Configuring the Voice Interfaces

This section explains how to configure Voice-over-IP (VoIP) routing on the supported interface cards of your Catalyst 4224 Access Gateway Switch. (You need to perform the software configuration only for the cards that you have installed.) To configure a voice interface, you must use configuration mode. In this mode, you enter Cisco IOS command-line interface (CLI) commands at the Gateway prompt.

This section includes the following topics:

Configuring Voice Interfaces

Use a voice interface card (VIC) for a voice connection. For information about installing these components in a Catalyst 4224, refer to the *Cisco Catalyst 4224 Access Gateway Switch Hardware Installation Guide*. For an explanation about how these components work and how they are identified, see the "First-Time Configuration" section on page 2-1.

When you start a Catalyst 4224, it automatically detects the voice network modules and VICs that have been installed. The first time you use a Catalyst 4224, you need to configure each voice port that you want to enable. If you replace a card after you configure each port, the gateway will recognize the new hardware component and use the previous configuration settings.

If you replace a module that was already configured, the gateway recognizes it and brings up the interface in the existing configuration.

To configure the Catalyst 4224 to boot in a new configuration, perform the following steps:

Step 1 Connect a terminal or a PC running terminal emulation software to the console port on the Catalyst 4224. Refer to the *Catalyst 4224 Access Gateway Switch Hardware Installation Guide* for instructions.

Step 2 Power on the Catalyst 4224. If the current configuration is no longer valid, you see the following prompt within about one minute:

```
Would you like to enter the initial dialog? [yes/no]:
```

Step 3 Enter **no**. You now enter the normal operating mode of the gateway.

> **Note** If the current configuration is valid, you automatically enter normal operating mode.

Step 4 After a few seconds, you see the user EXEC prompt (c4224>). Enter **enable** and the password to enter privileged EXEC mode, as follows:

```
c4224> enable
Password: <password>
c4224# enable
```

Configuration changes can be made only in privileged EXEC mode. When you enter privileged EXEC mode, the prompt changes to the host name followed by a pound sign (#), such as c4224#.

Step 5 Enter the commands required to configure the VICs installed in the Catalyst 4224.

Step 6 When you finish configuring the voice interfaces, return to global configuration mode using the **exit** command and return to enable mode by pressing **Ctrl-Z**.

Step 7   To see the current operating configuration, including any changes you just made, enter the **show running-config** command:

```
c4224# show running-config
```

To see the configuration currently stored in non-volatile random access memory (NVRAM), enter the **show startup-config** command at the enable prompt:

```
c4224# show startup-config
```

Step 8   The results of the **show running-config** and **show startup-config** commands differ from each other if you have changed the configuration but have not yet written the changes to NVRAM. To write your changes to NVRAM and make them permanent, enter the **copy running-config startup-config** command at the enable prompt:

```
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

MGCP Configuration

If you want to use Media Gateway Control Protocol (MGCP), configuration of the Catalyst 4224 differs depending on whether you are using it with Cisco CallManager 3.0 or 3.1.

With Cisco CallManager 3.1 and later, you can create the MGCP gateway configuration on the Cisco CallManager server and download the configuration to the Catalyst 4224. For the details of this configuration procedure, refer to the Cisco CallManager 3.1 online help and to *Configuring Cisco IP Telephony Gateways*, available online at Cisco.com.

With Cisco CallManager 3.0, you must configure each voice port for MGCP on the Catalyst 4224 and then duplicate this configuration in Cisco CallManager Administration.

This section contains the following Catalyst 4224 configuration topics:

For more information on using MGCP with Cisco CallManager 3.0, refer to the Cisco CallManager 3.0 online help and to *Configuring Cisco IP Telephony Gateways*.

Enabling MGCP

To configure the Catalyst 4224 so that it can be controlled by Cisco CallManager Release 3.0 using MGCP, you must identify the primary server and any backup Cisco CallManager servers in case the primary server becomes unavailable. You must also configure each voice gateway as an MGCP gateway in Cisco CallManager, as described in the *Cisco CallManager Administration Guide*. Finally, you must configure the voice ports installed on your gateway.

To enable generic MGCP support on a Cisco voice gateway, enter the following commands from the global configuration mode prompt:

```
Gateway(config)#mgcp
Gateway(config)#mgcp call-agent hostname
```

where *hostname* identifies the Cisco CallManager server (or possibly a generic MGCP call agent).

To enable support for Cisco CallManager within MGCP, enter the following command:

```
Gateway(config)#ccm-manager MGCP
```

Cisco CallManager controls dial-plan-related configuration elements, and they should not be configured in the Cisco voice gateway for MGCP-managed endpoints (those with **application MGCPAPP** in the dial-peer statement). You should *not* configure any of the following elements when using MGCP:

- Destination pattern
- Session target
- Expansion numbers
- Connection PLAR/tie-line/trunk (voice port)
- codec

Note H.323 and MGCP configurations will coexist when you enable MGCP gateway fallback or Survivable Remote Site Telephony (SRST).

Enabling Switchover and Switchback

To identify up to two backup Cisco CallManager servers, enter the following command:

```
Gateway(config)#ccm-manager redundant-host hostname1  hostname2
```

where *hostname1* identifies the first backup Cisco CallManager server using the DNS host name or dotted decimal format, and *hostname2* identifies the second backup Cisco CallManager server.

If you configure one or two backup Cisco CallManager servers, you can control how the gateway behaves if the primary server becomes unavailable at some point and then later becomes available again. This is called *switchback*.

To configure gateway switchback, enter the following command:

```
Gateway(config)#ccm-manager switchback
 {graceful|imm[ediate]|sch[edule-time] hh:mmm|uptime[-delay] minutes}
```

During switchover and switchback, Cisco CallManager maintains active connected calls. Transient calls (calls in progress or on hold without an active voice connection) are torn down. An exception applies for PRI interfaces that MGCP controls, in which case both active and transient calls are torn down during switchover and switchback. Table 5-1 describes each switchback option.

Table 5-1   Switchback Command Options

Switchback Command Option	Function
graceful	The default value. Completes all outstanding calls before returning the gateway to the control of the primary Cisco CallManager server.
immediate	Returns the gateway to the control of the primary Cisco CallManager server without delay, as soon as the network connection to the server is reestablished.
schedule-time *hh:mm*	Returns the gateway to the control of the primary Cisco CallManager server at the specified time, where *hh:mm* is the time according to a 24-hour clock. If the configured schedule time is earlier than the time at which the gateway reestablishes a network connection to the primary server, the switchback will occur at the specified time on the following day.
uptime-delay *minutes*	Returns the gateway to the control of the primary Cisco CallManager server. This occurs when the primary server runs for a specified number of minutes after a network connection is reestablished to the primary server. Permitted values range from 1 to 1440 (1 minute to 24 hours).

You can also manually redirect a Cisco voice gateway to the backup
Cisco CallManager server by entering the following command:

```
Gateway(config)#ccm-manager switchover-to-backup
```

The switchover occurs immediately with a manual redirect. This command does not switch the gateway to the backup Cisco CallManager server if you have the switchback option set to **immediate** and the primary Cisco CallManager server is still running.

To view the current configuration of a Cisco voice gateway, enter the **show ccm-manager** command from privileged EXEC mode. Example 5-1 illustrates a typical display that appears in response to this command.

Example 5-1   Output of the show ccm-manager Command

```
router# sh ccm-manager

MGCP Domain Name: router
Priority         Status                    Host
===============================================================
Primary          Registered                172.20.71.44
First backup     None
Second backup    None

Current active Call Manager:     172.20.71.44
Backhaul/Redundant link port:    2428
Failover Interval:               30 seconds
Keepalive Interval:              15 seconds
Last keepalive sent:             00:54:14 (elapsed time: 00:00:13)
Last MGCP traffic time:          00:54:14 (elapsed time: 00:00:13)
Last switchover time:            None
Switchback mode:                 Graceful
MGCP Fallback mode:              Not Selected
Last MGCP Fallback start time:   00:00:00
Last MGCP Fallback end time:     00:00:00

PRI Backhaul link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:    0
        Recv failures:    0
        Packets xmitted: 0
        Xmit failures:    0
    PRI Ports being backhauled:
        Slot 1, port 1
Configuration Auto-Download Information
=======================================
 No configurations downloaded
```

```
Current state: Automatic Configuration Download feature is disabled
Configuration Error History:
FAX relay mode: cisco-fax-relay
```

Configuring FXS and FXO Analog Ports

You use the same commands to configure both Foreign Exchange Service (FXS) and Foreign Exchange Office (FXO) ports. The gateway recognizes the type of voice interface card that is installed in each voice network module and applies the configuration you enter based on the port position you specify in the command.

To enable FXS or FXO ports with MGCP, enter the following commands:

```
Gateway(config)# dial-peer voice number pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port portnumber
```

To use these commands, replace *number* with a unique numeric ID, and replace *portnumber* with the port identifier in the form *slot#/voice module#/port#*. Use the **application MGCPAPP** command to place the port under control of the Cisco CallManager MGCP call agent.

For example, the following command string configures voice port 0 in voice interface card 1 with MGCP:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port 1/1/0
```

Configuring T1-CAS E&M Emulation

You can use MGCP with the following emulation types:

- E&M Wink Start
- E&M Delay Start

To configure T1-CAS E&M emulation with MGCP using Cisco CallManager Administrator, perform the following steps and configure the route pattern and dial plan:

Step 1   Identify the port number and then enter the configuration information provided by your local carrier, as in the following example:

```
Gateway(config)# controller T1 1/port#
Gateway(config-controller)# framing esf
Gateway(config-controller)# clock source internal
Gateway(config-controller)# linecode b8zs
```

Step 2   Assign time slots to the DS-0 group and identify the emulation type.

You can define each DS-0 group to use FXS, FXO, or E&M, using the following command:

```
Gateway(config-controller)# ds0-group group groupnumber timeslots
<timeslotnumber> type emulationtype
```

Replace *emulationtype* with e&m-wink-start or e&m-delay-dial.

Replace *groupnumber* with the DS-0 group number and replace *timeslotnumber* with the number of DS-0 time slots to allocate to the group. For example, the following command configures the first DS-0 group with one time slot using FXS emulation in loop-start mode:

```
Gateway(config-controller)# ds0-group 0 timeslots 1 type
fxs-loop-start
```

You can configure DS-0 hunt groups by assigning a range of time slots to a DS-0 group and then configuring multiple voice peers with the same destination pattern pointing to multiple voice ports.

For example, the following command assigns 12 time slots to DS-0 group 1:

```
Gateway(config-controller)# ds0-group 1 timeslots 1-12 type
fxs-loop-start
```

Step 3   Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
portnumber:ds0group
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Example 5-2 shows typical use of commands for configuring T-1 CAS E&M Emulation for MGCP.

Example 5-2   T-1 CAS E&M Emulation for MGCP

```
Gateway(config-controller)# ds0-group 0 timeslots 1 type
e&m-wink-start
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application MGCPAPP
Gateway(config-dial-peer)# port 1/0:0
```

Configuring T1/E1 (ISDN-PRI) Ports

To configure an E1/T1 multiflex interface with ISDN-PRI signaling, use the Cisco IOS command line interface to perform the procedures in this section.

Configuring T1 Interfaces

To configure a new T1 interface or to change the configuration of an existing interface, perform the following procedure:

Step 1   Identify the port number and enter line-specific information provided by your local carrier.

a.   Choose the T1/PRI interface to configure:

```
Gateway(config)# controller t1 1/0
```

This example configures a T1 interface in slot 1 and unit 0.

b. Specify which end of the circuit provides clocking:

```
Gateway(config-controller)# clock source line
```

Set the clock source to use internal clocking only for testing the network. Set one end of the T1 line to internal.

c. Specify the framing type:

```
Gateway(config-controller)# framing esf
```

d. Specify the line code format:

```
Gateway(config-controller)# linecode b8zs
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped:

```
Gateway(config-controller)# pri-group timeslots 1-24 service mgcp
```

For multiflex trunk interfaces, you can configure only channel 0.

b. Configure each PRI group as a virtual serial interface:

```
Gateway(config-controller)# interface serial 1/0:23
```

c. Specify the T1 interface and unit number to modify:

```
interface Serial 1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Step 3 Configure the PRI port by entering the following command:

```
Gateway(controller-t1)# pri-group timeslots 1-24 service mgcp
```

Step 4 Bind Layer 3 to the Cisco CallManager for PRI Q.931:

```
Gateway(config-if)# isdn bind-l3 ccm-manager
```

This command backhauls (tunnels) ISDN Layer 3 and above to the Cisco CallManager.

PRI/Q.931 signaling backhaul transports signals (Q.931 and higher layers) for processing from a PRI trunk to a MGCP call agent. The PRI trunk must be physically connected to an MGCP gateway.

The ISDN lower layer information (Q.921 and below) is terminated and processed on the gateway. The Layer 3 information (Q.931 and above) is transported over TCP to the Cisco CallManager (MGCP call agent).

Step 5   Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
<portnumber>:<ds0group>
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Step 6   To view the status of the PRI line, enter the following command:

```
Gateway # show ccm-manager backhaul
```

This command displays information about the status of the TCP backhaul link and the status of any PRI D-channels in the gateway. Example 5-3 shows the type of information the system displays.

Example 5-3   PRI Backhaul Status—T1

```
PRI Backhaul link info:
    Link Protocol:     TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:   0
        Recv failures:   0
        Packets xmitted: 0
        Xmit failures:   0
    PRI Ports being backhauled:
        Slot 1, port 1
        Slot 1, port 0
```

Sample Configuration

The following example shows the overall configuration required to enable MGCP on a T1/PRI line:

```
isdn switch-type primary-5ess
controller T1 1/0
 framing crc4
 linecode hdb3
 pri-group timeslots 1-24 service mgcp
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
 isdn bind-l3 ccm-manager
dial-peer voice 1 pots
 application mgcpapp
 port 1/0:0
```

Configuring E1 Interfaces

Use the following procedure to configure a new E1 interface (balanced or unbalanced) or to change the configuration of an existing interface:

Step 1   Identify the port number and enter line-specific information provided by your local carrier.

a.   Choose the E1/PRI interface to configure by entering the following command from Global configuration mode:

```
Gateway(config)# controller e1 1/0
```

This example configures an E1 interface in slot 1 and unit 0.

b.   Specify the framing type:

```
Gateway(config-controller)# framing crc4
```

c.   Specify the line code format:

```
Gateway(config-controller)# linecode hdb3
```

Step 2   Configure parameters for the voice interface.

    a.   Specify the PRI group and time slots to be mapped:

```
Gateway(config-controller)# pri-group timeslots 1-31 service mgcp
```

    b.   Configure each PRI group as a virtual serial interface:

```
Gateway(config-controller)# interface serial 1/0:15
```

    c.   Specify the E1 interface and unit number to modify:

```
interface Serial1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
```

Step 3   Configure the PRI port by entering the following command:

```
Gateway(controller-e1)# pri-group timeslots 1-31 service mgcp
```

Step 4   Bind Layer 3 to the Cisco CallManager for PRI Q.931:

```
Gateway(config-if)# isdn bind-l3 ccm-manager backhaul q931
```

This command backhauls (tunnels) ISDN Layer 3 and above to the Cisco CallManager.

PRI/Q.931 signaling backhaul transports signals (Q.931 and higher layers) for processing a PRI trunk that is physically connected to an MGCP call agent.

The ISDN lower layer information (Q.921 and below) is terminated and processed on the gateway. The Layer 3 information (Q.923 and above) is transported over TCP to the Cisco CallManager (MGCP call agent).

Step 5   Enable MGCP for the port by entering the following commands:

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# application mgcpapp port
<portnumber>:<ds0group>
```

Replace *portnumber* with the port number on the voice gateway you are configuring and *ds0group* with the DS0 group number.

Step 6   To view the status of the PRI line, enter the following command:

```
Gateway# show ccm-manager backhaul
```

This command displays information about the status of the PRI backhaul link and the status of any PRI D channels in the gateway. The following example shows the type of information the system displays.

Example 5-4   PRI Backhaul Status

```
PRI Backhaul link info:
    Link Protocol:      TCP
    Remote Port Number: 2428
    Remote IP Address:  172.20.71.44
    Current Link State: OPEN
    Statistics:
        Packets recvd:   0
        Recv failures:   0
        Packets xmitted: 0
        Xmit failures:   0
    PRI Ports being backhauled:
        Slot 1, port 1
        Slot 1, port 0
```

Sample Configuration

The following example shows the overall configuration required to enable MGCP on a E1/PRI line:

```
isdn switch-type primary-5ess
controller E1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-31 service mgcp
interface Serial 1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
 isdn bind-l3 ccm-manager backhaul q931
dial-peer voice 1 pots
 application mgcpapp
 port 1/0:0
```

Where to Go Next

At this point, make sure that Cisco CallManager is properly configured to provision the voice gateway and to configure MGCP endpoints or H.323 route patterns as required. Refer to the documentation and online help provided with Cisco CallManager. Refer to the Cisco IOS software configuration guide and command reference publications for more advanced configuration topics. For troubleshooting information, refer to the system error messages and debug command reference publications.

Access these publications on the Documentation CD-ROM that came with your gateway, or on the World Wide Web from the Cisco home page.

H.323 Gateway Configuration

Compared to Media Gateway Control Protocol (MGCP), H.323 requires more configuration on a gateway because the gateway must maintain the dial plan and route pattern. The gateway must have enough information to direct calls to the correct end point, which must be an H.323-capable device.

When using H.323, ensure that you configure Cisco CallManager correctly to provision the Catalyst 4224 as an H.323 gateway, with route patterns configured to route calls to a Catalyst 4224.

To provision the gateway using Cisco CallManager Administration, select the **Add a New Gateway** option from **Device** > **Gateway**. Assign the Gateway Type as H.323 Gateway, and the Device Protocol as H.225.

To configure a route pattern using CCM Administrator, select **Route Plan > Route Pattern** and enter the route pattern. Then, select **Cisco Catalyst 4224 gateway** from a drop-down list, click the **Route this option** button for the Route Option, and check the **Provide Outside Dial Tone** box for Offnet (the default is OnNet).

Perform the following steps to complete H.323 configuration:

Step 1   Identify the port number and enter line-specific information provided by your local carrier, as described in the following sections:

- Configuring T1-CAS Analog Emulation (H.323), page 5-19
- ISDN BRI Configuration (H.323), page 5-24
- T1/E1 Configuration (H.323), page 5-31
- E&M Trunk Line Configuration (H.323), page 5-35

Step 2   Configure parameters for the voice interface you are using, as described in the section referred to above.

Step 3   Configure H.323 endpoints connected to the Catalyst 4224 voice ports.

To configure plain old telephone service (POTS) dial peers, use the following command strings:

```
c4224(config)# dial-peer voice number pots
c4224(config-dial-peer)# destination-pattern endpoint#
c4224(config-dial-peer)# port 1/portnumber:DS0groupnumber
```

Replace:

- *number* with a unique numeric identifier for each dial peer
- *endpoint#* with the E.164 telephone extension of the POTS dial peer
- *portnumber* with 0 or 1, depending on which T1 port you are using
- *DS0groupnumber* with a numeric digit from 0 to 23 for each DS-0 group you are configuring

For example, the following commands could be used to route all calls with the prefix 222 to the DS-0 hunt group 1 of controller T1 1/0:

```
c4224(config)# dial-peer voice 222 pots
c4224(config-dial-peer)# destination-pattern 222....
c4224(config-dial-peer)# port 1/0:1
c4224(config-dial-peer)# prefix 222
```

The prefix command at the end is required to replace the digits that the Catalyst 4224 strips off from the dialed digit string based on the wildcard destination pattern.

Step 4   Configure H.323 endpoints connected to the Catalyst 4224 Ethernet port.

To configure H.323 endpoints, use the following command strings:

```
c4224(config)# dial-peer voice number voip
c4224(config-dial-peer)# destination-pattern endpoint#
c4224(config-dial-peer)# session target {ipv4:ipaddress|dns:hostname}
c4224(config-dial-peer)# codec codecid
```

Replace:

- *number* with a unique numeric identifier for each dial peer

- *endpointnumber* with the telephone extension of the dial peer

- *ipaddress* or *hostname* with the IP address or Domain Name System (DNS) host name of the VoIP dial peer

If you use the IP address, it must be preceded by the parameter **ipv4**. If you use the DNS host name, this must be preceded by the parameter **dns**, and the host name must resolve correctly to the IP address of the target. Finally, you must identify the coder-decoder (codec) used by the Voice over IP (VoIP) dial peer.

For example, the following commands assign extension 2001 to the IP device with the network address 192.168.100.1:

```
c4224(config)# dial-peer voice 1 voip
c4224(config-dial-peer)# destination-pattern 2001
c4224(config-dial-peer)# session target ipv4:192.168.100.1
c4224(config-dial-peer)# codec g711ulaw
```

Step 5   Direct calls using wildcard destination patterns, as needed.

You can use wildcard destination patterns to simplify your dial plan configuration. For instance, you can direct all incoming calls starting with specific digits, such as 525, to a Cisco CallManager configured as an H.323 endpoint. You might direct all calls starting with a 9 to voice ports connected to the Public Switched Telephone Network (PSTN), or direct all calls beginning with an 8 to a private branch exchange (PBX).

```
c4224(config-dial-peer)# destination-pattern pattern ....
```

For example, the following command directs all calls starting with 525 to a Cisco CallManager with the DNS host name CCM30:

```
c4224(config-dial-peer)# destination-pattern 525....
c4224(config-dial-peer)# session target dns:CCM30
```

The number of digits that you substitute for *pattern* plus the number of periods in the wildcard (...) must match the total number of digits configured for use by the Catalyst 4224 in Cisco CallManager Administration. Also, remember that the numbers that you substitute for *pattern* are removed by the Catalyst 4224. When the call is forwarded to the destination number, only the digits in the position of the wildcard pattern (...) are received by the destination endpoint. If you want to replace the digits that are stripped off (or add a different set of digits), use the **prefix** command.

Step 6   Complete and save the configuration by entering the following commands:

```
c4224# line con 0
c4224# transport input none
c4224# line aux 0
c4224# line vty 0 4
c4224# login
c4224# no scheduler allocate
c4224# end
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

Configuring T1-CAS Analog Emulation (H.323)

You can connect the T1-CAS (channel-associated signaling) port on a Catalyst 4224 to one of the following:

- The PSTN using Foreign Exchange Office (FXO) emulation
- A T1 channel bank using Foreign Exchange Station (FXS) emulation
- A PBX with a trunk (tie) line using Ear and Mouth (E&M) emulation

To configure T1-CAS analog emulation with H.323 T1, perform the following steps. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1 Identify the port number and then enter the configuration information provided by your telco, as in the following example:

```
c4224(config)# controller T1 1/port#
c4224(config-controller)# framing esf
c4224(config-controller)# clock source internal
c4224(config-controller)# linecode b8zs
```

Step 2 Assign time slots to the DS-0 group and identify the emulation type.

You can define each DS-0 group to use FXS, FXO, or E&M, using the following command:

```
c4224(config-controller)# dso-group group groupnumber
timeslots timeslotnumber type emulationtype
```

Replace:

- *groupnumber* with the DS-0 group number
- *timeslotnumber* with the number of DS-0 time slots to allocate to the group.
- *emulationtype* with one of the modes described in Table 5-2.

Table 5-2   T-1 Emulation Types

Emulation Type	Function
fxs-loop-start	Uses FXS emulation in loop-start mode.
fxs-ground-start	Uses FXS emulation in ground-start mode.
fxo-loop-start	Uses FXO emulation in loop-start mode.
fxo-ground-start	Uses FXO emulation in ground-start mode.
e&m-immediate-start	Uses E&M emulation in immediate-start mode.
e&m-wink-start	Uses E&M emulation in wink-start mode.
e&m-delay-dial	Uses E&M emulation in immediate-delay dial mode.

For example, the following command configures the first DS-0 group with one time slot using FXS emulation in loop-start mode:

```
c4224(config-controller)# dso-group 0 timeslots 1 type fxs-loop-start
```

You can configure DS-0 hunt groups by assigning a range of time slots to a DS-0 group, and then configuring multiple voice peers with the same destination pattern pointing to multiple voice ports.

For example, the following command assigns 12 time slots to DS-0 group 1:

```
c4224(config-controller)# dso-group 1 timeslots 1-12 type
fxs-loop-start
```

Note   After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Managing Input Gain for Cisco IP Voice Applications

When using the FXO ports on a Catalyst 4224, set the input gain to a value provides adequate audio quality for Cisco IP voice applications or the Cisco 7960 IP Phone. Input gain values higher than 12 may cause dual tone multifrequency (DTMF) recognition difficulties.

Cisco recommends that you use the Cisco CallManager graphical user interface to set the input gain. However, you can also enter the following series of commands from the Cisco IOS command line to set input gain:

```
c4224# configure terminal
c4224(config)# voice-port x/x/x input gain value
```

Permitted entries for *value* are from -6 to 14.

FXS Emulation Example

By connecting the T1-CAS port on a Catalyst 4224 to a T1 channel bank using FXS emulation, you can achieve high port density when connecting POTS and VoIP endpoints. You can configure the dial plan for this configuration by treating Cisco CallManager as the only H.323 endpoint, or by configuring H.323

endpoints on a Catalyst 4224. If you configure Cisco CallManager as an H.323 endpoint, you must use Cisco CallManager Administration to define the route patterns required to route calls to the Catalyst 4224.

The following example illustrates how to configure a single DS-0 group. Repeat the relevant commands to configure additional groups. This example is for a scenario in which all of the POTS devices connected to a T1 channel bank are configured with a destination number beginning with 526. In this example, Cisco CallManager has the host name CCM30 and is configured as an H.323 endpoint that manages all the telephones and other devices on the IP network. The devices on the IP network have numbers beginning with 525.

```
c4224(config)# interface FastEthernet5/0
c4224(config)# ipaddress 172.20.71.48 255.255.255.0
c4224(config)# no ip directed-broadcast
c4224(config)# no keepalive
c4224(config)# duplex auto
c4224(config)# speed 10

c4224(config)# controller T1 1/0
c4224(config-controller)# framing esf
c4224(config-controller)# clock source internal
c4224(config-controller)# linecode b8zs
c4224(config-controller)# dso-group 0 timeslots 1 type fxo-loop-start

c4224(config)# dial-peer voice 1 pots
c4224(config-dial-peer)# destination-pattern 526....
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 525....
c4224(config-dial-peer)# session target dns:CCM30
c4224(config-dial-peer)# codec g711ulaw

c4224# line con 0
c4224# transport input none
c4224# line aux 0
c4224# line vty 0 4
c4224# login
c4224# no scheduler allocate
c4224# end
c4224# copy running-config startup-config
Building configuration. . .
[OK]
c4224#
```

FXO Emulation Example

To use FXO emulation to connect the T1-CAS port to the PSTN, you must have Direct Inward Dialing (DID) enabled on incoming DS-0 groups. DID allows the gateway or Cisco CallManager to identify the extension to which each call on an incoming DS-0 group is directed. Because DID only works on incoming connections, you must have separate DS-0 groups allocated for incoming and outgoing calls. To configure the gateway to accept DID information, enter the following command:

```
c4224(config-dial-peer)# direct-inward-dial
```

The first and last parts of the configuration are the same as for the FXO example. However, you must configure your DS-0 groups for FXS by changing the emulation type and enabling direct inward dialing (DID). Then, enter the destination patterns required for routing voice calls to and from the PSTN. The commands required to make these changes are shown below:

```
c4224(config-controller)# dso-group 0 timeslots 1 type fxo-loop-start

c4224(config)# dial-peer voice 1 pots
c4224c4224(config-dial-peer)# direct-inward-dial
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 9.......
```

E&M Emulation Example

To connect the T1-CAS port to a trunk (tie) line using E&M emulation, you can enable one of the following modes:

- E&M immediate start
- E&M wink start
- E&M delay dial

The first and last parts of the configuration are the same as for the FXO example. However, you must configure your DS-0 groups for E&M by changing the emulation type. Then, enter the destination patterns required for routing voice

calls to and from the PBX to which the gateway is connected. The commands required to make these changes are shown below (all the extensions on the PBX begin with the prefix 625):

```
c4224(config-controller)# dso-group 0 timeslots 1 type
e&m-immediate-start
c4224(config)# dial-peer voice 1 pots
c4224(config-dial-peer)# port 1/0:0
c4224(config-dial-peer)# destination-pattern 625....
```

ISDN BRI Configuration (H.323)

To configure an ISDN BRI interface, perform the following steps. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1   Enter an ISDN switch type by entering the following command in global configuration mode:

```
c4224(config)# isdn switch-type switch-type
```

See Table 5-3 for a list of ISDN switch types.

✎

Note   Switch types configured in interface configuration mode override this setting for the configured interface.

Table 5-3   **ISDN Switch Types**

Region	ISDN Switch Type	Description
Australia	basic-ts013	Australian TS013 switches
Europe	basic-1tr6	German 1TR6 ISDN switches
	basic-nwnet3	Norwegian NET3 ISDN switches (phase 1)
	basic-net3	NET3 ISDN switches (UK and others)
	vn2	French VN2 ISDN switches
	vn3	French VN3 ISDN switches
Japan	ntt	Japanese NTT ISDN switches
New Zealand	basic-nznet3	New Zealand NET3 switches
North America	basic-5ess	AT&T basic rate switches
	basic-dms100	NT DMS-100 basic rate switches
	basic-ni1l	National ISDN-1 switches

Step 2   Assign the switch type to the interface by entering the following commands. The following example assigns the switch type basic-5ess:

```
c4224(config)# interface bri 0/0
c4224(config-if)# isdn switch-type basic-5ess
```

For details about configuring Cisco CallManager, refer to the *Cisco CallManager Administration Guide*.

Note   After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Configuring ISDN BRI Lines

Before using a Catalyst 4224 with an ISDN BRI interface, you must order a correctly configured ISDN BRI line from your local telecommunications service provider.

The ordering process varies from provider to provider and from country to country; however, here are some general guidelines:

- Ask for two channels to be called by one number.
- Ask for delivery of calling line identification, also known as caller ID or Automatic Number Identification (ANI).

ISDN BRI Provisioning by Switch Type

ISDN BRI provisioning refers to the types of services provided by the ISDN BRI line. Although provisioning is performed by your ISDN BRI service provider, you must tell the provider what you want.

Table 5-4 lists the provisioning you should order for your Catalyst 4224 for each switch type.

Table 5-4   ISDN Provisioning by Switch Type

Switch Type	Provisioning
5ESS Custom BRI	For voice only: • (Use these values only if you have an ISDN telephone connected.) • 2 B channels for voice • Multipoint • Terminal type = D • 2 directory numbers assigned by service provider • 2 SPID[1]s required, assigned by service provider • MTERM = 2 • Number of call appearances = 1 • Display = No • Ringing/idle call appearances = 1 • Autohold = no • Onetouch = no • Request delivery of calling line ID on Centrex lines • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2

Table 5-4   ISDN Provisioning by Switch Type (continued)

Switch Type	Provisioning
5ESS National ISDN (NI-1) BRI	• Terminal type = A • 2 B channels for voice • 2 directory numbers assigned by service provider • 2 SPIDs required, assigned by service provider • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2
DMS-100 BRI	• 2 B channels for voice • 2 directory numbers assigned by service provider • 2 SPIDs required, assigned by service provider • Functional signaling • Dynamic TEI[2] assignment • Maximum number of keys = 64 • Release key = no, or key number = no • Ringing indicator = no • EKTS = no • PVC = 2 • Request delivery of calling line ID on Centrex lines • Set speed for ISDN calls to 56 kbps outside local exchange Directory number 1 can hunt to directory number 2

1. Service profile identifier

2. Terminal endpoint identifier

Defining ISDN Service Profile Identifiers

Some service providers assign service profile identifiers (SPIDs) to define the services to which an ISDN device subscribes. If your service provider requires SPIDs, your ISDN device cannot place or receive calls until it sends a valid SPID to the service provider when initializing the connection.

A SPID is usually a seven-digit telephone number plus some optional numbers, but service providers might use different numbering schemes. SPIDs have significance at the local access ISDN interface only; remote Catalyst 4224s are never sent the SPID.

Currently, only DMS-100 and NI-1 switch types require SPIDs. Two SPIDs are assigned for the DMS-100 switch type, one for each B channel. The AT&T 5ESS switch type might support SPIDs, but Cisco recommends that you set up that ISDN service without SPIDs.

If your service provider assigns you SPIDs, you must define these SPIDs on the Catalyst 4224. To define SPIDs and the local directory number (LDN) on the gateway for both ISDN BRI B channels, use the following **isdn spid** commands:

```
c4224 (config-if)# isdn spid1 spid-number [ldn]
c4224 (config-if)# isdn spid2 spid-number [ldn]
```

Note Although the LDN is an optional parameter in the command, you might need to enter it so the gateway can answer calls made to the second directory number.

For further information on configuring ISDN, refer to the chapters "Configuring ISDN" and "Configuring DDR" in the *Wide-Area Networking Configuration Guide*.

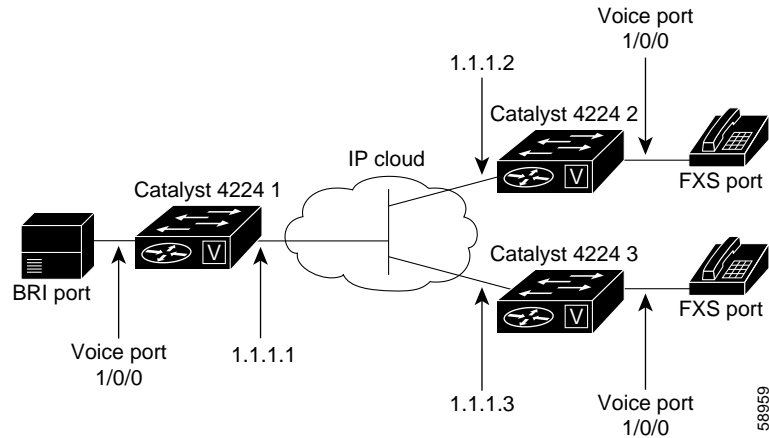# BRI Direct-Inward Dialing Configuration

The example illustrated in Figure 5-1 shows how to configure a BRI port for direct-inward dialing (DID). This configuration allows the called number information from the ISDN Q.931 setup message to be used for routing on an ISDN line.

In this example, a call comes in to Catalyst 4224 1 on the BRI port. The DID information allows the Catalyst 4224 to route the call based on the called number. If the called number is 2xxx, the call is routed to Catalyst 4224 2000; if the called number is 3xxx, the call is routed to Catalyst 4224 3000.

Figure 5-1 illustrates the topology of this connection example.

Figure 5-1   Configuring DID on a BRI Port

Gateway 1 Configuration

```
dial-peer voice 1 pots
 port 1/0/0
 destination-pattern 1...
 direct-inward-dial
dial-peer voice 2 voip
 session target ipv4:1.1.1.2
 destination-pattern 2...
dial-peer voice 3 voip
 session target ipv4:1.1.1.3
 destination-pattern 3...
```

Gateway 2 Configuration

```
dial-peer voice 1 pots
 port 1/0/0
 destination-pattern 2000
```

T1/E1 Configuration (H.323)

To configure an ISDN PRI interface or T1/E1 multiflex trunk interface on your Catalyst 4224, use configuration mode.

This section contains the following topics:

- Configuring T1 Interfaces, page 5-31
- T1/PRI Configuration Example, page 5-33
- Configuring E1 Interfaces, page 5-33
- E1/PRI Configuration Example, page 5-34

Configuring T1 Interfaces

Use the following procedure to configure a new T1 interface or to change the configuration of an existing interface. After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

Step 1   Identify the port number and enter line-specific information provided by your local carrier.

a.   Select the T1/PRI interface to configure.

```
c4224(config)# controller t1 1/0
```

This example configures a T1 interface in slot 1 and unit 0.

b. Specify which end of the circuit provides clocking.

```
c4224(config-controller)# clock source line
```

The clock source should be set to use internal clocking only for testing the network or if the full T1 line is used as the channel group. Only one end of the T1 line should be set to internal.

c. Specify the framing type.

```
c4224(config-controller)# framing esf
```

d. Specify the line code format.

```
c4224(config-controller)# linecode b8zs
```

Step 2 Configure parameters for the voice interface.

a. Specify the PRI group and time slots to be mapped.

```
c4224(config-controller)# pri-group timeslots 1-24
```

For multiflex trunk interfaces, only channel 0 can be configured.

b. Configure each pri-group as a virtual serial interface.

```
c4224(config-controller)# interface serial 1/0:15
```

c. Specify the T1 interface, unit number, and channel group to modify, as in the following example input:

```
interface Serial pri-group
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Note After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

T1/PRI Configuration Example

```
isdn switch-type primary-5ess
isdn voice-call-failure 0
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Configuring E1 Interfaces

Use the following procedure to configure a new E1 or CE1/PRI interface (balanced or unbalanced) or to change the configuration of an existing interface.

Step 1   Identify the port number and enter line-specific information provided by your local carrier.

   a.   Select the CE1/PRI interface to configure by entering the following command from global configuration mode. This example configures an E1 interface in slot 1 and unit 0.

```
c4224(config)# controller e1 1/0
```

   b.   Specify the framing type.

```
c4224(config-controller)# framing crc4
```

   c.   Specify the line code format.

```
c4224(config-controller)# linecode hdb3
```

Step 2   Configure parameters for the voice interface.

   a.   Specify the PRI group and time slots to be mapped.

```
c4224(config-controller)# pri-group timeslots 1-31
```

b. Configure each channel group as a virtual serial interface.

```
c4224(config-controller)# interface serial 1/0:31
```

c. Specify the E1 interface, unit number, and channel group to modify, as in the following example:

```
interface Serial1/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

Note   After completing these steps, configure the route pattern and dial plan and save your configuration, as described in "H.323 Gateway Configuration" section on page 5-16.

E1/PRI Configuration Example

```
isdn switch-type primary-5ess
isdn voice-call-failure 0
controller E1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-31
interface Serial 1/0:15
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice
 no cdp enable
```

E&M Trunk Line Configuration (H.323)

The section illustrates how to configure VoIP to link PBX users with E&M trunk lines.

This section contains the following topics:

Scenario

Suppose that a company wants to connect two offices: one in San Jose, California, and the other in Salt Lake City, Utah. Each office has an internal telephone network using a PBX connected to the voice network with an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and ImmediateStart signaling. Each E&M interface connects to the gateway using two voice interface connections. Users in San Jose dial 8-569 and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial 4-527 and then the extension number to reach a destination in San Jose.

Figure 5-2 illustrates the topology of this configuration.

Figure 5-2   Linking PBX Users with E&M Trunk Lines
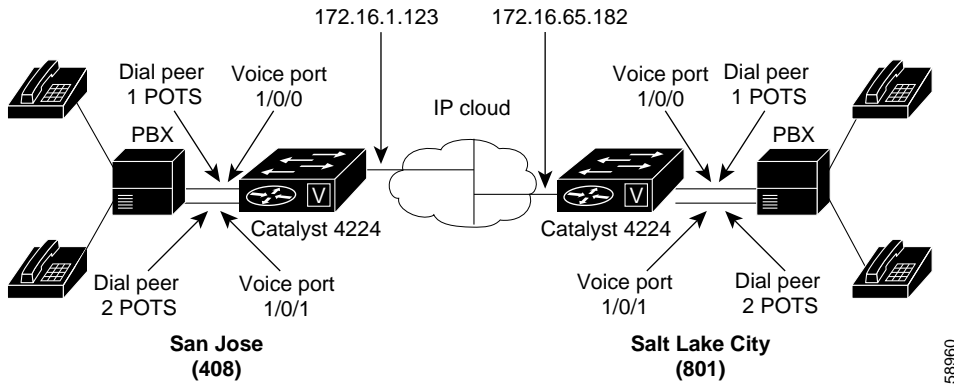
Note   This scenario assumes that the company already has established a working IP connection between its two remote offices.

Handling Incoming Caller ID Digits on an E&M Port

When using an H.323 T1-CAS E&M port on the Catalyst 4224, Incoming Caller ID Digits may not be processed correctly by Cisco CRA, such as Cisco IP Auto-Attendant.

Depending on the T1-CAS line provisioning, incoming dialed number identification service (DNIS) digits received by the Catalyst 4224 after its first wink to the Central Office (CO) are treated as user-entered digits and are sent to the remote endpoint as out-of-band DTMF digits. If the remote endpoint is a Cisco CRA application, the out-of-band digits will be interpreted as a user entry and will change the application response.

There are two ways to handle this situation:

- Ask the T1 service provider to stop sending DNIS digits.

- Configure an IP phone with the same directory number as the incoming DNIS, then modify the "forward all" selection for this phone so that it sends the incoming call to the desired destination.

Gateway San Jose Configuration

```
hostname sanjose
!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern +527....
 port 1/0/0
!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern +527....
 port 1/0/1
!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern +569....
 session target ipv4:172.16.65.182
!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
voice-port 1/0/1
 signal immediate
 operation 4-wire
 type 2
!Configure the serial interface
interface serial 0/0
 description serial interface type dce (provides clock)
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Gateway Salt Lake City Configuration

```
hostname saltlake
!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern +569....
 port 1/0/0
!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern +569....
 port 1/0/1
!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern +527....
 session target ipv4:172.16.1.123
```

```
!Configure the E&M interface
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
voice-port 1/0/0
 signal immediate
 operation 4-wire
 type 2
!Configure the serial interface
interface serial 0/0
 description serial interface type dte
 ip address 172.16.65.182
 no shutdown
```

Note   The PBXs should be configured to pass all DTMF signals to the gateway. Cisco recommends that you do not configure "store-and-forward" tone.

If you change the gain or the telephony port, make sure that the telephony port still accepts DTMF signals.

Note   After completing E&M configuration, configure the route pattern and dial plan and save your configuration, as described in the "H.323 Gateway Configuration" section on page 5-16.

At this point in the configuration process, make sure that Cisco CallManager is properly configured to provision the Catalyst 4224 and to configure MGCP endpoints or H.323 route patterns, as required. Refer to the *Cisco CallManager Administration Guide* or to the online help for the application.

Refer to the Cisco IOS configuration guides and command references for details about specific VoIP commands and options.