

Configuring Voice over IP

This chapter explains how to configure the Cisco Voice Gateway 200 (VG200) software for routing Voice over IP (VoIP) for the different hardware modules available. You will only need to complete the software configuration required for the hardware modules that you have installed. This chapter includes the following sections:

- Configuring Voice Interfaces, page 3-2
- Configuring Voice Ports with MGCP, page 3-4
- Using MGCP with Cisco CallManager, page 3-4
- MGCP Settings, page 3-7
- H.323 Gateway Configuration, page 3-10
- Configuring T1-CAS Analog Emulation (H.323), page 3-13
- ISDN BRI Configuration (H.323), page 3-18
- T1/E1 Configuration (H.323), page 3-24
- E&M Trunk Line Configuration (H.323), page 3-27
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Configuring Voice Interfaces

You need both a voice network module and a voice interface card for a voice connection. For information about installing these components in a Cisco VG200 gateway, refer to the *Hardware Configuration Guide for the Cisco VG200*. For an explanation about how these components work and how they are identified, refer to Chapter 2, "Basic Configuration," in this document.

When you start a Cisco VG200 gateway, it automatically detects the voice network modules and voice interface cards that have been installed. The first time you use a Cisco VG200 gateway, you will need to configure each voice port that you want to enable. After configuring each port, if you replace a card or module with a card or module of the same type, the gateway will recognize the new hardware component and use the previous configuration settings.

You must configure the Cisco VG200 software for each new interface, to change the configuration of an existing interface.

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

To configure a voice interface, you must use configuration mode (manual configuration). In this mode, you can enter Cisco IOS command-line interface (CLI) commands at the gateway prompt.

Perform the following steps to enter configuration mode (manual configuration):

- Step 1 Connect a terminal or a PC running terminal emulation software to the console port on the gateway. Refer to the installation chapter of your gateway installation and configuration guide for instructions.
- Step 2 Power on the gateway. If the current configuration is no longer valid, after about one minute you will see the following prompt:

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 enter normal operating mode automatically.



Step 4 After a few seconds, you see the user EXEC prompt (VG200>). Enter **enable** and the password to enter privileged EXEC mode, as follows:

VG200> **enable** Password: <*password*>

Configuration changes can be made only in privileged EXEC (enable) mode. When you enter privileged EXEC mode, the prompt changes to the host name followed by a pound sign (#), such as VG200#.

- Step 5 Enter the commands required to configure the voice interface cards installed in the VG200, as described in the following sections:
 - Configuring Voice Ports with MGCP, page 3-4
 - H.323 Gateway Configuration, page 3-10
- **Step 6** If you wish to enable MGCP for a voice interface, refer to the "Using MGCP with Cisco CallManager" section on page 3-4.
- **Step 7** When you finish configuring the voice interfaces, exit configuration mode and return to the enable prompt by pressing **Ctrl-Z**.
- Step 8 To see the current operating configuration, including any changes you just made, enter the **show running-config** command, as follows:

VG200# show running-config

To see the configuration currently stored in NVRAM, enter the **show startup-config** command at the enable prompt, as follows:

VG200# show startup-config

Step 9 The results of the show running-config and show startup-config commands differ from each other if you have made changes to the configuration, but have not yet written them to non-volatile random access memory (NVRAM). To write your changes to NVRAM and make them permanent, enter the copy running-config startup-config command at the enable prompt, as follows:

VG200# copy running-config startup-config Building configuration. . . [OK] VG200#

The gateway is now configured to boot in the new configuration.

Configuring Voice Ports with MGCP

You can use the Media Gateway Control Protocol (MGCP) with Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports on the VG200. This configuration is much simpler than H.323 configuration and takes full advantage of Cisco CallManager administration and redundant call agent features.

You use the same commands to configure FXS and 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:

```
VG200(config)# dial-peer voice <number> pots
VG200(config-dial-peer)# application MGCPAPP
VG200(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#*. As described in Chapter 1, "Using the Cisco Voice Gateway 200," voice ports are always installed in slot 1 of the Cisco VG200 gateway, and slots and ports begin at 0. 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:

```
VG200(config)# dial-peer voice 1 pots
VG200(config-dial-peer)# application MGCPAPP
VG200(config-dial-peer)# port 1/1/0
```

Using MGCP with Cisco CallManager

To configure the Cisco VG200 so it can be controlled by Cisco CallManager using MGCP, you must identify the primary and any backup Cisco CallManager servers that you wish to use in case the primary server becomes unavailable. You must also configure the Cisco VG200 as an MGCP gateway in Cisco CallManager, as described in the *Cisco CallManager Administration Guide*.



To enable generic MGCP support on a Cisco VG200 gateway, enter the following commands from the global configuration mode prompt:

VG200(config)#mgcp VG200(config)#mgcp call-agent hostname

Where *hostname* identifies the Cisco CallManager server (or possibly a generic MGCP call agent). You should use this host name to identify this gateway when configuring Cisco CallManager.

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

VG200(config)#ccm-manager MGCP

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

VG200(config)#ccm-manager redundant-host hostname1 hostname2

Where *hostname1* identifies the first backup Cisco CallManager server in dotted decimal format, and *hostname2* identifies the second backup Cisco CallManager server.

Note that all dial-plan related configuration elements are controlled by Cisco CallManager, and should not be configured in the Cisco VG200 gateway for MGCP-managed endpoints (those with **application MGCPAPP** in the dial-peer statement). You should *not* configure any of the following:

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

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:

VG200(config)#ccm-manager switchback {graceful|imm[ediate]|sch[edule-time] hh:mmm|uptime[-delay] minutes} During switchover and switchback, active connected calls are maintained. Transient calls (calls in progress or on hold without an active voice connection) are torn down.

Table 3-1 describes each of the switchback options.

Switchback Command Option	Function
graceful	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 <i>hh:mm</i>	Returns the gateway to the control of the primary Cisco CallManager server at the specified time, where <i>hh:mm</i> 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 when the primary server runs for a specified number of minutes after a network connection is reestablished to the primary server. The range of permitted values is from 1 to 1440 (1 minute to 24 hours).

Table 3-1 Switchback Command Options



You can also manually redirect a Cisco VG200 gateway to the backup Cisco CallManager server by entering the following command:

VG200(config)#ccm-manager switchover-to-backup

In this case, the switchover will occur immediately. Note that this command will 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 VG200 gateway, enter the **show ccm-manager** command from global configuration mode. Figure 3-1 illustrates a typical display that appears in response to this command.

Figure 3-1 Output of the show ccm-manager Command

Total number of host: 3 Priority Status	Host
Primary Registered First backup Backup ready Second backup Idle Current active Call Manager: Current backup Call Manager: Redundant link port: Failover Interval: Keepalive Interval: Last keepalive check: Last MGCP traffic time: Last switchover time: Switchback mode:	172.20.71.30 172.20.71.26 172.20.71.26 172.20.71.30 172.20.71.30 172.20.71.26 2428 30 seconds 15 seconds 00:00:00 00:00:32 None Immediate

MGCP Settings

The **show mgcp** command displays the option settings that determine how the gateway behaves within the MGCP system. The only options you need to set from the Cisco VG200 CLI are the following:

VG200(config) **MGCP call-agent** *hostname*|*ipaddress* VG200(config)# mgcp dtmf-relay codec all mode out-of-band

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The first of these commands, described earlier in this section, enables MGCP on a Cisco VG200 gateway. The second command enables support for dual-tone multi-frequency (DTMF) relay on the gateway.



Make sure that you use the MGCP configuration command for enabling DTMF relay. If this command is not configured, DTMF tones will be not be regenerated correctly on the remote endpoint.

You can view the current settings for the MGCP parameters by entering the **show mgcp** command from global configuration mode. Figure 3-2 illustrates the typical output from this command.



router#sh ccm-manager MGCP Domain Name: router.cisco).com	
Priority Status	Host	
Primary Down	144.254.219.118	
First backup Registered	172.20.71.47	
Second backup Undefined		
Current active Call Manager:	172.20.71.47	
Current backup Call Manager:	None	
Redundant link port:	2428	
Failover Interval:	30 seconds	
Keepalive Interval:	15 seconds	
Last keepalive sent:	02:01:05 (elapsed time: 00:00:10)	
Last MGCP traffic time:	02:01:05 (elapsed time: 00:00:10)	
Last switchover time:	01:57:48 from (144.254.219.118)	
Switchback mode:	Not selected (Default:Graceful)	33
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Table 3-2 summarizes the meaning of these settings. You can display this information by issuing the **MGCP** ? command from privileged EXEC mode.



Tabl	le 3	3-2	MC	GCP	Set	tings

MGCP Setting	Purpose
1025-65535	The UDP port number used with MGCP. The default is 2427.
block-newcalls	Takes down active connections in an orderly way.
call-agent	The address of the call agent.
codec type	The CODEC rate to be attempted for MGCP-controlled connections.
default-package	The default package capability supported by MGCP.
dtmf-relay	Identifies if DTMF relay is enabled.
ip-tos low delay high throughput high reliability low cost	The IP type of services for MGCP-controlled connections.
max-waiting-delay	Maximum waiting delay(MWD), which prevents restart avalanches.
modem	MGCP modem parameters.
package-capability	The package capabilities supported by MGCP.
playout mode	The jitter buffer packet size for MGCP-controlled connections.
quality-threshold	Voice quality-related threshold values.
request timeout request retries	Configuration for MGCP requests sent by this gateway.
restart-delay	The restart delay timer value.
sdp	Specify Session Description Protocol (SDP) operation for MGCP.
vad	Identifies if voice activity detection (silence suppression) is enabled for MGCP-controlled endpoints.

H.323 Gateway Configuration

Compared to MGCP, H.323 requires more configuration on the 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, make sure that you configure Cisco CallManager correctly to provision the Cisco VG200 as an H.323 gateway, with route patterns configured to route calls to a Cisco VG200 gateway.

To provision the gateway using Cisco CallManager Administrator, 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 Cisco CallManager Administrator, select **Dial Plan/Route Pattern** and enter the route pattern. Then select the Cisco VG200 gateway, set the Route Option to Route this pattern, and specify the Network Location to be OnNet or OffNet.

To configure a specific interface or line signalling type, refer to the appropriate section provided later in this chapter. In general, you need to 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 decribed in the following sections:
 - Configuring T1-CAS Analog Emulation (H.323), page 3-13
 - ISDN BRI Configuration (H.323), page 3-18
 - T1/E1 Configuration (H.323), page 3-24
 - E&M Trunk Line Configuration (H.323), page 3-27
- **Step 2** Configure parameters for the voice interface you are using, as decribed in the section referred to above.
- Step 3 Configure H.323 endpoints connected to the Cisco VG200 voice ports.

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

```
VG200(config)# dial-peer voice <number> pots
VG200(config-dial-peer)# destination-pattern endpoint#
VG200(config-dial-peer)# port 1/<portnumber>:<DS0groupnumber>
```



Replace *<number>* with a unique numeric identifier for each dial peer, and replace *endpoint#* with the E.164 telephone extension of the POTS dial peer. Replace *<portnumber>* with 0 or 1 depending on which T1 port you are using. Replace *<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:

```
VG200(config)# dial-peer voice 222 pots
VG200(config-dial-peer)# destination-pattern 222....
VG200(config-dial-peer)# port 1/0:1
VG200(config-dial-peer)# prefix 222
```

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

Step 4 Configure H.323 endpoints connected to the Cisco VG200 Ethernet port.

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

```
VG200(config)# dial-peer voice <number> voip
VG200(config-dial-peer)# destination-pattern <endpointnumber>
VG200(config-dial-peer)# session target {ipv4:ipaddress|dns:hostname}
VG200(config-dial-peer)# codec codecid
```

Replace *<number>* with a unique numeric identifier for each dial peer, and replace *<endpointnumber>* with the telephone extension of the dial peer. Replace *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, identify the CODEC used by the VoIP dial peer.

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

```
VG200(config)# dial-peer voice 1 voip
VG200(config-dial-peer)# destination-pattern 2001
VG200(config-dial-peer)# session target ipv4:192.168.100.1
VG200(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 PSTN, or direct all calls beginning with an "8" to a PBX.

VG200(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:

VG200(config-dial-peer)# destination-pattern 525.... VG200(config-dial-peer)# session target dns:CCM30

The number of digits in 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 gateway in Cisco CallManager Administrator. Also, keep in mind that the numbers that you substitute for *pattern* are removed by the gateway. When the call is forwarded to the destination number, only the digits in the position of the wildcard pattern (...) will be received by the destination endpoint. If you want to replace the digits that are stripped off or (or add a different set of digits), use the **prefix** command.

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

```
VG200# line con 0
VG200# transport input none
VG200# line aux 0
VG200# line vty 0 4
VG200# login
VG200# no scheduler allocate
VG200# end
VG200# copy running-config startup-config
Building configuration. . .
[OK]
VG200#
```



Configuring T1-CAS Analog Emulation (H.323)

As described in Chapter 1, "Using the Cisco Voice Gateway 200," you can connect the T1-CAS port on a Cisco VG200 gateway to the PSTN using FXO emulation, to a T1 channel bank using FXS emulation, or to a PBX with a trunk (tie) line using E&M emulation. To configure T1-CAS analog emulation with H.323 T1, perform the following step. 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 3-10.

Step 1 Identify the T1 port number and enter the information provided by your local carrier.

However you configure the T1 port, you first identify the port number and then enter the configuration information provided by your telco, as in the following example:

```
VG200(config)# controller T1 1/port#
VG200(config-controller)# framing esf
VG200(config-controller)# clock source internal
VG200(config-controller)# linecode b8zs
```

Step 2 Assign timeslots 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:

VG200(config-controller)# dso-group group<groupnumber> timeslots <timeslotnumber> type emulationtype

Replace *emulationtype* with one of the modes described in Table 3-3.

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.

Table 3-3 T-1 Emulation Types

Emulation Type	Function	
fxo-ground-start	Uses FXO emulation in ground-st 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.	

Table 3-3 T-1 Emulation Types (continued)

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

VG200(config-controller)# dso-group 0 timeslots 1 type fxs-loop-start

As mentioned earlier, you can configure DS-0 hunt groups by assigning a range of timeslots 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 timeslots to DS-0 group 1:

VG200(config-controller)# dso-group 1 timeslots 1-12 type fxs-loop-start



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 3-10.



Managing Input Gain for Cisco IP Voice Applications

When using the FXO ports on a Cisco VG200, set the input gain greater than 10 to achieve adequate audio quality for use with Cisco IP voice applications or the Cisco IP Phone 7960. Enter the following series of commands from the Cisco IOS command line to set the correct value for input gain:

```
vg200# configure terminal
vg200(config)# voice-port x/x/x input gain <value>
```

Permitted entries for <value> are from -6 to 14. Gain values higher than 12 may cause DTMF recognition difficulties.

FXS Emulation Example

By connecting the T1-CAS port on a Cisco VG200 gateway to a T1 channel bank using FXS emulation, you can achieve high port density when interconnecting POTS and VoIP endpoints. You can configure the dial plan for this configuration by treating the Cisco CallManager as the only H.323 endpoint, or by configuring H.323 endpoints on a Cisco VG200 gateway. If you configure Cisco CallManager as an H.323 endpoint, you must use Cisco CallManager Administrator to define the route patterns required to route calls to the Cisco VG200 gateway.

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. 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, which have numbers beginning with 525.

```
VG200(config)# interface FastEthernet0/0
VG200(config)# ipaddress 172.20.71.48 255.255.255.0
VG200(config)# no ip directed-broadcast
VG200(config)# no keepalive
VG200(config)# duplex auto
VG200(config)# speed 10
VG200(config)# controller T1 1/0
VG200(config-controller)# framing esf
VG200(config-controller)# clock source internal
VG200(config-controller)# linecode b8zs
VG200(config-controller)# dso-group 0 timeslots 1 type fxo-loop-start
VG200(config)# dial-peer voice 1 pots
VG200(config-dial-peer)# destination-pattern 526....
VG200(config-dial-peer)# port 1/0:0
VG200(config-dial-peer)# destination-pattern 525....
VG200(config-dial-peer)# session target dns:CCM30
VG200(config-dial-peer)# codec g711ulaw
VG200# line con 0
VG200# transport input none
VG200# line aux 0
VG200# line vty 0 4
VG200# login
VG200# no scheduler allocate
VG200# end
VG200# copy running-config startup-config
Building configuration. . .
[OK]
VG200#
```

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:

VG200(config-dial-peer)# **direct-inward-dial**

The first part and last part of the configuration is the same as for the FXO example. However, you must configure your DS-0 groups for FXS by changing the emulation type and enabling 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:

VG200(config-controller)#dso-group 0 timeslots 1 type fxo-loop-start

```
VG200(config)# dial-peer voice 1 pots
VG200(config-dial-peer)# direct-inward-dial
VG200(config-dial-peer)# port 1/0:0
VG200(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 part and last part of the configuration is 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, where all the extensions on the PBX begin with the prefix 625:

```
VG200(config-controller)# dso-group 0 timeslots 1 type
e&m-immediate-start
VG200(config)# dial-peer voice 1 pots
VG200(config-dial-peer)# port 1/0:0
VG200(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 3-10.

Step 1 Enter an ISDN switch type by entering the following command from the Global configuration mode prompt:

vg200(config)# **isdn switch-type basic-5ess** See Table 3-4 for a list of ISDN switch types.



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

Step 2 Assign the switch type to the interface by entering the following command:

Enter the Interface configuration mode by entering the following command:

vg200(config)# interface bri 0/0
vg200(config-if)#

You have entered interface configuration mode when the prompt changes to vg200(config-if)#.

vg200(config-if)# isdn switch-type basic-5ess

Table 3-4 ISDN Switch Types

Country	ISDN Switch Type	Description
Australia	basic-ts013	Australian TS013 switches
Europe	basic-1tr6 German 1TR6 ISDN switch	
	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



Country	ISDN Switch Type	Description
North America	basic-5ess	AT&T basic rate switches
	basic-dms100	NT DMS-100 basic rate switches
	basic-nil1	National ISDN-1 switches

Table 3-4 ISDN Switch Types (continued)

For a detailed description about configuring Cisco CallManager, refer to the *Cisco CallManager Administration Guide*.



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 3-10.

Configuring ISDN BRI Lines

Before using a gateway 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 3-5 lists the provisioning you should order for your gateway.

Switch Type	Provisioning	
5ESS Custom BRI	For voice	
	(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 service profile (SPIDs) required, assigned by service provider	
	MTERM = 2	
	Number of cal 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 3-5 ISDN Provisioning by Switch Type



Switch Type	Provisioning		
5ESS National ISDN	Terminal type = A		
(NI-1) BRI	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 terminal endpoint identifier (TEI) 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		

Table 3-5ISDN Provisioning by Switch Type (continued)

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 gateways 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 gateway. To define SPIDs and the local directory number (LDN) on the gateway for both ISDN BRI B channels, use the following **isdn spid** commands:

```
vg200 (config-if)# isdn spid1 spid-number [ldn]
vg200 (config-if)# isdn spid2 spid-number [ldn]
```



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, see the chapters "Configuring ISDN" and "Configuring DDR" in the *Wide-Area Networking Configuration Guide* publication.

BRI Direct-Inward Dialing Configuration

The following example 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 gateway 1 on the BRI port. The DID information allows the gateway to route the call based on the called number. If the called number is 2xxx, the call is routed to gateway 2000, and if the called number is 3xxx, the call is routed to gateway 3000.

Figure 3-3 illustrates the topology of this connection example.



ISDN BRI Configuration (H.323)



Figure 3-3 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, use the configuration software provided with your gateway or network module, if any. Otherwise, for greatest power and flexibility use configuration mode (manual configuration). In this mode, you enter Cisco IOS commands at the gateway prompt.

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 3-10.

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

1. Select the T1/PRI interface to configure.

vg200(config)# controller t1 1/0

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

2. Specify which end of the circuit provides clocking.

vg200(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.

3. Specify the framing type.

vg200(config-controller)# framing esf

4. Specify the line code format.

vg200(config-controller)# linecode b8zs

- **Step 2** Configure parameters for the voice interface.
 - 1. Specify the PRI group and time slots to be mapped.

vg200(config-controller)# pri-group timeslots 1-24

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



T1/E1 Configuration (H.323)

```
2. Configure each pri-group as a virtual serial interface.
```

vg200(config-controller)# interface serial 1/0:15

Specify the T1 interface, unit number, and channel group to modify.

```
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
```

```
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 3-10.

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.
 - 1. Select the CE1/PRI interface to configure by entering the following command from Global configuration mode:

vg200(config)# controller e1 1/0

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

- **2**. Specify the framing type.
- vg200(config-controller)# framing crc4
- 3. Specify the line code format.

vg200(config-controller)# linecode hdb3

Step 2 Configure parameters for the voice interface.

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

vg200(config-controller)# pri-group timeslots 1-31

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

vg200(config-controller)# interface serial 1/0:31

Specify the E1 interface, unit number, and channel group to modify.

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



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 3-10.



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 following example shows how to configure VoIP to link PBX users with E&M trunk lines.

In this example, 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 PBX, connected to the voice network by 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 3-4 illustrates the topology of this connection example.



Figure 3-4 Linking PBX Users with E&M Trunk Lines

Note

This example 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 Cisco VG200, Incoming Caller ID Digits may not be processed correctly by Cisco IP IVR applications, such as Cisco IP Auto-Attendant.

Depending on the T1-CAS line provisioning, incoming DNIS digits received by the Cisco VG200 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 IP IVR application, the out-of-band digits will be interpreted as a user entry and change the application response. There are two ways to handle this situation:

- Request 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 SJ 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 SLC 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

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

Note

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 "H.323 Gateway Configuration" section on page 3-10.



Where to Go Next

At this point you should make sure that Cisco CallManager is properly configured to provision the Cisco VG200 gateway and to configure MGCP endpoints or H.323 route patterns as required. Refer to the *Cisco CallManager Administration Guide*, or to the online help panels on the Cisco Connection Online (CCO) Web page.

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

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Where to Go Next

Chapter 3 Configuring Voice over IP

