Specifies the number of significant digits to collect, from 0 to 32.  Significant digits are counted from the right (last digit) of the number called.	This field is used to process incoming calls and indicates the called number used to route call.		
Number of digits expected on the inbound side of the trunk.	Leave zero as the default value if you are unsure. This field is rarely used.		
The default is 1500 ms.	Cisco recommends you always use default values for this timer.		
to Sath de la constant de la constan	ignificant digits re counted from the right (last ligit) of the light called.  Jumber of digits expected on the light has a specific digit.  The default is		

Offhook Validation Timer (100 -1000 ms)	The default is 100 ms.	Cisco recommends you always use default values for this timer.
Onhook Validation Timer (100 -1000 ms)	The default is 250 ms.	Cisco recommends you always use default values for this timer.
Hookflash Timer (100 - 1500 ms)	The default is 1000 ms.	Cisco recommends you always use default values for this timer.
SMDI Port number	The default is 0.	Use this field for analog access ports that connect to a voice-mail system

13. From the Gateway Configuration screen, click **Reset Gateway**.



14. Click Reset.



15. You have now completed the basic steps required in order to add and configure this gateway. After a couple of minutes the WS-X6624 module finishes its registration process with the Cisco CallManager server. Use the **show port** command on the switch in order to verify that the registration process is successful.

Greece (	enable)	show port	7				
Port Na	me		Status		Duple	x Speed	Туре
7/1				 79	ful	1 64k	FXS
7/2			onhook	79	ful	1 64k	FXS
7/3			onhook	79	ful	1 64k	FXS
7/4			onhook		ful	1 64k	FXS
7/5			onhook		ful	1 64k	FXS
7/6			onhook	79	ful	1 64k	FXS
7/7			onhook		ful	1 64k	FXS
7/8			onhook	79	ful	1 64k	FXS
7/9			onhook	79	ful	1 64k	FXS
7/10			onhook		ful	1 64k	FXS
7/11			onhook		ful	1 64k	FXS
7/12			onhook	79	ful	1 64k	FXS
7/13			onhook		ful	1 64k	FXS
7/14			onhook	79	ful	1 64k	FXS
7/15			onhook	79	ful	1 64k	FXS
7/16			onhook	79	ful	1 64k	FXS
7/17			onhook		ful	1 64k	FXS
7/18			onhook	79	ful	1 64k	FXS
7/19			onhook		ful	1 64k	FXS
7/20			onhook	79	ful	1 64k	FXS
7/21			onhook	79	ful	1 64k	FXS
7/22			onhook		ful	1 64k	FXS
7/23			onhook	79	ful	1 64k	FXS
7/24			onhook	79	ful	1 64k	FXS
Port	DHCP	MAC-Addre		IP-Addres	s Sub	net-Mas	k
7/1-24	disable	00-01-64-	-13-d4-6e	10.48.79.	205 255	.255.25	5.0
Port	Call-Ma	nager(s)	DHCP-Sei	rver T	FTP-Server	. Ga	teway

7/1-24 (*): Pri	10.48.80.27* 10.48.80.34	-	10.48.80.27	10.48.79.1
( )	mar y			
Port	DNS-Server(s)	Domain		
7/1-24	10.48.80.44	-		
Port	CallManagerState	DSP-Type		
7/1-24	registered	C549		

# Verify the Catalyst/CallManager Configuration

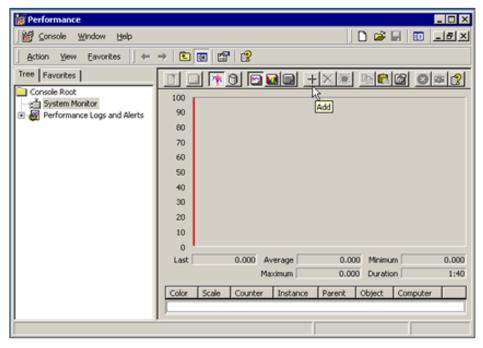
This section provides information you can use in order to confirm that your configuration works properly.

Certain **show** commands are supported by the <u>Output Interpreter Tool</u> (<u>registered</u> customers only), which allows you to view an analysis of **show** command output.

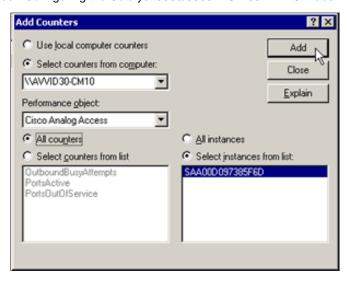
## Use Performance Monitor to Analyze WS-X6624 Calls and Status Changes

Use this procedure in order to analyze WS-X6624 calls and status changes.

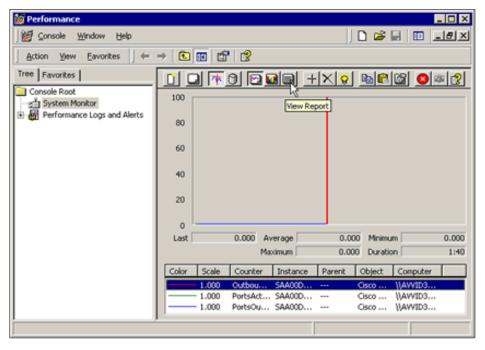
1. Select **Start** > **Programs** > **Administrative Tools** > **Performance option** and click the **Add** (+) button to start Performance Monitor.



2. From the Add Counters screen, select **Cisco Analog Access** as the Performance object, select the **All Counters** option, select the gateway (in this case **SAA00D097385F6D**), and then click **Add** and **Close**.

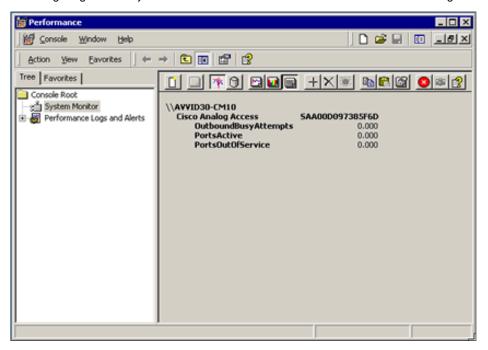


3. A window similar to this appears.



4. Select the **View Report** button.

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5. Try to make inbound and outbound calls via the gateway. You see changes in this screen that reflect the calls you make.

## **Troubleshoot**

This section provides information you can use to troubleshoot your configuration.

If you do not see the instance of the gateway you created, it is possible it has not registered with the Cisco CallManager server.

The most common problem is that the MAC address of the port is entered incorrectly in the Cisco CallManager server configuration. Verify that you have entered the correct MAC address before you proceed with troubleshooting.

If you continue to have problems, try to reset the module from the Catalyst switch using the **reset** <**mod\_num**> command. Wait until the registration process is completed. Check this by the use of the **show port** <**mod\_num/port\_num>** command and look for the IP address of the Cisco CallManager server.

If these suggestions do not resolve the problem, continue with these suggestions:

Make sure that the port has the correct IP addresses configured. At a minimum the port needs its own IP address and mask and the IP address of the TFTP (CallManager) server. If the IP address of the port is on a different subnet (VLAN) it also requires a gateway address. Finally, if your network relies on DNS, the port needs its DNS server address and domain name configured. If you use DHCP, refer to <a href="Configure Windows 2000 DHCP Server for Cisco CallManager">CallManager</a> for further information on the configuration and use of DHCP. If you configure the IP parameters manually see the <a href="Configure the IP Settings">Configure the IP Settings on the WS-X6624Blade</a> section of this document.

In order to find the correct TFTP (Cisco CallManager) address, log on to the Cisco CallManager server and check the IP addresses used under the **System** > **Server** menu.

For both DHCP and non-DHCP configurations, verify that the VLAN is correct. It is not possible to set the port's VLAN via DHCP. This must be done at the CLI of the switch. The syntax is **set vlan** <code><vlan\_number><mod\_num/port\_num></code>. Also verify that the port status is not disabled. The syntax for enabling a port is **set port enable** <code><mod\_num/port\_num></code>.

If you use DHCP and/or DNS and you still have problems, try to:

- Manually configure the IP parameters to eliminate DHCP from the equation.
- Use IP addresses instead of DNS hostnames.

If none of these steps resolve the problem, open a case with the support center that you use for Cisco support.

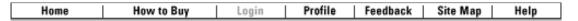
## **NetPro Discussion Forums - Featured Conversations**

Networking Professionals Connection is a forum for networking professionals to share questions, suggestions, and information about networking solutions, products, and technologies. The featured links are some of the most recent conversations available in this technology.

NetPro Discussion Forums - Featured Conversations for Voice
Service Providers: Voice over IP
Voice & Video: Voice over IP
Voice & Video: IP Telephony
Voice & Video: IP Phone Services for End Users
Voice & Video: Unified Communications
Voice & Video: IP Phone Services for Developers
Voice & Video: General

## **Related Information**

- Voice Technology Support
- Voice and IP Communications Product Support
- Recommended Reading: <u>Troubleshooting Cisco IP Telephony</u>
- Technical Support Cisco Systems



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