How to Use Cisco CallManager to Configure a Catalyst WS-X6608-T1 Port as a T1 VoIP Gateway

Contents

Introduction

This document explains how to configure the Cisco CallManager server and the Catalyst 6000 WS-X6608-T1/E1 Blade for Voice over the public switched telephone network (PSTN).

The Catalyst 6000 Family 8-port T1/E1 PSTN Interface Module is a high-density, eight port, T1/E1 Voice over IP (VoIP) module that 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.

How the ports function is dependent on the software you download; the ports can serve as T1/E1 interfaces or the ports can support transcoding and conferencing. Transcoding and conferencing functions are mutually exclusive. For every transcoding port in use, there is one less conferencing port available. Likewise, for every conferencing port in use, there is one less transcoding port available.

Most of the configuration parameters are entered on the Cisco CallManager server. The WS-X6608-T1/E1 blade in the Catalyst 6000 Switch receives its configuration from the Cisco CallManager server using TFTP.

When the WS-X6608-T1/E1 blade is used as a T1 or E1 gateway, it uses the Skinny protocol to communicate with the Cisco CallManager server, to set up and tear down calls. Skinny is a subset of the H.323 protocol.

Note: If you do not configure or disable all of the ports on a WS-X6608 blade, this error message is generated:

```
%SYS-4-MODHPRESET:Host process (860) mod_num/port_num got reset asynchronously
```

Note: This system message continually appears on your console screen and in your syslogs, if you have them configured. This is the expected behavior for this blade. It does not affect system performance.

Symptoms

You can encounter these symptoms when you configure the Catalyst WS-X6608-T1 with CallManager:

- PRI channels appear in a hung state, and calls do not get through even though gateway shows ports as idle. Refer to CSCsb91325 and CSCsa91414.
- PRI ports do not register with CallManager. Ensure that the PRI port is connected to the Telco line and that layer 1 and 2 are up.
- When you use two PRI ports as one trunk group with one D-channel, only 23 channels appear. NFAS, Non-Facility Associated Signaling, is not supported on MGCP.
- When calls are made from an IP phone using G.729 to PSTN and vice versa, a wind blowing sound is heard. To resolve this, in CallManager Service Parameters set **Strip G.729 Annex B (Silence Suppression) from Capabilities** to true.
- When you use PRI protocol DMS-100, no calls can get through. In order to resolve this, ensure you click the check box labelled **MCDN Channel Number Extension Bit Set to Zero** under the Gateway Configuration page.

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:

- Catalyst 6000 Switch / CatOS 6.1(3)
- WS-X6608 Blade
- MCS7835 Cisco CallManager 3.(0)7
- MCS7835 Cisco CallManager 4.0

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

Refer to the Cisco Technical Tips Conventions for more information on document conventions.

Configure the IP Settings on the WS-X6608-T1 Port

In this task the IP parameters of the WS-X6608-T1 blade are configured. This task is not required if your configuration uses a Dynamic Host Configuration Protocol (DHCP) server to provide this information.

Note: This is the default behavior for all ports on a WS-X6608-T1 blade.

Note: If you plan to use DHCP, but you are not sure that your ports are currently configured properly, step 2 provides the syntax necessary to enable DHCP.

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

Step-by-Step Instructions

Complete these steps to configure the IP parameters of the WS-X6608-T1 blade:

1. Issue the **set port voice interface help** command to view the syntax to set the IP parameters on a port.

   Here is a sample output from the Catalyst 6000 Switch:

   ```
   Console> (enable)  set port voice interface help
   Usage:  set port voice interface <mod/port> dhcp enable [vlan <vlan>]
           set port voice interface <mod/port> dhcp disable <ipaddrspec>
                                tftp <ipaddr> [vlan <vlan>]
                                [gateway <ipaddr>] [dns [ipaddr] [domain_name]]
         (ipaddr_spec: <ipaddr> <mask>, or <ipaddr>/<mask>
   ```

```
              <mask>: dotted format (255.255.255.0) or number of bits (0..31)
                vlan:   1..1005,1025..4094
          System DNS may be used if disabling DHCP without DNS parameters)

Console> (enable)
```

2. Issue the **set port voice interface 5/4 dhcp enable** command to enable DHCP on a port.

   This sample output from the Catalyst 6000 Switch shows this:

   ```
   Console> (enable) set port voice interface 5/4 dhcp enable
   Port 5/4 DHCP enabled.

   Console> (enable)
   ```

   Repeat this step for each port that your configuration requires. If you are using DHCP, skip the next step and proceed with Task 2: Create the Catalyst 6000 T1 VoIP Gateway in CallManager 3.x or Task 3: Create the Catalyst 6000 T1 VoIP Gateway in CallManager 4.x.

   For more information, refer to Configuring Windows 2000 DHCP Server for Cisco CallManager.

3. Issue the **set port voice interface 5/1 dhcp disable *<ip_address/mask>* tftp *<tftp-server-ip-address>* gateway *<gateway-ip-address>*** command to disable DHCP on a port and assign IP parameters manually.

   In this example the IP address or mask is *172.16.14.73/27*. The TFTP server (Cisco CallManager server in this case) address is *172.16.14.66*. The gateway address is *172.16.14.65*.

   Here is a sample output from the Catalyst 6000 Switch:

   ```
   AV-6509-1 (enable) set port voice interface 5/1 dhcp disable 172.16.14.70/
   Port 5/1 DHCP disabled.
   System DNS configurations used.
   AV-6509-1 (enable)
   ```

   Repeat this step for each port that your configuration requires.

   Note: You cannot specify more than one port at a time on the WS-X6608-T1 blade because a unique IP address must be set for each port.

   Note: The WS-X6608 port cannot register with Cisco CallManager until it has been configured on the CallManager server. The next steps explain how to add the new gateway.

Create the Catalyst 6000 T1 VoIP Gateway in Cisco CallManager 3.x

This task explains how to configure the T1 gateway port on the Cisco CallManager server.

Step-by-Step Instructions

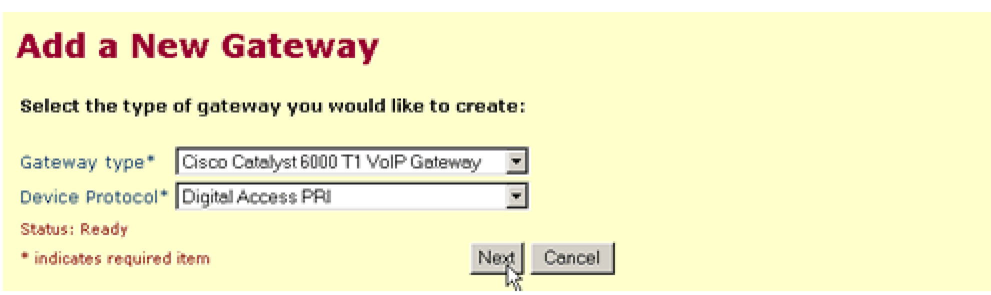
Complete these steps to configure the T1 gateway port:

Note: The E1 configuration is very similar.

1. Select **Gateway** from the **Device** menu. A screen similar to this appears:

2. Click **Add a New Gateway**. The Find and List Gateways screen appears:

3. Select the Gateway Type as **Cisco Catalyst 6000 T1 VoIP Gateway** and Device Protocol as **Digital Access PRI**.



Click **Next**.

4. Fill in the MAC Address of the ports on the WS-X6608-T1 blade.

The MAC Address in this example is from port 5/1 of the WS-6608-T1 blade on the Catalyst 6000 Switch. You can find this information if you issue the **show port** command.

```
AV-6509-1 (enable) sh port 5

(Text Deleted)

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 5/1     disable 00-10-7b-00-10-10 172.16.14.70    255.255.255.224
 5/2     disable 00-10-7b-00-10-11 172.16.14.71    255.255.255.224
 5/3     disable 00-10-7b-00-10-12 172.16.14.73    255.255.255.224
 5/4     enable  00-10-7b-00-10-13 0.0.0.0         0.0.0.0
 5/5     disable 00-10-7b-00-10-14 172.16.14.25    255.255.255.224
 5/6     disable 00-10-7b-00-10-15 172.16.14.26    255.255.255.224
 5/7     disable 00-10-7b-00-10-16 172.16.14.81    255.255.255.224
 5/8     disable 00-10-7b-00-10-17 172.16.14.80    255.255.255.224
```

Access Digital PRI Gateway Configuration Settings

Field	Description	Usage Notes
MAC Address	Identifies hardware-based telephones and device name.	Value must be 12 hexadecimal characters.
Description	Clarifies the purpose of the device.	
Device Pool	Specifies the collection of properties for this device including CallManager Group, Date/Time Group and Region.	
Calling Search Space	Specifies the collection of partitions to be searched to determine how a collected (originating) number should be routed.	
Location	Specifies the location this device is to be associated with when using Call Admission Control.	
Load Information	Specifies the custom software for gateway.	Values entered here override the default values for this gateway.
TX-Level CSU	Specifies the transmit level based on the distance between the gateway and the nearest repeater. The default is full power (0dB).	Select one of the alternative settings to attenuate the line. • -7.5dB • -15dB

		• -22.5dB
Channel Selection Order	Specifies the order in which ports are enabled from first (lowest number port) to last (highest number port), or from last to first.	Valid entries are TOP_DOWN (last to first) or BOTTOM_UP (first to last). If you are not sure which port order to use, choose TOP_DOWN.
PCM Type	Specifies the digital encoding format.	Choose from these options: • **A-law -** Use for Europe and the rest of the world • **µ-law -** Use for North America and Japan
Clock Reference	Specifies from where the master clock is derived. Cisco Catalyst 6000 blades have eight ports on the same hardware card, each of which can be used as a clock reference by other ports on the same blade.	Select Internal or Network. • **Internal -** When clocking is derived from the card and is then distributed at the span • **Network -** When the Cisco Access Digital Trunk Gateway receives its clocking from the network • **Span 1 to Span 8 -** When the Cisco Access Digital Trunk Gateway receives

		clocking from another port on the same Cisco Catalyst 6000 blade.
Protocol Side	This setting is used for Cisco Access Digital gateways depending on whether gateway is connected to a Central Office (CO)/Network device or to a User device.	The two ends of the PRI connection should use opposite settings. For example, if you are connected to a PBX and the PBX uses User as its protocol side, Network should be chosen for this device. Typically, this option is User for central office (CO) connections.
Caller ID Directory Number (DN)	The pattern you want to use for Caller ID, from 0 to 24 digits.	For example, in North America: • 555XXXX = Variable Caller ID, where X is equal to an extension number. The CO appends the number with the area code if you do not specify it. • 5555000 = Fixed Caller ID. Use when you want the Corporate number to be sent instead of the exact extension from which the call is placed. The CO appends the number with the area

		code if you do not specify it.
Calling Party Selection	Determines which dialed number (DN) is sent. Any outbound call on a gateway can send DN information.	These options specify which DN is sent: • **Originator** - Send the DN of the calling device. • **First Redirect Number** - Send the DN of the redirecting device. • **Last Redirect Number** - Send the DN of the last device to redirect the call.
Channel IE Type	Indicates whether channel selection is presented as a channel map or a slot map.	Select one of the following from the drop-down list box: • **Number** - B-channel usage is always a channel map format. • **Slotmap** - B-channel usage is always a slotmap format • **Use Number When 1** - B-Channel usage is a channel map for one B-channel but is a slotmap if more than one B-channel

Interface Identifier Present	The purpose of this parameter is to interoperate with a Nortel PBX when the PBX is configured to use the DMS100 protocol. When this box is checked, it indicates that an Interface Identifier is present.	The default for this setting is disabled. This setting only applies to the DMS100 protocol for digital access gateways in the Channel Identification information element (IE) of the SETUP, CALL PROCEEDING, ALERTING, and CONNECT messages.
Interface Identifier Value	This value should be obtained from the PBX provider.	Applies to DMS100 protocol only. Valid values range from 0 to 255.
Display IE Delivery	When this box is checked, it enables delivery of the display information element (IE) in SETUP and CONNECT messages for the calling and called party name delivery service.	By default, Display IE Delivery is disabled.
Redirecting Number IE Delivery	When this box is checked (enabled), the Redirecting Number IE is included in the SETUP message to indicate the first redirecting number and the redirecting reason of the call when Call Forward happens.	By default, this setting is disabled. This setting applies to the SETUP message only on all protocols for digital access gateways.
Delay for first restart (1/8 sec ticks)	Controls the rate at which the spans are brought in service when Inhibit Restarts at PRI Initialization is disabled.	Use this option when many PRI spans are enabled on a system and Inhibit Restarts at PRI Initialization is disabled. For example, set the first five cards to 0, and set the next five cards to 16. (Wait two seconds before

		you bring them in service.)
Delay between restarts (1/8 sec ticks)	Determines the length of time between restarts if Inhibit Restarts is disabled, when a PRI RESTART is sent.	
Num Digits	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 if you enable Sig Digits. It is used for the processing of incoming calls and indicates the number of digits starting from the last digit of the called number used to route calls coming into the PRI span. See Prefix DN and Sig Digits.
Sig Digits	Represents the number of final digits a PRI span should retain on inbound calls. A trunk with significant digits enabled truncates all but the final few digits of the address provided an inbound call.	Enable or disable this box depending on whether you want to collect significant digits. If disabled, the Cisco CallManager does not truncate the inbound number. If enabled, you also need to choose the number of significant digits to collect.
Prefix DN	Specifies the prefix digits that are appended to the digits this trunk receives on incoming calls.	The Cisco CallManager adds prefix digits after it truncates the number in accordance with the Num Digits setting.
Presentation Bit	Determines whether the central office transmits or blocks caller ID.	Select Allowed if you want the CO to send caller ID. Select Restricted if you do not want the

		CO to send caller ID.
Called party IE number type unknown	The format for the type of number in called party DNs. Cisco CallManager sets the called DN type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as North American Numbering Plan (NANP) or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when connecting to PBXs using routing as a non-national type number.	Use these definitions for each of the variables: • **CallManager** - The Cisco CallManager sets the DN type. • **International** - Use when you dial outside the dialing plan for your country. • **National** - Use when you dial within the dialing plan for your country. • **Unknown** - The dialing plan is unknown.
Calling party IE number type unknown	The format for the type of number in calling party DNs. Cisco CallManager sets the calling DN type. Cisco recommends you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does	Use these definitions for each of the variables: • **CallManager** - The Cisco CallManager sets the DN type. • **International** - Use when you dial outside the dialing plan for your country. • **National** - Use

	not recognize European national dialing patterns. You can also change this setting when connecting to PBXs using routing as a non-national type number.	when you dial within the dialing plan for your country. • **Unknown -** The dialing plan is unknown.
Called Numbering Plan	The format for the numbering plan in called party DNs. Cisco CallManager sets the called DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when connecting to PBXs using routing as a non-national type number.	Use these definitions for each of the variables: • **CallManager** - The Cisco CallManager sets the Numbering Plan in the DN. • **ISDN -** Use when you dial outside the dialing plan for your country. • **National Standard -** Use when you dial within the dialing plan for your country. • **Private -** Use when you dial within a 'private' network. • **Unknown -** The dialing plan is unknown.
		Use these definitions for each of the

Calling Numbering Plan	The format for the numbering plan in calling party DNs. Cisco CallManager sets the calling DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans, such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when connecting to PBXs using routing as a non-national type number.	variables: **CallManager -** The Cisco CallManager sets the Numbering Plan in the DN. **ISDN -** Use when you dial outside the dialing plan for your country. **National Standard -** Use when you dial within the dialing plan for your country. **Private -** Use when you dial within a 'private network. **Unknown -** The dialing plan is unknown.
PRI Protocol Type	The communications protocol for the span: • 4E=AT&T InterExchange carrier • 5E8 Custom=Nortel PBX's • 5E9 and NI2=AT&T family local exchange switch or carrier • Australian=European ISDN • DMS=MCI family local exchange switch or carrier	Determine the switch to which you are connecting and the preferred protocol, as such: • Nortel Meridian=5E8 Custom • Lucent Definity=4ESS or 5E8 • Madge (Teleos) box=5E8 Teleos • Intecom PBX=5E8 Intecom Alternatively, select

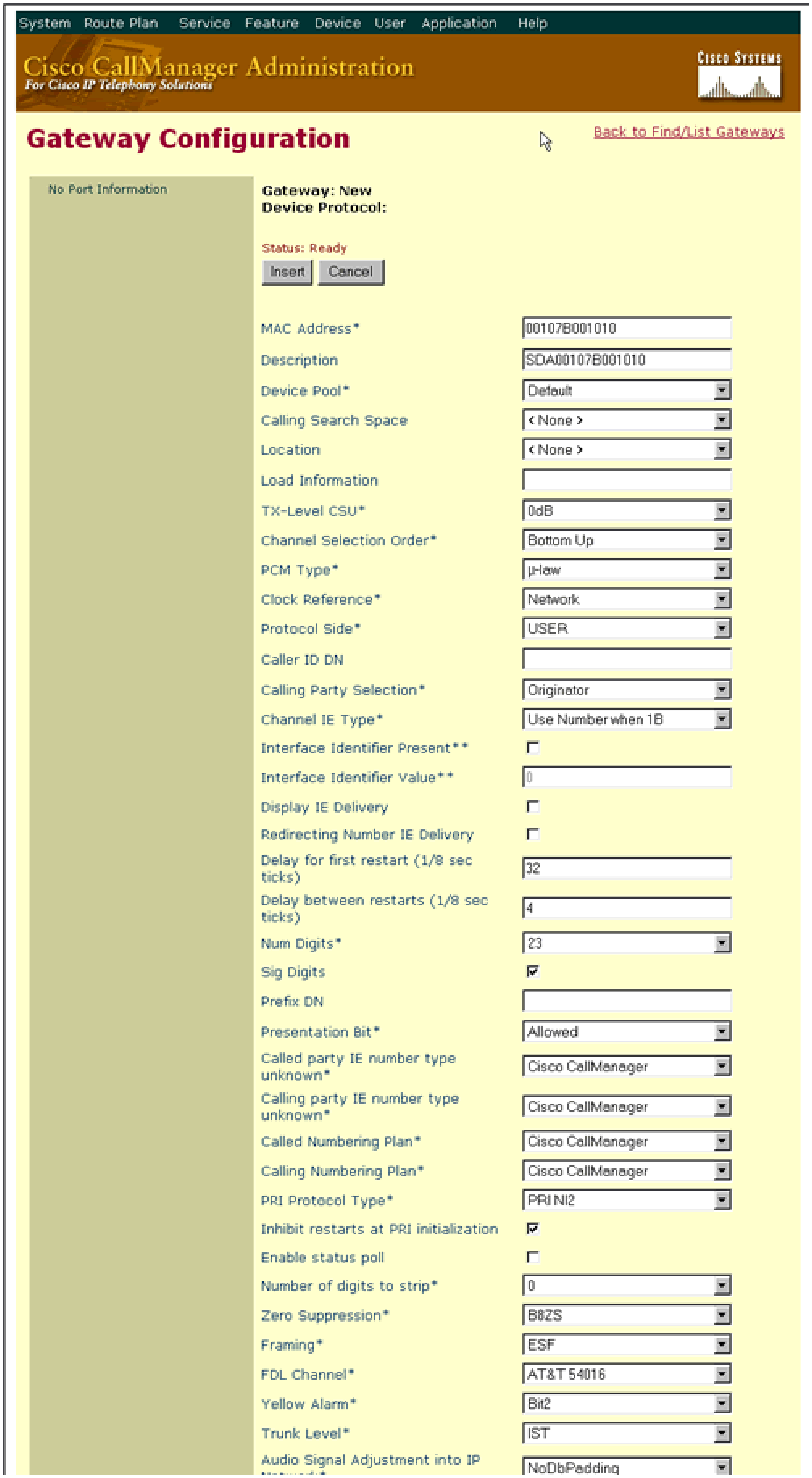
	• ETSI SC=European local exchange carrier on T1; also Japanese local exchange • Euro=European ISDN	the protocol based on the carrier: • MCI=DMS-250 • Sprint=DMS-250 or DMS-100 • AT&T=4ESS
Inhibit restarts at PRI initialization	A RESTART is a message that confirms the status of the ports on a PRI span. If RESTARTs are not sent, they are assumed to be in service.	Enable or disable. When the D-Channel successfully connects with another PRI's D-Channel it sends restarts when this option is disabled.
Enable status poll	Enable to view the B-channel status in the debug window.	
Number of digits to strip	The number of digits to strip on outbound calls, from 0 to 32.	For example, 8889725551234 is dialed, and the number of digits to strip is 3. In this example, 888 is stripped from the outbound number.
Zero Suppression	Determines how the T1 or E1 span electrically codes binary 1's and 0's on the wire (line coding selection).	For a T1, this could be AMI (Alternate Mark Inversion) or B8ZS (Bipolar 8-Zeros Substitution). For an E1, this could be AMI or HDB3 .
Framing	Determines the multiframe format of the span.	The choices are (for T1): • **SF -** Superframe which consists of 12 frames • **ESF -** Extended

		superframe which consists of 24 frames. E1 is always Extended Superframe (ESF) (consisting of 16 frames)
FDL Channel	Determines what kind, if any, facility data link (FDL) is supported by the span. The FDL is a maintenance channel that allows remote troubleshooting of link-layer problems, and remote monitoring of performance statistics of the link.	Only relevant on T1 spans. The choices are: • ANSI T.401 • AT&T PUB 54016 • none
Yellow Alarm	Determines how a remote alarm indication is coded on a T1 span. A yellow alarm indicates that the other end of the link has lost frame synchronization on the signal being transmitted by this end.	Choices include F-bit (out of band signaling; allows 64kbps clear channel bearer capability per B-channel), or bit-2 (in band signaling; robs bit 2 of every channel).
Trunk Level	Adjusts the gain of audio that enters or leaves the span.	
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 you want applied to the received audio signal: • No DbPadding • Plus1db • Plus2db • Plus3db • Plus4db • Plus5db

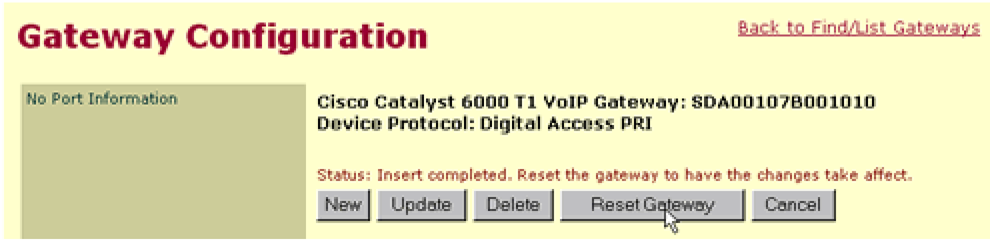
		• Plus6db • Plus7db • Plus8db • Plus9db • Plus10db
Audio Signal Adjustment from IP Network	Specifies the gain or loss applied to the transmitted audio signal relative to the port application type.	Select the gain you want applied to the received audio signal: • No DbPadding • Plus1db • Plus2db • Plus3db • Plus4db • Plus5db • Plus6db • Plus7db • Plus8db • Plus9db • Plus10db
Card	Select the slot position from the drop-down list box. A slot position refers to the	Only appears on a DT-24 Gateway. When you add a new card to the digital access, always add cards from right to left when viewing the gateway from the back. The first (oldest) card should be in the right-most slot, and each

Location	peripheral component interconnect (PCI) card slot into which the digital signal processor (DSP) card is plugged.	subsequent card should be installed in the next available slot position, moving from right to left. If you have existing cards that were not installed in the right-most positions, move the original cards to the right-most slots before adding the new card.

Specify the appropriate parameters for your environment:

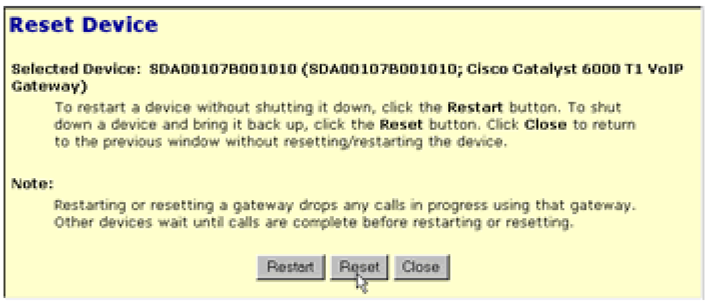

System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Gateway Configuration
Back to Find/List Gateways
No Port Information
Gateway: New
Device Protocol:
Status: Ready
Insert
Cancel
MAC Address* 00107B001010
Description SDA00107B001010
Device Pool* Default
Calling Search Space < None >
Location < None >
Load Information
TX-Level CSU* 0dB
Channel Selection Order* Bottom Up
PCM Type* µ-law
Clock Reference* Network
Protocol Side* USER
Caller ID DN
Calling Party Selection* Originator
Channel IE Type* Use Number when 1B
Interface Identifier Present**
Interface Identifier Value** 0
Display IE Delivery
Redirecting Number IE Delivery
Delay for first restart (1/8 sec ticks) 32
Delay between restarts (1/8 sec ticks) 4
Num Digits* 23
Sig Digits
Prefix DN
Presentation Bit* Allowed
Called party IE number type unknown* Cisco CallManager
Calling party IE number type unknown* Cisco CallManager
Called Numbering Plan* Cisco CallManager
Calling Numbering Plan* Cisco CallManager
PRI Protocol Type* PRI NI2
Inhibit restarts at PRI initialization
Enable status poll
Number of digits to strip* 0
Zero Suppression* B8ZS
Framing* ESF
FDL Channel* AT&T 54016
Yellow Alarm* Bit2
Trunk Level* IST
Audio Signal Adjustment into IP NoDbPadding

Click **Insert** when you have completed this screen. The Gateway Configuration screen appears:



Note: Only the top of the screen appears in the picture.

5. Click **Reset Gateway**.



6. Click **Reset**.

7. Repeat steps 4 and 5 as necessary for the rest of the ports that your configuration uses.

8. You have now completed the basic steps required to add and configure this T1 gateway. After a couple of minutes, the WS-X6608 port finishes the registration process with the Cisco CallManager server. Issue the **show *<module>*** command on the switch to verify that the registration process has been successful.

   In this case, registration is successful. The `Type = T1`, `Call-Manager = 172.16.14.66`, and `CallManagerState = registered`.

```
AV-6509-1 (enable) sh port 5/1


Port  Name                 Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 5/1                       notconnect 64           full 1.544 T1

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 5/1     disable 00-10-7b-00-10-10 172.16.14.72    255.255.255.224

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 5/1     172.16.14.66      -               172.16.14.66    172.16.14.65

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
 5/1     172.16.13.130     -
```

```
Port     CallManagerState DSP-Type
-------- ---------------- --------
 5/1     registered       C549
```

Create the Catalyst 6000 T1 VoIP Gateway in Cisco CallManager 4.x

This task explains how to configure the T1 gateway port on the Cisco CallManager server.

Note: The E1 configuration is very similar.

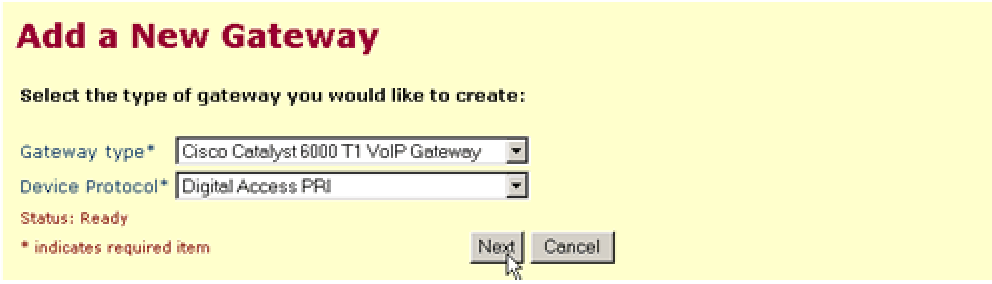
Step-by-Step Instructions

Complete these steps to configure the T1 gateway port:

1. Select **Gateway** from the **Device** menu. A screen similar to this appears:

2. Click **Add a New Gateway**. The Find and List Gateways screen appears:

3. Select the Gateway Type as **Cisco Catalyst 6000 T1 VoIP Gateway** and Device Protocol as **Digital Access PRI**.



Click **Next**.

4. Fill in the MAC Address of the ports on the WS-X6608-T1 blade.

   The MAC Address in this example is from port 5/1 of the WS-6608-T1 blade on the Catalyst 6000 switch. You can find this information if you issue the **show port** command, as shown in this example:

```
AV-6509-1 (enable) show port 5


(Text Deleted)

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 5/1     disable 00-10-7b-00-10-10 172.16.14.70    255.255.255.224
 5/2     disable 00-10-7b-00-10-11 172.16.14.71    255.255.255.224
 5/3     disable 00-10-7b-00-10-12 172.16.14.73    255.255.255.224
 5/4     enable  00-10-7b-00-10-13 0.0.0.0         0.0.0.0
 5/5     disable 00-10-7b-00-10-14 172.16.14.25    255.255.255.224
 5/6     disable 00-10-7b-00-10-15 172.16.14.26    255.255.255.224
 5/7     disable 00-10-7b-00-10-16 172.16.14.81    255.255.255.224
 5/8     disable 00-10-7b-00-10-17 172.16.14.80    255.255.255.224
```

Access Digital PRI Gateway Configuration Settings

Note: Cisco CallManager supports these aspects of the Q Signaling (QSIG) protocol: QSIG basic call, direct dialing inward, name identification and restriction, call diversion, message waiting indication, and transfer services. In order to determine whether your gateway supports the QSIG protocol, refer to the gateway product documentation.

This table provides descriptions of the fields in the E1 and T1 PRI Gateway Configuration settings window:

Device Information	
MAC Address	For Non-Cisco IOS Media Gateway Control Protocol (MGCP) gateways, this field contains the MAC address of the particular port on the 6608 blade.
Description	Enter a description that clarifies the purpose of the device.

Device Pool	From the drop-down list box, choose the appropriate device pool. The device pool specifies a collection of properties for this device including CallManager Group, Date/Time Group, Region, and Calling Search Space for auto-registration of devices.
Network Locale	From the drop-down list box, choose the locale that is associated with the gateway. The network locale identifies a set of detailed information to support the hardware in a specific location. The network locale contains a definition of the tones and cadences that are used by the device in a specific geographic area. **Note:** Choose only a network locale that is already installed and supported by the associated devices. The list contains all available network locales for this setting, but not all are necessarily installed. If the device is associated with a network locale that it does not support in the firmware, the device fails to come up.
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 (MOH) server, among the available media resources according to the priority order that is defined in a Media Resource List.
Location	Choose the appropriate location for this device. The location specifies the total bandwidth that is available for calls to and from this location. A location setting of None means that the locations feature does not keep track of the bandwidth that is consumed by this device.
AAR Group	Choose the automated alternate routing (AAR) group for this device. The AAR group provides the prefix digits that are used to route calls that are otherwise blocked due to

	insufficient bandwidth. An AAR group setting of None specifies that no rerouting of blocked calls is attempted.
Load Information	Enter the appropriate firmware load information for the gateway. The value that you enter here overrides the default firmware load for this gateway type.
Multilevel Precedence and Preemption (MLPP) Information	
MLPP Domain (such as 0000FF)	Enter a hexadecimal value between 0 and FFFFFF for the MLPP domain associated with this device. If you leave this field blank, this device inherits its MLPP domain from the value set for the MLPP Domain Identifier enterprise parameter.
MLPP Indication	If available, this setting specifies whether a device capable of playing precedence tones uses the capability when it places an MLPP precedence call. From the drop-down list box, choose a setting to assign to this device from these options: • **Default -** This device inherits its MLPP indication setting from its device pool. • **Off -** This device does not handle nor process indication of an MLPP precedence call. • **On -** This device does handle and process indication of an MLPP precedence call. **Note:** Do not configure a device with this combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.
	If available, this setting specifies

MLPP Preemption	whether a device capable of preempting calls in progress uses the capability when it places an MLPP precedence call. From the drop-down list box, choose a setting to assign to this device from these options: • **Default -** This device inherits its MLPP preemption setting from its device pool. • **Disabled -** This device does not allow preemption of lower-precedence calls to take place when necessary for completion of higher-precedence calls. • **Forceful -** This device allows preemption of lower-precedence calls to take place when necessary for completion of higher-precedence calls. **Note:** Do not configure a device with this combination of settings: MLPP Indication is set to Off or Default (when default is Off) while MLPP Preemption is set to Forceful.
Interface Information	
	Choose the communications protocol for the span. If you include this gateway in a route group, you cannot switch between QSIG and non-QSIG protocol types until you remove the gateway from the route group. T1 PRI spans have several options, which depend on the carrier or switch; for example: • **PRI 4ESS -** AT&T Interexchange carrier • **PRI 5E8 Custom -** Cisco IP Phone • **PRI 5E9 -** AT&T family local

PRI Protocol Type	exchange switch or carrier • **PRI DMS -** MCI family local exchange switch or carrier; Canadian local exchange carrier • **PRI ETSI SC -** European local exchange carrier on T1; also, Japanese, Taiwan, Korean, and Hong Kong local exchange • **PRI NI2 -** AT&T family local exchange switch or carrier • **PRI NTT -** Japanese NTT exchange switch • **PRI ISO QSIG T1**-PBX T1 tie trunk using International Organization for Standardization (ISO) QSIG • **PRI ISO QSIG E1 -** PBX E1 tie trunk using ISO QSIG Determine the switch to which you are connecting and the preferred protocol; for example: • **Nortel Meridian -** DMS, 5E8 Custom • **Lucent Definity** - 4ESS or 5E8 • **Madge (Teleos) box -** 5E8 Teleos • **Intecom PBX -** 5E8 Intecom
Protocol Side	Choose the appropriate protocol side. This setting specifies whether the gateway connects to a CO or Network device or to a User device. Make sure that the two ends of the PRI connection use opposite settings. For example, if you connect to a PBX and the PBX uses the User as its protocol side, choose Network for this device. Typically, use User for this

	option for CO connections.
Channel Selection Order	Choose the order in which channels or ports are enabled from first (lowest number port) to last (highest number port), or from last to first. Valid entries include TOP_DOWN (first to last) or BOTTOM_UP (last to first). If you are not sure which port order to use, choose TOP_DOWN.
Channel IE Type	Choose one of these values to specify whether channel selection is presented as a channel map or a slot map: • **Timeslot Number--B -** Channel usage always indicates actual timeslot map format (such as 1 through 15 and 17 through 31 for E1). • **Slotmap--B -** Channel usage always indicates a slot map format. • **Use Number When 1B -** Channel usage indicates a channel map for one B-channel but indicates a slot map if more than one B-channel exists. • **Continuous Number -** Configures a continuous range of slot numbers (1 through 30) as the E1 logical channel number instead of the noncontinuous actual timeslot number (1 through 15 and 17 through 31).
Pulse Code Modulation (PCM) Type	Specify the digital encoding format. Choose one of these formats: • **A-law -** Use for Europe and the rest of the world. • **mu-law -** Use for North America, Hong Kong, Taiwan, and Japan.
	Enter the rate at which the spans are

Delay for first restart (1/8 second ticks)	brought in service. The delay occurs when many PRI spans are enabled on a system and the Inhibit Restarts at PRI Initialization check box is unchecked. For example, set the first five cards to 0 and set the next five cards to 16. (Wait two seconds before you bring them in service.)
Delay between restarts (1/8 second ticks)	Enter the time between restarts. The delay occurs when a PRI RESTART gets sent if the Inhibit Restarts check box is unchecked.
Inhibit restarts at PRI initialization	A RESTART or SERVICE message confirms the status of the ports on a PRI span. If RESTART or SERVICE messages are not sent, Cisco CallManager assumes the ports are in service. When the D-Channel successfully connects with another PRI D-Channel, it sends a RESTART or SERVICE message when this check box is unchecked.
Enable status poll	Check the check box to view the B-channel status in the debug window.
Call Routing Information	
Inbound Calls	
Significant Digits	Choose the number of significant digits to collect, from 0 to 32 or All. Cisco CallManager counts significant digits from the right (last digit) of the number called. If you choose All , the Cisco CallManager does not truncate the inbound number. For example, the digits received are 123456. The Significant Ddigits setting is 4. Digits translated are 3456. Use to process of incoming calls and to indicate the number of digits, starting from the last digit of the called number, that are used to route calls that come into the PRI span. Also, see the Prefix DN field.
	Choose the appropriate calling search

Calling Search Space	space. A calling search space designates a collection of route partitions that are searched to determine how a collected (originating) number should be routed.
AAR Calling Search Space	Choose the appropriate calling search space for the device to use when automated alternate routing (AAR) is performed. The AAR calling search space specifies the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth.
Prefix DN	Enter the prefix digits that are appended to the digits that this trunk receives on incoming calls. The Cisco CallManager adds prefix digits after it first truncates the number in accordance with the Num Digits setting.
Calling Line ID Presentation	Choose whether you want the Cisco CallManager to transmit or block the caller's phone number. Choose Default if you do not want to change calling line ID presentation. Choose Allowed if you want Cisco CallManager to send `Calling Line ID Allowed` on outbound calls. Choose Restricted if you want Cisco CallManager to send `Calling Line ID Restricted` on outbound calls. For more information about this field, refer to Table 17-6 in the "Calling Party Number Transformations Settings" section in Understanding Route Plans.
	Any outbound call on a gateway can send DN information. Choose which DN is sent: • **Originator -** Send the DN of the calling device.

Calling Party Selection	• **First Redirect Number -** Send the DN of the redirecting device. • **Last Redirect Number -** Send the DN of the last device to redirect the call. • **First Redirecting Party (External) -** Send the DN of the first redirecting device with the external phone mask applied. • **Last Redirecting Party (External) -** Send the DN of the last redirecting device with the external phone mask applied.
Called party IE number type unknown	Choose the format for the number type in called party DNs. Cisco CallManager sets the called DN type. It is recommended that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the called DN to be encoded to a non-national type numbering plan. Choose one of these options: • **CallManager -** The Cisco CallManager sets the DN type. • **International -** Use when you dial outside the dialing plan for your country. • **National -** Use when you dial within the dialing plan for your country.

	• **Unknown -** The dialing plan is unknown.
Calling party IE number type unknown	Choose the format for the number type in calling party DNs. Cisco CallManager sets the calling DN type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the calling DN to be encoded to a non-national type numbering plan. Choose one of these options: • **CallManager -** The Cisco CallManager sets the DN type. • **International -** Use when you dial outside the dialing plan for your country. • **National -** Use when you dial within the dialing plan for your country. • **Unknown -** The dialing plan is unknown.
	Choose the format for the numbering plan in called party DNs. Cisco CallManager sets the called DN numbering plan. It is recommended that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting

Called Numbering Plan	to PBXs by using routing as a non-national type number. Choose one of these options: • **CallManager -** The Cisco CallManager sets the Numbering Plan in the DN. • **ISDN -** Use when you dial outside the dialing plan for your country. • **National Standard -** Use when you dial within the dialing plan for your country. • **Private -** Use when you dial within a private network. • **Unknown -** The dialing plan is unknown.
Calling Numbering Plan	Choose the format for the numbering plan in calling party DNs. Cisco CallManager sets the calling DN numbering plan. It is recommended that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco CallManager does not recognize European national dialing patterns. You can also change this setting when you are connecting to PBXs by using routing as a non-national type number. Choose one of these options: • **CallManager -** The Cisco CallManager sets the Numbering Plan in the DN. • **ISDN -** Use when you dial outside the dialing plan for your country.

	National Standard - Use when you dial within the dialing plan for your country. **Private -** Use when you dial within a private network. **Unknown -** The dialing plan is unknown.
Number of digits to strip	Choose the number of digits to strip on outbound calls, from 0 to 32. For example, when 8889725551234 is dialed, and the number of digits to strip is 3, Cisco CallManager strips 888 from the outbound number.
Caller ID DN	Enter the pattern that you want to use for caller ID, from 0 to 24 digits. For example, in North America: 555xxxx = Variable caller ID, where x equals an extension number. The CO appends the number with the area code if you do not specify it. 5555000 = Fixed caller ID, where you want the Corporate number to be sent instead of the exact extension from which the call is placed. The CO appends the number with the area code if you do not specify it.
SMDI Base Port	Enter the first Simplified Message Desk Interface (SMDI) port number of the T1 span.
Type Specific Information	
Display IE Delivery	Check the check box to enable delivery of the display information element (IE) in SETUP, and NOTIFY messages (for DMS protocol) for the calling and connected party name delivery service. **Note:** Default leaves the check box unchecked.

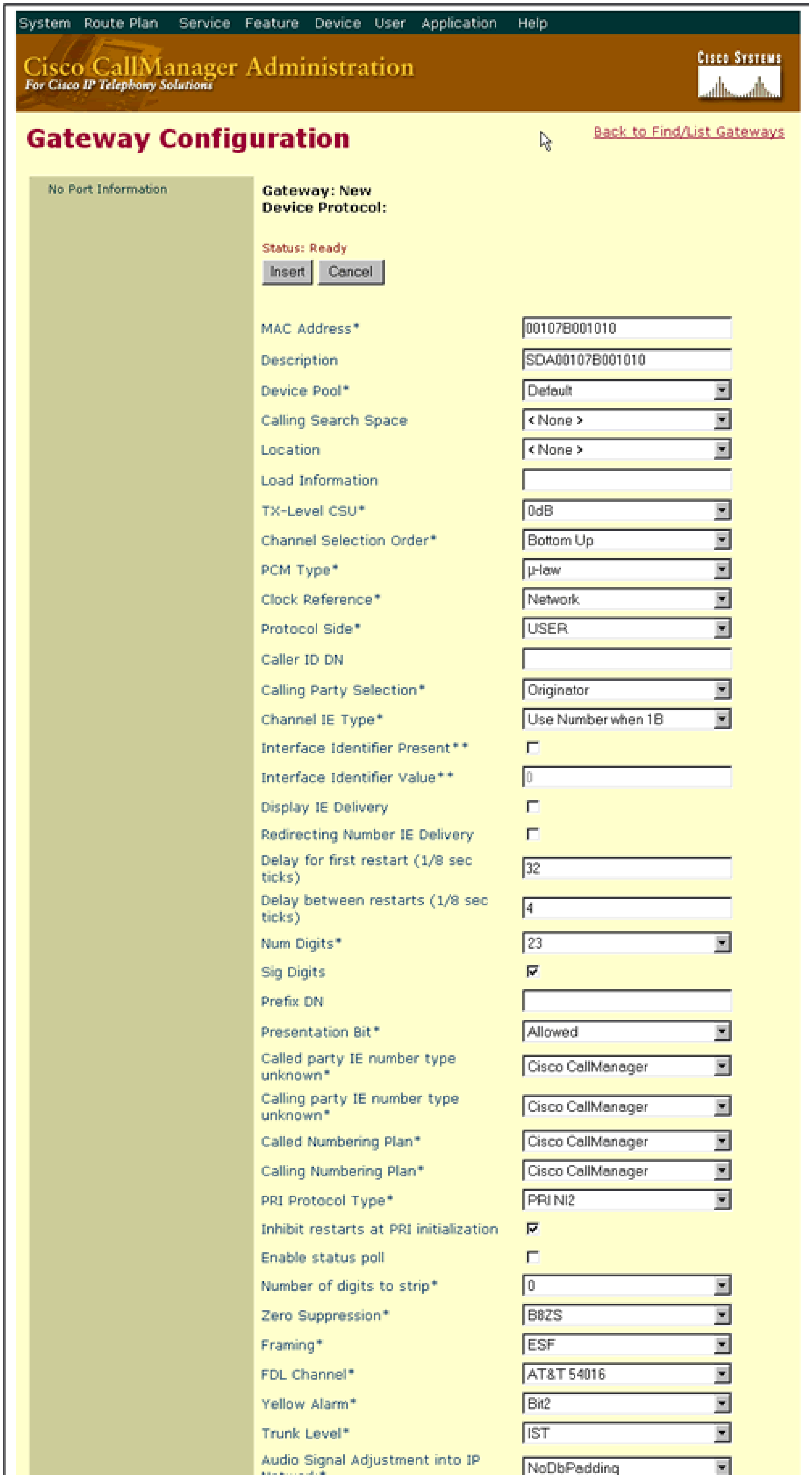
Redirecting Number IE Delivery--Outbound	Check this check box to include the Redirecting Number IE in the outgoing SETUP message from the Cisco CallManager to indicate the first redirecting number and the redirecting reason of the call when the call is forwarded. Uncheck the check box to exclude the first redirecting number and the redirecting reason from the outgoing SETUP message. You use the Redirecting Number IE for voice-mail integration only. If your configured voice-mail system supports Redirecting Number IE, you should check the check box. **Note:** Default leaves the check box unchecked.
Redirecting Number IE Delivery--Inbound	Check this check box to accept the Redirecting Number IE in the incoming SETUP message to the Cisco CallManager. Uncheck the check box to exclude the Redirecting Number IE in the incoming SETUP message to the Cisco CallManager. You use Redirecting Number IE for voice-mail integration only. If your configured voice-mail system supports Redirecting Number IE, you should check the check box. **Note:** Default leaves the check box unchecked.
Send Extra Leading Character in DisplayIE	Check this check box to include a special leading character byte (non ASCII, nondisplayable) in the DisplayIE field. Uncheck this check box to exclude this character byte from the DisplayIE field. This check box only applies to the

	DMS-100 protocol and the DMS-250 protocol. Default leaves this setting disabled (unchecked).
Setup non-ISDN Progress Indicator IE Enable	Default leaves this setting disabled (unchecked). Enable this setting only if users do not receive ringback tones on outbound calls. When this setting is enabled, the Cisco CallManager sends Q.931 SETUP messages out digital (that is, non-H.323) gateways with the Progress Indicator field set to non-ISDN. This message notifies the destination device that the Cisco CallManager gateway is non-ISDN and that the destination device should play in-band ringback. This problem usually associates with Cisco CallManagers that connect to PBXs through digital gateways.
MCDN Channel Number Extension Bit Set to Zero	To set the channel number extension bit to zero, check the check box. To set the extension bit to 1, uncheck the check box. This setting only applies to the DMS-100 protocol.
Send Calling Name in Facility IE	Check the check box to send the calling name in the Facility IE field. By default, the Cisco CallManager leaves the check box unchecked. Set this feature for a private network that has a PRI interface enabled for ISDN calling name delivery. When this check box is checked, the calling party's name gets sent in the Facility IE of the SETUP or FACILITY message, so the name can display on the called party's device.

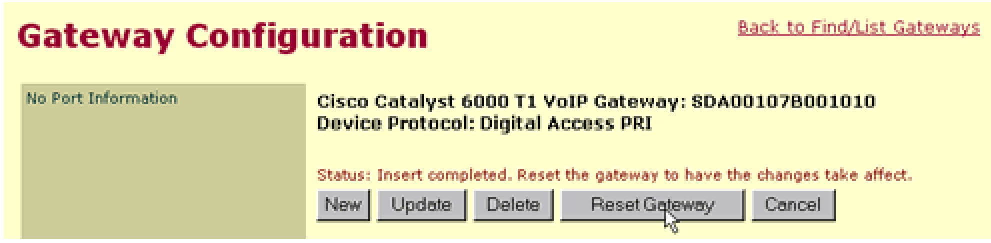
	Set this feature for PRI trunks in a private network only. Do not set this feature for PRI trunks connected to the PSTN. **Note:** This field applies to the NI2 protocol only.
Interface Identifier Present	Check the check box to indicate that an interface identifier is present. By default, the Cisco CallManager leaves the check box unchecked. This setting only applies to the DMS-100 protocol for digital access gateways in the Channel Identification information element (IE) of the SETUP, CALL PROCEEDING, ALERTING, and CONNECT messages.
Interface Identifier Value	Enter the value that was obtained from the PBX provider. This field applies to only the DMS-100 protocol. Valid values range from 0 to 255.
Connected Line ID Presentation	Choose whether you want the Cisco CallManager to allow or block the connected party's phone number from displaying on an inbound caller's phone. This field applies only to gateways that are using QSIG protocol. The gateway applies this setting for incoming calls only. Choose Default if you do not want to change the connected line ID presentation. Choose Allowed if you want Cisco CallManager to send the `Connected Line ID Allowed` message to enable the connected party's number to display for the calling party. Choose Restricted if you want Cisco CallManager to send the `Connected Line ID Restricted` message to block the connected party's number from

	displaying for the calling party. For more information about this field, refer to Table 17-9 in the "Connected Party Presentation and Restriction Settings" section in Understanding Route Plans.
Connected PBX Model	Choose the type and model of the PBX or VoIP switch with which this gateway communicates. This field applies only to gateways that are using QSIG protocol. Options include: • Siemens Hicom • Ericsson MD-110 • Alcatel PBX • Meridian Option 11C • Lucent Definity G3 • IPC MX • Cisco CallManager
Product-Specific Configuration	
Model-specific configuration fields that are defined by the gateway manufacturer	The model-specific fields under product-specific configuration define the gateway manufacturer. Because they are dynamically configured, they can change without notice. To view field descriptions and help for product-specific configuration items, click the **i** information icon to the right of the Product Specific Configuration heading to display help in a popup dialog box. If you need more information, refer to the documentation for the specific gateway that you are configuring or contact the manufacturer.

Specify the appropriate parameters for your environment:

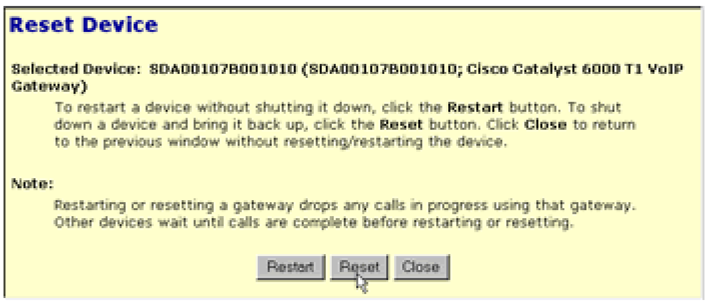

System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Cisco Systems
Gateway Configuration
Back to Find/List Gateways
No Port Information
Gateway: New
Device Protocol:
Status: Ready
Insert
Cancel
MAC Address* 00107B001010
Description SDA00107B001010
Device Pool* Default
Calling Search Space < None >
Location < None >
Load Information
TX-Level CSU* 0dB
Channel Selection Order* Bottom Up
PCM Type* µ-law
Clock Reference* Network
Protocol Side* USER
Caller ID DN
Calling Party Selection* Originator
Channel IE Type* Use Number when 1B
Interface Identifier Present**
Interface Identifier Value** 0
Display IE Delivery
Redirecting Number IE Delivery
Delay for first restart (1/8 sec ticks) 32
Delay between restarts (1/8 sec ticks) 4
Num Digits* 23
Sig Digits
Prefix DN
Presentation Bit* Allowed
Called party IE number type unknown* Cisco CallManager
Calling party IE number type unknown* Cisco CallManager
Called Numbering Plan* Cisco CallManager
Calling Numbering Plan* Cisco CallManager
PRI Protocol Type* PRI NI2
Inhibit restarts at PRI initialization
Enable status poll
Number of digits to strip* 0
Zero Suppression* B8ZS
Framing* ESF
FDL Channel* AT&T 54016
Yellow Alarm* Bit2
Trunk Level* IST
Audio Signal Adjustment into IP Network* NoDbPadding

Click **Insert** when you have completed this screen. The Gateway Configuration screen appears:



Note: Only the top of the screen is displayed in the picture above.

5. Click **Reset Gateway**.



6. Click **Reset**.

7. Repeat steps 4 and 5 as necessary for the rest of the ports that your configuration uses.

8. You have now completed the basic steps required to add and configure this T1 gateway. After a couple of minutes the WS-X6608 port finishes the registration process with the Cisco CallManager server. Issue the **show** ***<module>*** command on the switch to verify whether the registration process is successful.

   In this sample output, registration is successful. The `Type = T1`, the `Call-Manager = 172.16.14.66`, and the `CallManagerState = registered`.

```
AV-6509-1 (enable) sh port 5/1


Port  Name                 Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 5/1                       notconnect 64           full 1.544 T1

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
 5/1     disable 00-10-7b-00-10-10 172.16.14.72    255.255.255.224

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
 5/1     172.16.14.66      -               172.16.14.66    172.16.14.65

Port     DNS-Server(s)     Domain
-------- ----------------- -----------------------------------------------
 5/1     172.16.13.130     -
```

```
Port     CallManagerState DSP-Type
-------- ---------------- --------
 5/1     registered       C549
```

Verify the Catalyst/CallManager Configuration

This section provides information you can use to confirm your configuration is working properly.

The Output Interpreter Tool (registered customers only) (OIT) supports certain **show** commands. Use the OIT to view an analysis of **show** command output.

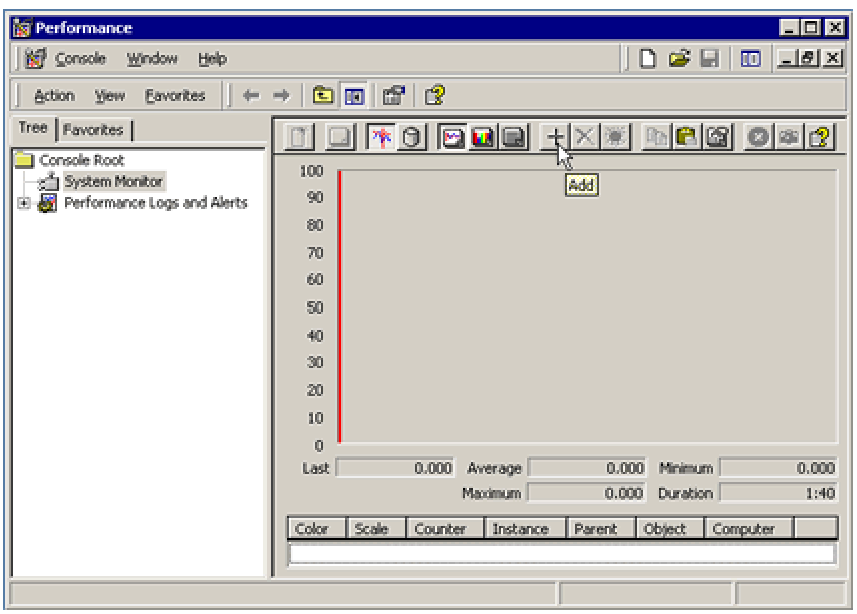
Use Performance Monitor to Analyze WS-X6608-T1 Calls and Status Changes

This task demonstrates how to use the performance monitor to analyze WS-X6608-T1 calls and status changes.

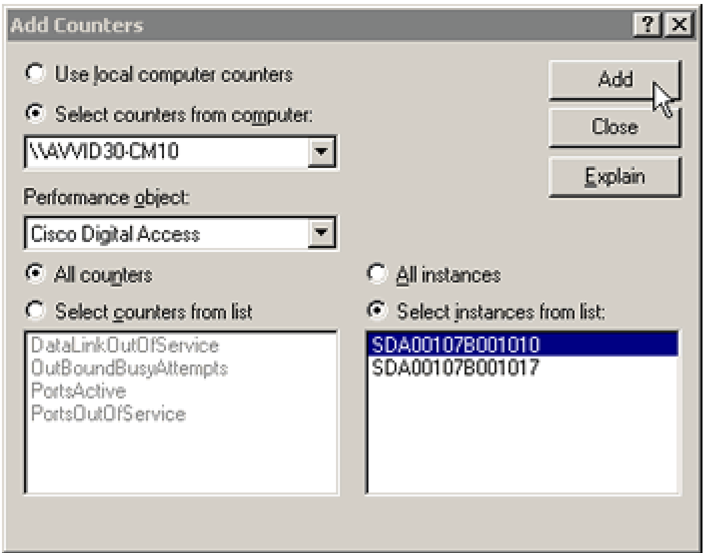
Step-by-Step Instructions

Complete these steps to analyze WS-X6608-T1 calls and status changes with the performance monitor:

1. Start Performance Monitor from the **Start > Programs > Administrative Tools > Performance** option. A screen similar to this appears:



2. Select the**Add (+)** function. The Add Counters screen appears:



Select **Cisco Digital Access** as the Performance Object. Select the **All Counters** option. Finally select the gateway. In this case *SDA00107B001010*. Then click **Add**, and then click **Close**.

If you do not see the instance of the gateway you created check whether the gateway has registered with the Cisco CallManager server. Sometimes, this registration fails.

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

If you still have a problem, try to reset the module from the Catalyst switch with the **reset <*mod_num*>** command. Wait until the registration process is complete. In order to check this, issue the **show port <*mod_num/port_num*>** command and look for the IP address of the Cisco CallManager server.

If the first set of steps does not resolve the problem, continue with these steps:

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 port's IP address 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 are using DHCP, refer to Configuring Windows 2000 DHCP Server for Cisco Call Manager for further information on how to configure and use DHCP. If you want to configure the IP parameters manually, see the Configure the IP Settings on the WS-X6608-T1 Port section.

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

For both DHCP and non DHCP configurations, verify that the VLAN is correct. You cannot set the VLAN of the port with DHCP. You must do so at the CLI of the switch. The syntax is **set vlan <*vlan_number*><*mod_num/port_num*>**. Also verify that the port status is not disabled. The syntax to enable a port is **set port enable <*mod_num/port_num*>**.

Note: Remember that unlike the WS-X6624, you have to configure the IP parameters for each port on the WS-X6608 independently.

This sample output shows the correct IP parameters for this example:

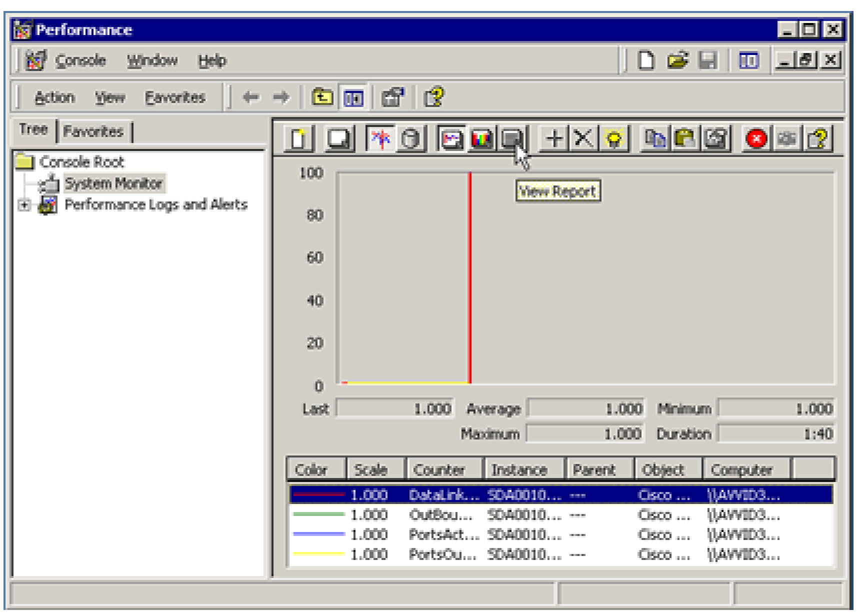
```
AV-6509-1 (enable) sh port 5/1

Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
5/1                     notconnect  64           full 1.544  T1

Port     DHCP    MAC-Address       IP-Address      Subnet-Mask
-------- ------- ----------------- --------------- ---------------
5/1     disable  00-10-7b-00-10-10  172.16.14.72     255.255.255.224

Port     Call-Manager(s)   DHCP-Server     TFTP-Server     Gateway
-------- ----------------- --------------- --------------- ---------------
5/1     172.16.14.66      -                172.16.14.66    172.16.14.65

Port     DNS-Server(s)     Domain
-------- ----------------- -------------------------------------------------
5/1      172.16.13.130     -

Port     CallManagerState DSP-Type
-------- ---------------- --------
5/1      registered       C549
```

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

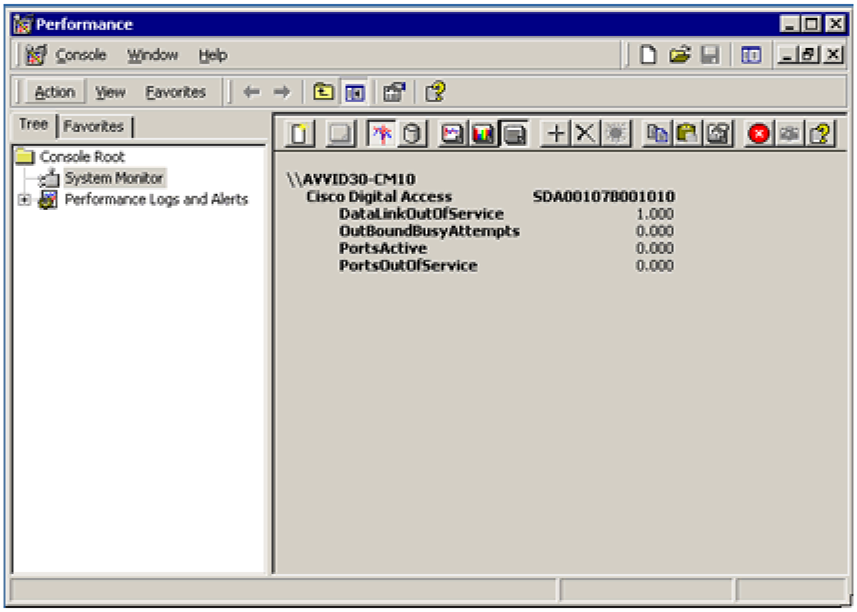
- Configure the IP parameters manually to eliminate DHCP from the equation
- Use IP addresses instead of DNS hostnames

If none of these steps resolves the problem, open a case with Cisco Technical Support.

3. You must see a screen similar to this:



4. Select the **View Report** feature. A screen similar to this appears:



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

Use Performance Monitor for the verification of busied out B channels in WS-X6608-T1

For the verification of busied out B channels in WS-X6608-T1 module you can use the same procedure described in the Use Performance Monitor to Analyze WS-X6608-T1 Calls and Status Changes section.

Complete these steps to view the busied out B channels in a T1 PRI MGCP configuration.

1. Start Performance Monitor from the **Start > Programs > Administrative Tools > Performance** option.

2. Select **Add (+)** function

   . Select **Cisco MGCP PRI Devices** as the Performance Object from the pull down. Select the **All Counters** option. Then click **Add**, and then click **Close**.

3. Now go to **View report,** you must be able to see the B-channel and D-channel status for individual channel. The status of the B-Channels associated with this MGCP PRI device are represented with these possible values: **0 (Unknown)** indicates the status of the channel could not be determined; **1 (Out of service)** indicates that this channel is not available for use; **2 (Idle)** indicates that this channel has no active call and is ready for use; **3 (Busy)** indicates an active call on this channel; **4 (Reserved)** indicates that this channel has been reserved for use as a D-channel or for use as a Synch-Channel for E-1.

Use the Catalyst CLI to Analyze WS-X6608-T1 Activity

This task demonstrates some of the commands that you can use on the Catalyst 6000 to verify that

Catalyst 6000 communicates with the Cisco CallManager server. This section of the document also shows some commands that you can use to track calls and call related statistics.

Step-by-Step Instructions

Complete these steps to verify the Catalyst 6000 communicates with the Cisco CallManager server:

1. Issue the **show port <*mod/port*>** command to display the status of any module.

   In this case ***show port 7*** displays the current state of the ports on the WS-X6608 blade. In the sample output here, all eight ports have registered with the Cisco CallManager server as T1 gateways. The other types are Unknown, media termination point (MTP) and Conf Bridge.

```
Console> show port 7

Port  Name               Status     Vlan       Duplex Speed Type
----- ------------------ ---------- ---------- ------ ----- ------------
 7/1                     connected  1          full   1.544 T1
 7/2                     connected  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                     connected  1          full   1.544 T1
 7/8                     connected  1          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 172.20.34.78    255.255.255.0
 7/8     enable  00-10-7b-00-0a-5f 172.20.34.69    255.255.255.0

Port     Call-Manager(s)   DHCP-Server     TFTP-Sever      Gateway
-------- ----------------- --------------- --------------- ---------------
 7/1     172.20.34.207*    172.20.34.207   172.20.34.207   172.20.34.20
 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   172.20.34.20
 7/4     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20
 7/5     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20
 7/6     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20
 7/7     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20
 7/8     172.20.34.207     172.20.34.207   172.20.34.207   172.20.34.20

Port     DNS-Server(s)   Domain
-------- --------------- -------------------------------------------------
 7/1     172.20.34.207   cisco.com
 7/2     172.20.34.207*  cisco.com
 7/3     172.20.34.207   cisco.com
 7/4     172.20.34.207   cisco.com
 7/5     172.20.34.207   cisco.com
 7/6     172.20.34.207   cisco.com
 7/7     172.20.34.207   cisco.com
```

```
 7/8     172.20.34.207   cisco.com

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  disabled   disabled
 7/2  disabled   disabled
 7/3  disabled   disabled
 7/4  disabled   disabled
 7/5  disabled   disabled
 7/6  disabled   disabled
 7/7  disabled   disabled
 7/8  disabled   disabled

(*): Primary
Console>
```

Syntax Description

mod/port	(Optional) Number of the module and optionally, the number of the port on the module.

Command Types

Switch command

Command Nodes

Normal

Usage Guidelines

- If you do not specify a *mod*, the ports on all modules are shown.
- If you do not specify a *port*, all the ports on the module are shown.
- The output for an 8-port T1/E1 PSTN interface module configured for transcoding and/or conferencing displays a transcoding port type as MTP or a conference port type as *conf bridge*.
- The output for an 8-port T1/E1 PSTN interface module displays a transcoding port type as *transcoding* or a conference port type as *conferencing*.

2. Issue the **show port voice fdl** command to display the facilities data link statistics for the specified ports: **show port voice fdl <*mod/port*>**. FDL is a link management protocol used to help diagnose problems and gather statistics.

   In this case, **show port voice fdl *7/1-3*** displays information for the ports on the WS-X6608-T1 blade.

```
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)
```

   This table describes the possible fields (depending on the port type queried) in the **show port voice fdl** command output.

   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.

Syntax Description

mod [/port]	Optional) Number of the module and (optional) ports.

Defaults

This command has no default setting.

Command Types

Switch command

Command Modes

Privileged

Usage Guidelines

This command is not supported by the network applications management (NAM).

3. Issue the **show port voice active** command to display active call information on a port: **show port voice active [*mod/port*] [all | call | conference | transcode] [*ipaddr*]**. There are no active calls on this system.

   This output shows 0 ports active and 0 calls active.

```
Console> show port voice active

Port  Type         Total Conference-ID/ Party-ID IP-Address
                         Transcoding-ID
----- ------------ ----- -------------- -------- ---------------

Total: 0
Console> (enable)
Console> (enable) show port voice active call

Port  Total IP-Address
----- ----- ---------------
Total: 0 calls
Console> (enable)
```

Syntax Description

mod/port	(Optional) Number of the module and port on the module.
all	(Optional) Keyword to display all calls (regular calls, conference calls, and transcoding calls) in the system.
call	(Optional) Keyword to display call information for the 24-port FXS analog interface and the 8-port T1/E1 PSTN

	interface modules.
conference	(Optional) Keyword to display call information for the 8-port T1/E1 PSTN interface module configured for conferencing.
transcode	(Optional) Keyword to display call information for the 8-port T1/E1 PSTN interface module configured for transcoding.
ipaddr	(Optional) Remote IP address.

Defaults

The default is all active calls are displayed.

Command Types

Switch command

Command Modes

Normal

Usage Guidelines

- The information that appears when you issue the **show port voice active** command is not available through the supervisor engine Simple Network Management Protocol (SNMP) agent.
- The 24-port FXS analog interface and the 8-port T1/E1 PSTN interface modules support the **call** keyword.
- The 8-port T1/E1 PSTN interface module supports the **conference** and **transcode** keywords.
- You can use the ***optional mod*** or ***mod/port*** variables to display calls that belong to the specified module or port in detailed format.
- There are up to eight calls per port for the 8-port T1/E1 ISDN PRI services-configured module but only one call per port for the 24-port FXS analog station interface services-configured module.
- The ***ipaddr*** option displays one specific call for the specified IP address. You can also use an IP alias.
- The NAM does not support this command.

Troubleshoot

Catalyst 6608 Unable to Register to Cisco CallManager 5.x/6.x

The Catalyst 6608 or 6624 gateway with a factory-preinstalled load is unable to register to the Cisco CallManager 5.x/6.x. The workaround for this issue is to register each port on the Catalyst 6608 with Cisco CallManager 4.x to let it download the new load. After this, it can be registered to CallManager 5.x/6.x. This issue is documented in Cisco bug ID CSCeg20715 (registered customers only) .

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
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Voice & Video: Voice over IP
BLF Question - Mar 26, 2008 FXS idle problem - Mar 26, 2008 What is Half NAT entries for the VOIP means - Mar 26, 2008 NAT for H.323 on Cisco 2800 router - Mar 26, 2008 Desktop-Video: Bandwidth needed - Mar 26, 2008
Voice & Video: IP Telephony
FXO disconnection - Mar 27, 2008 soft phone - Mar 27, 2008 B-ACD calls are not directing to a VM - Mar 27, 2008 Cisco Unified Communication Manager 6.1 Fail over - Mar 27, 2008 CCME gives error while Installing - Mar 27, 2008
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Conference bridge failed - Mar 27, 2008 On cucm6, configure ?Personal Fast Dials? as button service? - Mar 26, 2008 Hunt Pilot / Pickup Group - Mar 26, 2008 Extension Mobility issues CUCM 6.0 - Urgent - Mar 25, 2008 Cisco 7940 Call Interupt - Mar 25, 2008
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Cisco 7937 Conference Phone w/UC520? - Mar 27, 2008 directory of ccm in ADS - Mar 27, 2008 Backup Out of Disk Space - Mar 26, 2008 JTAPI call transfer talkpath problem - Mar 26, 2008 Unity VM 4.2 and 2003 Off-box message store - Mar 26, 2008

Voice & Video: IP Phone Services for Developers
Service Parameter - Mar 26, 2008 AXL .net CallManager 6.1 and 4.1(3) - Mar 26, 2008 AXL wsdl and wsdl.exe in .NET: Help! - Mar 26, 2008 error in AXLSoap.xsd - DirectoryNumber ist not an nonNegativeInteger - Mar 26, 2008 Night Ringer - Mar 25, 2008
Voice & Video: General
CME setup for AVAYA IP Agent - Mar 27, 2008 No DND softkey option when call is connected? - Mar 27, 2008 Hunt Pilot Caller ID - Mar 26, 2008 one to many call - Mar 26, 2008 Mutiple UC500 systems at same site. - Mar 26, 2008

Related Information

- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Recommended Reading: Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation - Cisco Systems**