C H A P T E R   **4**

Voice Gateways

Router PSTN connectivity is generically referred to as *voice gateway* functionality, offering a *gateway* for voice over IP (VoIP) calls to, and from, traditional analog or digital PSTN or private branch exchange (PBX) calls. You can use a router voice gateway to connect to PSTN central office (CO) switches, private branch exchanges (PBXs), Key Systems, time-division multiplexing (TDM)-based interactive voice response (IVR) systems, traditional TDM-based voice mail systems, and any other legacy (non-IP) voice processing or telephone equipment.

This chapter explores several aspects of Cisco Unified CallManager Express (Cisco Unified CME) connectivity to the PSTN, including the following:

- Standards-based telephony signaling systems and protocols supported by Cisco IOS, which, in turn, determine what traditional TDM or analog systems to which you can connect and what features you get when using this type of connection

- A brief overview of the Cisco voice gateway hardware choices and the voice port densities and features provided

- Sample Cisco IOS configurations for different types of PSTN connections

- Network design and call switching considerations for connecting to the PSTN from your IP network

Traditional telephony terminology is used throughout this chapter. It has a more precise meaning here than in other chapters, because the topic of discussion is connecting a traditional telephony system, the PSTN.

Voice gateway considerations for Cisco Unified CME deployment are in the following sections:

Note   For additional information, see the "Related Documents and References" section on page xii.

Trunk Signaling Systems

Cisco IOS PSTN connectivity complies with the relevant standard signaling systems used by the PSTN and other telephony-switching systems. Cisco IOS routers support most signaling variations in general use. No matter where your business is located, you should be able to connect easily to the PSTN with the analog or digital signaling options described in this section.

Analog Signaling

Low-density PSTN connectivity typically implies an analog connection. In some geographies Basic Rate Interface (BRI) is used instead, as discussed in the "Digital Signaling" section on page 4-3. Analog signaling is also used for connections to analog stations (such as fax machines and traditional analog phones). Table 4-1 summarizes the analog signaling variations supported by Cisco IOS voice gateways.

Table 4-1   Analog Signaling Support by Cisco IOS Software

Signaling	Description	Typical Use
Analog DID	Analog Direct Inward Dial	Used to connect to an analog PSTN line that has DID service for incoming calls on it.
CAMA	Centralized Automatic Message Accounting	Used to connect to the PSTN for emergency services (911 calls) in North America.
E&M	Ear and Mouth	Used to connect to an analog PBX.
FXO	Foreign Exchange Office	Generally used to connect to an analog PSTN line. Also used to connect to a PBX or Key System FXS interface. Can be connected to any interface where a standard analog phone is currently connected.
FXS	Foreign Exchange Station	Used to connect to analog phone sets or fax machines. Occasionally also used to connect to a PBX or Key System if it offers only FXO interfaces.

To connect your Cisco Unified CME system to the PSTN for normal analog business line service, use FXO interfaces. FXO ports, like all the other analog interfaces, carry one call per port, so each RJ-11 port on your Cisco Unified CME router connects to one line from the PSTN and carries a single call at a time. A second call is given a busy tone if it tries to use the same port or line.

Note On voice interface cards, such as the NM-HDA and EVM-HD-8FXS/DID, which contain a single RJ-21 50-pin connector, the individual analog ports carried in the single cable are broken into separate RJ-11 ports by a break-out box.

The FXS and FXO voice interfaces are asymmetric, but most of the other signaling methods are symmetric. This means that if the PSTN offers an FXS interface (a normal business line), your Cisco Unified CME router connects to that with an FXO interface. On the other hand, you might have a Key System with FXO interfaces being used to connect to the PSTN. If you want to connect those same ports to your Cisco Unified CME router, you will require FXS interfaces on the router to connect to these ports.

Asymmetric also means that although you can make calls in both directions across FXS and FXO connections, services typically work in only one direction. For example, caller ID is *sent* on an FXS interface and *received* on an FXO interface, but not the other way around.

Analog trunks all support a single call per physical connection or port, so you need as many ports connected to the PSTN as you require simultaneous calls from your business to the PSTN.

FXO connections do not provide dialed digits (DNIS), introducing challenges in providing automatic call switching. More information about this is provided in the "PSTN Trunk Failover" section on page 4-18. Analog DID is a variation of FXO that provides DNIS on what is, essentially, an FXO interface. Note, though, that these trunks are one-way and can only receive calls from the PSTN (they cannot make calls to the PSTN). If you use analog DID for incoming calls from the PSTN, you still need FXO trunks as well to be able to make outgoing calls to the PSTN.

Digital Signaling

If you require only a small number of simultaneous calls to the PSTN, you will most likely use analog FXO connections. In geographic locations outside North America, ISDN BRI is a likely alternative option for low-density PSTN connectivity. However, if you have a larger office and require more than approximately 10 to 16 simultaneous calls to the PSTN, a digital T1 or E1 trunk might provide a more cost-effective option. Table 4-2 summarizes the digital signaling variations supported by Cisco IOS routers.

Table 4-2   Digital Signaling Support by Cisco IOS Software

Signaling	Description	Typical Use
BRI Q.931	Basic Rate Interface	An ISDN connection to the PSTN or a PBX carrying two simultaneous voice calls. It uses the Q.931 ISDN specification. Calls are controlled via a dedicated channel called the D channel. The term 2B+D is often used for BRI describing two voice channels (or bearer [B] channels) and one signaling channel (or data [D] channel).
BRI QSIG	Basic Rate Interface	Used for PBX ISDN connectivity. It uses the Q Signaling (QSIG) variation of the basic ISDN specification.
T1 CAS	T1 Channel Associated Signaling	Used widely in North America to connect to the PSTN or PBXs. Several variations of this signaling exist, including T1 FXS, T1 FXO, and T1 E&M. T1 FXS and T1 FXO support loop start and ground start signaling.T1 E&M signaling supports delay dial, wink, and immediate dial.
T1 FGD	Feature Group D	The T1 CAS variations generally cannot convey caller ID. T1 FGD can. It is used to connect to the PSTN where caller ID is required and PRI is not an option. T1 FGD is an asymmetric protocol.
T1 and E1 PRI	Primary Rate Interface	An ISDN connection to the PSTN carrying 23 (T1) or 30 (E1) simultaneous voice calls, giving rise to the terms 23B+D and 30B+D. It uses the Q.931 ISDN specification. Calls are controlled via a dedicated signaling channel (D channel).

Table 4-2   Digital Signaling Support by Cisco IOS Software

Signaling	Description	Typical Use
T1 PRI NFAS	Nonfacility Associated Signaling	A variation of PRI available only on T1 that uses a single D channel to control multiple spans of T1s with only B channels (voice calls).
T1 and E1 QSIG	Primary Rate Interface	Used for PBX ISDN connectivity. It uses the QSIG variation of the basic ISDN specification.
E1 R2	The Regional System 2 (R2) CAS protocol	Used in South America and Asia for PSTN connectivity. Numerous country-specific variations of the R2 protocol exist.
J1	Japan interface	PBX connectivity in Japan. Japan also uses the T1 standard.

BRI connectivity on the Cisco IOS routers is supported only for switch (PSTN, PBX, or key system) connectivity—not for ISDN BRI phones.

All ISDN variations listed in Table 4-2 support both DID and caller ID, which is implicitly supported in the ISDN protocol. The CAS protocols (T1 CAS and E1 R2) might or might not support caller ID. Typically T1 CAS does not, but T1 FGD is a variation that does. All digital trunk types support DNIS and DID.

Cisco IOS PSTN Telephony Interfaces

You can add numerous modular cards to your Cisco Unified CME router to support PSTN connections of the types discussed in the preceding section. These technologies and hardware cards are not particular to Cisco Unified CME. They can be used on any Cisco router that supports the card in question—independent of whether Cisco Unified CME is enabled on the router. For example, you can choose to have two separate routers in your office—one configured for Cisco Unified CME and the other as the PSTN voice gateway—as an alternative, or you can combine both functions in the same router.

The following sections cover these hardware choices in greater detail:

Analog Trunks

Voice interfaces range from two- and four-port FXO/FXS/E&M/DID cards up to 96/120-channel quad T1/E1 interfaces. The physical telephony interface for analog and BRI ports is provided by a plug-in voice interface card (VIC) and for a T1/E1 port by a voice or WAN interface card (VWIC).

Using various combinations of VICs and VWICs on a Cisco IOS router, you can build a Cisco Unified CME system that includes a range of physical telephone interfaces. You can assemble a small analog telephony system with a few FXO ports used to connect to PSTN subscriber lines, or you

can use digital telephony interfaces such as T1/E1 and ISDN BRI/PRI, or any combination of these. The specific hardware cards offering analog trunk and station (analog phone or fax machine) interfaces are discussed next.

Analog Trunk and Station Hardware

The analog interface cards listed in Table 4-3 are used to provide low-density analog PSTN interfaces. VICs are placed in a Voice Interface Card (VIC) or WAN interface card (WIC) slot (supported on the Cisco 1751 and Cisco 1760), in a high-speed WIC (HWIC) slot on the router (supported on the Cisco 2800 and Cisco 3800 series), or inside a network module (Cisco 2600, Cisco 2800, Cisco 3700, and Cisco 3800 series) such as the NM-HD-1V, NM-HD-2V, NM-HD-2VE, or NM-HDV2. For high-density analog PSTN interfaces, the NM-HDA (supported on the Cisco 2600, Cisco 2811, Cisco 2821, Cisco 2851, Cisco 37xx, and Cisco 38xx) or the EVM-HD-8FXS/DID card (supported on the Cisco 2821, Cisco 2851, and Cisco 38xx) can be used.

Table 4-3   Analog Interfaces, Signaling, and Density

Interface Card	Signaling	Density
VIC-4FXS/DID	FXS and analog DID	4 ports
VIC2-2FXO	FXO and CAMA	2 ports
VIC-2DID	Analog DID	2 ports
VIC2-4FXO	FXO and CAMA	2 ports
VIC2-2FXS	FXS	2 ports
VIC2-2E/M	E&M	2 2
NM-HDA-4FXS, EM-HDA-8FXS, and EM-HDA-4FXO	FXS and FXO	4 ports on the baseboard, but can be expanded up to 12 FXS ports by adding an EM-HDA-8FXS card to the network module (NM), or up to 8 FXO ports by adding 2 EM-HDA-4FXO cards to the NM.
EVM-HD-8FXS/DID, EM-HDA-8FXS, EM-HDA-6FXO cards, and EM-HDA-3FXS/4FXO	FXS, FXO, CAMA, and analog DID	8 ports on the baseboard that can be FXS or DID. You can expand the EVM-HD to up to 24 FXS ports by adding 2 EM-HDA-8FXS cards, or up to 12 FXO ports by adding 2 EM-HDA-6FXO cards, or various combinations of FXS and FXO by adding 1 or 2 EM-HDA-3FXS/4FXO cards. The EVM-HD supports any combination of 2 EM cards.

The cards that support multiple signaling systems (such as FXS or DID, and FXO or CAMA) can be software configured on a per-port basis to support one or the other. For example, the VIC2-4FXO card can be configured to support one CAMA and three FXO ports, or two CAMA and two FXO ports.

Configuring Analog Trunks and Stations

All PSTN interfaces are configured as voice ports on the router. When you insert the card into the router, the configuration automatically creates and shows the corresponding voice ports. Directing calls to a voice port is based on the dial plan and is implemented with plain old telephone service (POTS) dial peers.

Note that the use of the 9T directive in the **destination-pattern** command of the dial peer in the following example configuration:

```
dial-peer voice 100 pots
 description PSTN
 destination-pattern 9T
 port 1/0/0

dial-peer voice 100 pots
 description PSTN
 destination-pattern 9T
 port 1/0/1
```

This command is a quick way of dealing with variable-length PSTN dial plans. The T denotes a timeout. The command **destination-pattern 9T** instructs the dial peer to match any dialed digits that start with a nine, regardless of how many digits follow. When the timeout expires, the digits are forwarded from the voice port to the PSTN. There are other, more explicit ways to make your **destination-pattern** commands match calls to the PSTN more exactly, including **9911**, **9411**, **91T**, and **9[2-9]**.

The dial peers shown in the following configuration example direct all calls (of a varying number of digits) that start with a nine to the two PSTN FXO trunks, ports 1/0/0 and 1/0/1. If no preference is given on the dial peers and both trunks are free, the Cisco IOS software chooses one of the two trunks based on an internal algorithm that considers idle times and usage of the trunks. You can control the order in which they are chosen by adding a **preference** command to the dial peer. The **dial-peer hunt** command offers additional control over the sequence in which dial peers, and therefore voice ports, are chosen.

You can also direct calls to different destinations over different trunks if required. This is shown in the following example, where calls to the 408 area code always use voice port 1/0/0, and calls to the 415 area code always use voice port 1/0/1:

```
voice-port 1/0/0

voice-port 1/0/1

dial-peer voice 100 pots
 description PSTN
 destination-pattern 9408.......
 port 1/0/0

dial-peer voice 101 pots
 description PSTN
 destination-pattern 9415.......
 port 1/0/1
```

The preceding example illustrates how you can connect to each independently and direct different types of calls to the correct trunks, if you have different local and long-distance PSTN provider connections.

If you require CAMA connectivity to comply with North American emergency calling regulations, you can configure one or more of your FXO ports for CAMA operation. This is shown in the following example, where port 2/0/3 on a VIC2-4FXO card is configured for CAMA signaling. The following example configuration illustrates this command usage:

```
voice-port 2/0/0

voice-port 2/0/1
   signal ground-start

voice-port 2/0/2

voice-port 2/0/3
   signal cama KP-NPD-NXX-XXXX-ST
```

✎

Note   For more information about the **signal** command, see your applicable Cisco IOS Command Reference. The following applies to Cisco IOS 12.3(T): http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_command_reference_chapter0918 6a0080.

Analog Trunk Features

With analog FXO interfaces, caller ID information received for an incoming PSTN call is displayed on the IP phones. You can optionally enable the Flash softkey on your IP phones. Pressing the Flash softkey on the IP phone generates a hookflash signal on the FXO port and allows you to exercise PSTN subscriber line services, such as PSTN call waiting and three-way calling. However, Cisco IOS FXO ports do not support PSTN call waiting caller ID display.

You can also set up a direct link between a specific PSTN telephone line and an individual button on an IP phone. This is useful if you want to use PSTN-based voice mail services where a stutter dial tone on the PSTN line indicates that a message is waiting.

As mentioned earlier, in the "Analog Signaling" section on page 4-2. FXO interfaces are asymmetric. As such, calls can be disconnected in only one direction in pure FXO operation. The historic reasons for this are beyond the scope of this publication. Suffice it to say that today FXO ports are widely used as two-way trunks, and special care must be taken that calls disconnect properly in both directions and do not hang the port. You can use the following Cisco IOS commands on the voice port to facilitate proper call disconnect on FXO ports:

- **battery-reversal**
- **signal groundstart**
- **supervisory disconnect anytone**
- **supervisory disconnect dualtone**

The selection of a particular command depends on the complementary features provided by your PSTN CO switch. It also varies based on geographic location and the technology available in the CO.

In addition, FXO signaling does not receive dialed digits (DNIS). This means that an incoming call from the PSTN to an FXO port cannot be switched automatically by your Cisco Unified CME system to an extension, because there are no digits from the PSTN to tell Cisco Unified CME where to switch it. You can overcome this shortcoming of FXO signaling by using auto-terminate directives on the FXO voice port to switch the call to a predetermined destination. Commands you can explore include **connection plar** and **connection plar-opx**, which are described in the "PSTN Call Switching" section on page 4-12.

Digital Trunks

Digital trunks can be low-density (for example, BRI with two calls per port) or high-density (such as T1 or E1 ports with 24 or 30 calls per port, respectively). The specific hardware cards offering digital trunk interfaces are discussed in the following sections.

Digital Trunk Hardware

The digital interface cards listed in Table 4-4 are used to provide a range of low- to high-density digital PSTN interfaces.

Table 4-4   Digital Interfaces, Signaling, and Density

Interface Card	Signaling	Density
VIC2-2BRI-NT/TE	Q.931 or QSIG BRI	2 ports with 2 voice channels each.
NM-HDV	T1 and E1	Up to 2 T1/E1 ports. Up to 48 (T1) or 60 (E1) voice channels. Used in conjunction with a VWIC-1MFT-T1/E1 or VWIC-2MFT-T1/E1.
NM-HD-2VE	Analog, BRI, T1, and E1	Up to 4 T1/E1 ports, or 2 T1/E1 and 2 BRI ports, or 4 BRI ports. Up to 24 voice channels. Used in conjunction with a VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, or VIC2-2BRI-NT/TE.
NM-HDV2	Analog, BRI, T1, and E1	Up to 4 T1/E1 ports, or 2 T1/E1 ports and 2 BRI ports. Up to 120 voice channels. Has up to 2 onboard T1/E1 ports. For the additional ports, a VWIC-1MFT-T1/E1 or VWIC-2MFT-T1/E1 is used. For BRI, the VIC2-2BRI-NT/TE card is used inside the NM.
EVM-HD-8FXS/DID, EM-4BRI-NT/TE	Analog and BRI	Up to 8 BRI ports (16 voice channels).
VWIC-1MFT-T1/E1 or VWIC-2MFT-T1/E1 in a WIC slot	T1 and E1	Up to 2 T1/E1 ports. Channel density depends on the router platform and where the DSPs are accessed from.

In the general case, a T1 port offers 24 voice channels, and an E1 port offers 30 voice channels. When using ISDN signaling, where one channel is dedicated to call control signaling (the D channel), a T1 carries 23 voice channels, and an E1 carries 30 voice channels. (An E1 always has a channel dedicated to signaling, no matter what type of protocol is used. With T1 this is not normally the case; using ISDN takes away one of the standard channels.)

You might not use the maximum number of channels on these ports, depending on what your PSTN service provider offers. You can configure your Cisco IOS router with any number of channels on the T1 or E1 interface, but it must be complemented by what is configured on the PSTN CO on the other side.

Fractional T1 service is quite common in North America and allows you to subscribe to PSTN T1 service with, for example, only 12 or 16 channels of service (and this service costs less than a full T1 of 24 channels). This service can be either T1 CAS or T1 PRI. Another service is to multiplex your WAN connection (Frame Relay or Point-to-Point Protocol [PPP]) on some channels of the same physical T1 used for your PSTN voice connection. For example, channels 1 to 6 could offer a 384-Kbps PPP WAN connection; channels 10 to 20 could offer ten channels of PSTN voice service using T1 E&M signaling.

Fractional E1 service is much less common. Your lower-density PSTN connectivity options in geographies that use E1 connectivity may be multiples of BRI until such time as a full E1 makes sense for your business.

Configuring Digital Trunks

Digital PSTN interfaces are configured in general just like analog interfaces—that is, as voice ports and POTS dial peers on the router to direct calls to the ports. The dial peer control and configuration are exactly the same, regardless of what type of voice port used.

T1/E1 ports, however, show up as controllers in a basic configuration (by just inserting the hardware into the router). Unlike an analog interface, the voice port is not created until you add more configuration details to the controller. T1/E1 ports are used for both data and voice access. Until you add specific configuration statements, the router does not know what your intention is with the T1/E1 port. Add a voice configuration to a T1/E1 port by using either the **ds0-group**, **tdm-group,** or **pri-group** command. A data T1/E1 port is configured with the **channel-group** command.

You often see the terms CAS and *common channel signaling* (*CCS*) when reading about T1/E1 trunks. CAS generally means that the signaling to control the call uses the same channel (or timeslot) as the call's media path. This is common on T1 interfaces. (It is also called *robbed-bit signaling* because a few bits out of the 64-kbps channel are "stolen" from the media path to convey call control information, such as on-hook and off-hook.) CCS means that a channel is dedicated to signaling. This channel carries the call control information for all the voice calls (media paths) on that same T1/E1 interface. For example, channel 16 on an E1 is used exclusively for call control and carries the control information for all the other channels (1 to 15 and 17 to 31) on that interface.

The following configuration example illustrates a T1 CAS (E&M immediate start) PSTN connection using a **ds0-group** configuration. In this example, you can see that the second port on the VWIC shows up as controller T1 2/1. This means that the hardware has been detected but no configuration has been done for this port

```
controller T1 2/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 0 timeslots 1-24 type e&m-immediate-start

controller T1 2/1

voice-port 2/0:0
 signal immediate

dial-peer voice 100 pots
 description PSTN
 destination-pattern 9T
 port 2/0:0
```

In this example, all 24 channels on the T1 are configured. But you could as easily have stated **ds0-group 0 timeslots 1-10** if you agreed with your provider to get only ten channels of PSTN service on this T1 (fractional T1 service). The result of the **ds0-group** command is that voice port 2/0:0 is created. The POTS dial peer, in this example, looks the same as the one in the FXO example earlier, except that it now points to voice port 2/0:0, which is a T1 port.

> **Note**
> For more information about digital trunks, see the following document:
> http://www.cisco.com/en/US/tech/tk652/tk653/technologies_configuration_example09186a008010f05d.shtml

If you are using ISDN PRI service to the PSTN, you use the **pri-group** command to insert a voice configuration on a T1 or E1 controller. The following configuration example shows a sample configuration for a T1 PRI trunk.

```
isdn switch-type primary-5ess

controller T1 2/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-24
```

```
interface Serial2/0:23
 no ip address
 isdn switch-type primary-5ess
 isdn incoming-voice voice

voice-port 2/0:23
 echo-cancel coverage 64

dial-peer voice 100 pots
 description PSTN
 destination-pattern 9T
 port 2/0:23
```

Geographic variants of ISDN are controlled by the **switch-type** setting. A default router setting, seen in the preceding example as the first line in the configuration, is specified at the Cisco IOS global level (the **isdn switch-type** command). This default can be overridden on a per-interface basis by the **switch-type** statement under the controller. In the preceding example, both are set to **primary-5ess**, but they could be different. If they are different, the statement on the controller takes precedence.

The D- channel interface (**interface Serial 2/0:23**) and voice-port (**voice-port 2/0:23**) commands are automatically created by the insertion of the **pri-group** command on the controller. The POTS dial peer again looks exactly the same as in previous examples. You must adjust the voice port to which it refers.

Digital Trunk Features

For PRI/BRI interfaces using ISDN signaling, you can optionally allow the IP phone's full DID name and number to be used as the calling party's identity for outgoing calls. This puts extension-specific information into the PSTN billing records for the call. This can be useful if you want to rely on the PSTN provider's billing information to track the internal origin point of PSTN calls made from your Cisco Unified CME system. Alternatively, you can block IP phone extension-specific information from the outgoing ISDN call and instead substitute the general public phone number for your system.

Generally, PSTN providers do not use name information delivered to the PSTN by a subscriber system. Although the name can be included in the ISDN call setup, the PSTN typically overrides this with the information associated with the subscriber in the PSTN's own databases. You can, however, receive name display information from the PSTN on ISDN trunks, and display this on the IP phones in your business.

All digital trunks provide DID (or DNIS) information. ISDN trunks also provide caller ID delivery. Fractional CAS and PRI are supported on the Cisco IOS routers. If you configure fractional PRI, the D channel for the T1 must be on channel 24 and for E1 on channel 16. This cannot be customized. The voice channels (B channels) can be any subset of the remaining channels.

ISDN channels cannot be customized to be incoming only or outgoing only. However, through creative use of dial peers, you can limit the number of incoming or outgoing calls to and from your business. You cannot specify the exact channel each call should use. With T1 CAS, you have more granular control, because you can specify separate ds0-groups (up to a ds0-group per channel). Each ds0-group creates a separate voice port that you can control via dial peers as to what calls may reach those channels. The following configuration example illustrates this feature.

```
controller T1 2/0
 framing esf
 clock source internal
 linecode b8zs
 ds0-group 0 timeslots 1-10 type e&m-immediate-start
 ds0-group 1 timeslots 15-20 type e&m-immediate-start

controller T1 2/1
```

```
voice-port 2/0:0
 signal immediate

voice-port 2/0:1
 signal immediate

dial-peer voice 100 pots
 description PSTN
 destination-pattern 9408.......
 port 2/0:0

dial-peer voice 101 pots
 description PSTN
 destination-pattern 9415.......
 port 2/0:1
```

The **ds0-group 0 timeslots 1-10** command results in voice port 2/0:0, and the **ds0-group 1 timeslots 15-20** command creates voice port 2/0:1.

DSP Hardware

Digital signal processor (DSP) technology provides voice compression, echo cancellation, tone generation, and voice packetization functions for servicing voice interfaces and converting the voice for transport over packet networks. To drive a PSTN voice connection, the analog or digital voice port must have access to a DSP for the call.

Some voice NMs include internal slots into which DSP modules can be plugged, and others have fixed DSP configurations. In some router models, such as the Cisco 1760, Cisco 2800, and Cisco 3800 series, you can plug DSP cards directly into the router's motherboard.

VWIC cards offer only physical T1/E1 port connections, and VIC cards offer only the physical analog or BRI ports. If a VIC or VWIC card is inserted into a router WIC slot (supported on the Cisco 1751, Cisco 1760, Cisco 28xx, and Cisco 38xx), the DSPs are typically provided by the onboard DSP cards. A VIC or VWIC inserted into an NM typically draws on DSPs resident on the NM itself.

One other variation is to use a VWIC in a WIC slot on the Cisco 2600 or Cisco 3700 series platforms, which do not support onboard DSPs. For this configuration, you can use a DSP AIM card such as the AIM-VOICE-30 or the AIM-ATM-VOICE-30 card. An Advanced Integration Module (AIM) is an internal plug-in module that fits on the router's motherboard. The AIM-based DSPs cannot drive analog or BRI VIC cards, only T1/E1 VWICs.

DSP cards for motherboard and NM-based slots come in many densities and use various DSP technologies. All are called packet voice/fax DSP module (PVDM) cards.

PSTN Trunks Integrated with or Separate from Cisco Unified CME

In a typical deployment, the PSTN connectivity for your business is integrated into your Cisco Unified CME router. However, you could also use a separate router platform as your PSTN gateway. You may choose to do this because you already have a router that acts as your PSTN gateway in your office or because the slot density on your Cisco Unified CME router is insufficient for the PSTN connectivity your office requires.

For PSTN trunks integrated onto your Cisco Unified CME router, the voice call is switched directly from the POTS interface to the IP phone and is straightforward to configure. Placing the PSTN gateway on a different platform gives you an H.323 (or SIP) call leg between the PSTN gateway and the Cisco Unified CME call controller where the IP phones are managed. This requires POTS dial peers on

the PSTN gateway to direct calls to the PSTN interfaces, as shown in the previous configuration examples in this chapter. It requires H.323 dial peers to direct calls from the PSTN gateway to IP phones and from the IP phones to the PSTN gateway. From an H.323 standpoint, this configuration is similar to connecting two separate Cisco Unified CME systems via an H.323 VoIP interface between them. This is shown in Figure 4-1.

Figure 4-1   Integrated or Separate PSTN Gateways
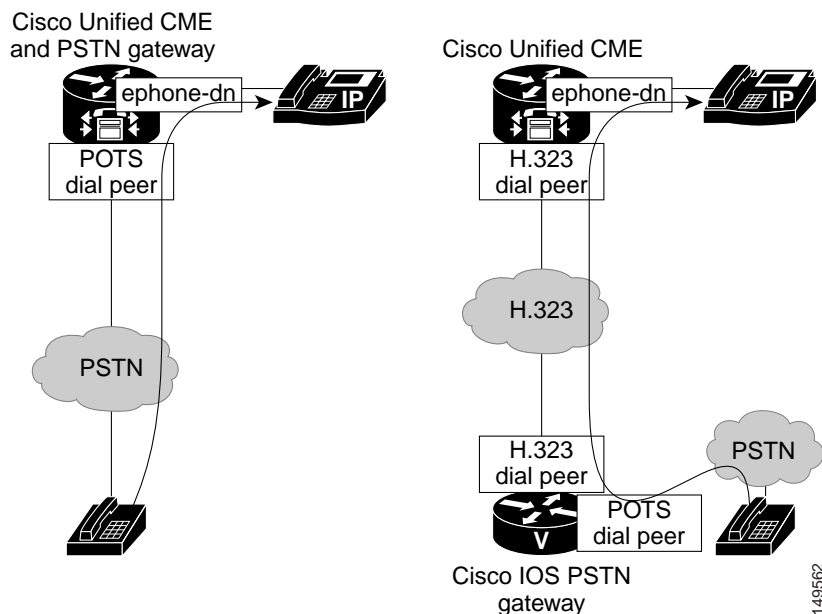
We recommend that you deploy Cisco Unified CME with an integrated PSTN gateway, because this approach results in a much simpler network design and configuration. If Cisco Unity Express is used for the automated attendant (AA) or voice mail on your Cisco Unified CME system, the H.323 VoIP leg must be converted to a SIP call leg before the call can successfully terminate on the Cisco Unity Express application. For this type of implementation, we recommend Cisco Unified CME 3.2 because with Cisco Unified CME 3.2 and later, you can use the Cisco IOS translation shown in the following configuration example.

```
voice service voip
 allow-connections h323 to sip
```

PSTN Call Switching

The preceding sections explored the various PSTN trunk types, signaling methods, and router hardware you can use to connect to the PSTN. But there are more considerations than just physical connectivity. One thing to keep in mind is that the PSTN numbers and your internal extension numbers are almost certainly not the same, at least not the same length. Digit translation must occur to map one set of numbers to another. You will learn more about this in the "Digit Manipulation" section on page 4-15.

Another consideration is what PSTN numbers (and how many) your business has or needs. Do you have just one main office number, and the receptionist directs all calls to the correct employee? Do you need an AA menu to have callers switch to the person or service they want to speak to? Should you have DID numbers for all or some of your employees? Do you prefer a Key System type of operation, where a series of PSTN numbers appear as distinct line appearances on a number of phones, and any employee can

pick up any call? There is interaction between your business needs, the PSTN service you get from your provider, the capabilities of the physical connection to the PSTN, and the Cisco Unified CME configuration (IP phone button appearances) to use.

The physical connection is likely dictated by cost, your office's geographic location, and the number of voice channels your office needs. You might want DID service for your business. However, if it is not offered in your area at a cost-effective level, you have little choice but to implement for non-DID service.

The following sections explore considerations about how calls may be routed depending on whether you have DID service and how you can handle calls to non-DID destinations within your business:

- PSTN Call Switching with DID Enabled, page 4-13
- PSTN Call Switching with DNIS (No DID), page 4-13
- PSTN Call Switching with No DNIS (FXO Trunks), page 4-14

PSTN Call Switching with DID Enabled

Many offices deploying Cisco Unified CME have DID capability from the PSTN provider for some subset of employees. PSTN calls to DID destinations can be switched automatically to the employee's phone without any manual intervention.

There are two situations to consider, depending on how DID numbers are allocated to destinations within your business:

- All employees, the AA, and voice mail pilot numbers have DID numbers assigned. In this configuration, PSTN calls can be switched as follows:
  - The main office number (non-DID calls) terminates on the AA pilot (for AA assistance) or on an IP phone extension (for receptionist assistance).
  - Employee DID numbers terminate on the extension for that person's IP phone.
  - The number employees call from PSTN locations to check their voice mail terminates on the voice mail pilot number.

- Some employees have DID numbers assigned, and others do not. In this configuration, PSTN calls are handled as follows:
  - Calls to the main office number and for PSTN voice mail checking are handled as per the preceding scenario.
  - Calls to employees with DID numbers terminate on the extension for that person's phone.
  - Calls to employees without DID numbers terminate on the AA (or receptionist's extension). These callers then can dial through or be transferred to the extension of the person they want to reach.

PSTN Call Switching with DNIS (No DID)

PSTN call switching with DNIS and no DID is not a likely configuration, but it is possible. In this configuration, your business does not have DID service and, therefore, has only a single main office number from the PSTN provider although you have multiple trunks. Or even if you have multiple PSTN numbers, they are not associated with particular employees, but instead are just alternate main office numbers.

Although the dialed number is delivered via DNIS from the PSTN to the PSTN gateway, it is of little use to switch calls to individual destinations. In this situation, you have two configuration choices:

- Regardless of the dialed number, all PSTN calls are terminated on the AA pilot (for AA assistance) or on a specific phone extension (for receptionist assistance).

- All PSTN lines appear on multiple phones (Key System operation). Any employee at these phones can answer any of the lines, regardless of what number the caller dialed.

If all calls are directed to the AA (or a receptionist), *caller-busy* conditions must be carefully considered. For example, you need to determine what should happen if all AA ports are busy or all the receptionists are busy. If you do not want busy tone returned, more ports or receptionists may be required, or alternate destinations to switch calls to (lower-preference dial peers) or DID service may be needed for high-volume destinations in your business.

PSTN Call Switching with No DNIS (FXO Trunks)

If the office has only FXO trunks, no DNIS (or DID) capability is technically possible. This scenario is very common for a small standalone office or a small branch of a bigger network that has only a few business lines from the local CO.

Because no dialed digits (DNIS) are available on FXO trunks, these calls must be autoterminated on a predetermined destination (most often the AA or the receptionist's extension) — or these calls must ring all phones that have the line appearance on them (in Key-system mode). This can be achieved with a private line automatic ringdown (PLAR) configuration on the voice port where a particular destination extension is associated with the trunk, and all calls arriving on that trunk are switched as if they had dialed the configured extension. This syntax is shown in the following configuration example, where all calls arriving on the FXO trunk on slot 1/0/0 are switched as if they had dialed extension 6800.

```
voice-port 1/0/0
  connection plar opx 6800
```

Most small offices have multiple FXO trunks to the PSTN because each trunk can carry only a single call. One or more PSTN numbers may be associated with these trunks or this trunk group, depending on the PSTN service the business subscribes to. Generally, there are two possibilities:

- *A single PSTN main office number*—In this configuration, all calls from the PSTN are terminated (via the PLAR feature) to the AA pilot (for AA assistance) or to a specific phone extension (for receptionist assistance).

- *Multiple PSTN numbers*—This might consist of one PSTN number for the main office and another for voice mail retrieval by employees. In this case, calls on main office trunks are switched as in the preceding case, and calls to voice mail are switched (via PLAR) to the voice mail pilot number. Clearly, these two types of calls must be delivered on different physical trunks or trunk groups so that each is autoterminated to the correct destination.

Sharing all FXO trunks across all PSTN calls (the first case in the preceding list) results in better trunk utilization than assigning distinct FXO trunk(s) to the main office number and other distinct FXO trunk(s) to the voice mail pilot number (the second case).

Digit Manipulation

There are various reasons to manipulate the digits dialed by the caller on a voice system. The most common reason is to allow both internal calls (from other extensions) and external calls from the PSTN (where a full E.164 phone number is delivered) to terminate directly on the user's phone without needing a receptionist to intercept and redirect the call.

Note   E.164 is an International Telecommunication Union (ITU) recommendation that describes international telephone dial plans. It specifies phone number attributes such as international dialing codes, regional (area) codes, and the minimum and maximum length of each field in the phone number. Voice systems use the E.164 criteria to parse and interpret phone numbers. PSTN numbers are always fully qualified E.164 numbers, whereas extensions within your business typically are not, because they are private numbers of local significance only.

Here are some other reasons to translate (or manipulate) digits:

- To allow IP phone users to call each other directly by extension, and also to access the PSTN
- To allow for site access codes in a multisite on-net dial plan and to strip these digits to extract the extension as soon as the destination site is reached
- To allow for variable-length external (off-net) dialing while maintaining fixed-length internal dialing
- To block calls to certain numbers
- To redirect calls to certain numbers

For example, suppose your employee, UserU1, is at extension 3001, and her PSTN DID number is 4xx-5yy-3001. Without some form of digit manipulation or live intercept, a call incoming from the PSTN that dialed 4xx-5yy-3001 will not match the ephone-dn definition for UserU1's phone, which contains only her extension, 3001. Therefore, a method is needed to translate the string 4xx5yy3001 to 3001.

Several Cisco IOS digit manipulation tools can translate phone numbers. The following are the most common:

- Dial peer commands
- Cisco Unified CME **dialplan-pattern** command
- Cisco IOS translation rules

Dial Peer Commands

You can include several commands on a POTS dial peer to add, suppress, or substitute the digits forwarded to the PSTN trunk interface:

- **destination-pattern**
- **digit-strip**
- **forward-digits**
- **prefix**
- **translate-outgoing**
- **translation-profile**

Dial peer commands are useful if only small changes to the beginning or end of the dialed number are necessary, such as prefixing an area code, prefixing a CO designator (NXX) to an extension number, or forwarding only the last four digits of a longer number. The wildcard matching within the **destination-pattern** command automatically deletes the numbers explicitly matched. For example, when 5yy3001 is dialed and is matched by a dial peer that contains the command destination-pattern 5yy...., the default operation is to forward only the digits 3001.

Note   For more information about **digit-manipulation** commands referenced in this section, see the Cisco IOS Command Reference for you Cisco IOS Release listed with the appropriate Cisco IOS Documentation Set at the following URL:
http://cisco.com/en/US/products/sw/iosswrel/tsd_products_support_category_home.html

Cisco IOS Translation Rules

For more extensive digit manipulation, such as a wholesale change of a number or substituting digits in the middle of a number, translation rules are much more powerful. Translation rules are regular expressions attached to the dial peer with the **translation-profile** command.

Note   For more information about voice translation rules, see the "Voice Translation Rules (Doc ID: 16083)" presented on Cisco.com.

Like the other dial peer commands discussed in the preceding section, translation rules are a generic Cisco IOS feature that allows manipulation of called numbers, calling numbers, and number types. It can also be attached in such a way that it translates calls in only one direction, either incoming or outgoing.

The following example illustrates configuration commands for a T1 PRI trunk, with translation profile **to_261x** attached for incoming calls (calls from the PSTN to the Cisco IOS PSTN gateway). Translation profile **to_261x**, in turn, references **translation rule 23**, which has ten rules specified. This CLI segment intercepts all calls incoming from the PSTN over this T1 PRI that contains a dialed number ending in the range 12610 to 12619. It does not matter what (or how many) numbers precede this range; for example, it could be 5xx-3y1-2610 or 5y1-2618. The numbers that match the rule (12610 to 12619) are translated to a completely unrelated number so that none of the original digits survive. To illustrate, if a call with a dialed number of 5xx-3y1-2610 arrives, it is translated to 32085, and an IP phone (or other dial peer) associated with that extension receives the call. A PSTN call with a dialed number of 5y1-2618 results in extension 79988 receiving the call.

```
voice translation-rule 23
 rule 1 /12611/ /37002/
 rule 2 /12612/ /37262/
 rule 3 /12613/ /37990/
 rule 4 /12614/ /57514/
 rule 5 /12615/ /30631/
 rule 6 /12616/ /50043/
 rule 7 /12617/ /28787/
 rule 8 /12618/ /79988/
 rule 9 /12619/ /68278/
 rule 10 /12610/ /32085/

voice translation-profile to_261x
 translate called 23

controller T1 2/0
 framing esf
 linecode b8zs
```

```
 pri-group timeslots 1-24

interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-5ess
 isdn incoming-voice voice

voice-port 2/0:23
 echo-cancel coverage 64

dial-peer voice 1261 pots
 translation-profile incoming to_261x
 incoming called-number 1261.
 direct-inward-dial
 port 2/0:23
```

The syntax for translation rules can be cryptic if you are unfamiliar with regular expressions, but these rules can provide a powerful facility to manipulate digits. Translation rules are not tied to Cisco Unified CME, so you can use it on any Cisco IOS voice-enabled router.

Here are some considerations when using the Cisco IOS voice translation rules feature:

- These rules make for a very powerful overall feature that can do almost any translation of digits required, but using these rules can be complicated and, therefore, prone to errors for inexperienced implementers.

- Being a generic Cisco IOS feature, the feature's rules apply to all calls that traverse the router. It can be applied at a global level, dial peers, and ephone-dns (Cisco Unified CME IP phones).

- The digits are manipulated before dial peer matching and call termination.

- Calling and/or called numbers can be manipulated on every call based on what is configured.

- The rules can be directionally applied to incoming or outgoing calls (or both).

When applying translation rules to ephone-dns, there is a side effect that if no rule is matched, an extra post-dial delay is incurred. As a workaround, create a dummy translation rule that acts as a pass-through. For example, if no rule is applied to extension-to-extension calls, and the extensions all start with 5, add a rule that "translates" 5 to 5, just to make sure that a rule is always matched, and the delay is not incurred.

Cisco Unified CME dialplan-pattern Command

The Cisco Unified CME **dialplan-pattern** command allows E.164 numbers to be mapped to extension numbers or, put another way, to extract the extension number from a longer DID number. The **dialplan-pattern** command does not actually translate the number (although the result from a call routing point of view is the same). It instead creates multiple dial peers that allow different dialed numbers to terminate on the same phone.

The **dialplan-pattern** command can be used in some cases (calls to IP phones) to achieve the same call routing as can be achieved by using translation rules. Because these two features operate differently, you should think carefully about which method to use. If you use both methods, you should be clear about how these might interplay with each other to affect your call routing. The **dialplan-pattern** command is explained in more detail in Chapter 6, "Connecting Multiple Cisco Unified CallManager Express Systems with VoIP."

The CLI shown in the following example illustrates the same mapping as the number translation discussed previously for employee UserU1.

```
telephony-service
```

```
load 7960-7940 P00303020214
max-ephones 48
max-dn 192
ip source-address 10.1.3.1 port 2000
system message CUE System 2691
create cnf-files version-stamp 7960 Jul 15 2003 13:48:12
dialplan-pattern 1 510395.... extension-length 4
voicemail 6800
max-conferences 8
web admin system name cue password cue
dn-webedit
time-webedit
```

Some considerations about using the Cisco Unified CME **dialplan-pattern** feature include the following:

- Feature provides a straightforward method to translate from full E.164 numbers to shorter extension numbers on Cisco Unified CME.

- It is a Cisco Unified CME feature, so it applies only to calls to and from IP phones controlled by Cisco Unified CME. It does not apply to calls from the PSTN gateway directly to the AA or voice mail pilot numbers. Therefore, if digit manipulation is needed on these calls, one of the other two methods must be used.

- The IP phone extension must have at least one digit in common with the original called number and be in the same sequence. (If the extension is completely different from the called number, or not in sequence, Cisco IOS translation rules must be used to manipulate the digits.)

- Like Cisco IOS translation rules, the digits are manipulated before dial peer matching and call termination.

- It manipulates the *called* number on a call to an IP phone and the *calling* number of a call from an IP phone. This operation is implicit and cannot be controlled or altered.

- The E.164 number patterns generated by the **dialplan-pattern** command can be registered to an H.323 gatekeeper or SIP proxy. Digit translations done with Cisco IOS translation rules are not registered to H.323 gatekeepers or SIP proxies.

PSTN Trunk Failover

Larger offices that use a digital trunk, such as a PRI, often need a backup method to connect to the PSTN. This requirement results in the PRI being the main PSTN connection point in addition to FXO trunks (typically used to back up a T1) or BRI interfaces (typically to back up an E1) used if the main interface is down.

In this configuration, the dial peers directing calls to the main interface must be duplicated to also point to the backup interface. You can prioritize calls to use the main interface when it is available by using the **preference** command on the dial peers pointing to these trunks.

Another need is to have a backup mechanism for a small office with FXO trunks if a power failure occurs. FXO hardware supports a feature called FXO Power Failover that allows a hardware (relay) connection between a *red* phone (a specially dedicated analog telephone in your office that normally is not in use) and the PSTN line, in case the router is not powered.

On Cisco voice hardware, the NM-HDA-4FXS FXO expansion card (the EM-HDA-4FXO) and the EVM-HD-8FXS/DID FXO expansion card (the EM-HDA-6FXO) each have one port per card that has this power failover capability. Other Cisco FXO hardware cards do not support this feature.

Note   For more information about the **preference** command in relation to dial peer configuration, see the following URL:

http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_command_reference_chapter0918 6a00801a7f30.html#wp1109642