Configuring Dial Plans, Dial Peers, and Digit Manipulation

This chapter describes how to implement dial plans by configuring dial peers and using dial peer matching and digit manipulation features. This chapter contains the following sections:

For a complete description of the commands used in this chapter, refer to the *Cisco IOS Voice, Video, and Fax Command Reference*. To locate documentation of other commands that appear in this chapter, use the command reference master index or search online.

To identify the hardware platform or software image information associated with a feature in this chapter, use the Feature Navigator on Cisco.com to search for information about the feature or refer to the software release notes for a specific release. For more information, see the "Identifying Supported Platforms" section in the "Using Cisco IOS Software" chapter.

Dial Plan Overview

A dial plan essentially describes the number and pattern of digits that a user dials to reach a particular telephone number. Access codes, area codes, specialized codes, and combinations of the number of digits dialed are all part of a dial plan. For instance, the North American Public Switched Telephone Network (PSTN) uses a 10-digit dial plan that includes a 3-digit area code and a 7-digit telephone number. Most PBXs support variable length dial plans that use 3 to 11 digits. Dial plans must comply with the telephone networks to which they connect. Only totally private voice networks that are not linked to the PSTN or to other PBXs can use any dial plan they choose.

Dial plans on Cisco routers are manually defined using dial peers. Dial peers are similar to static routes; they define where calls originate and terminate and what path the calls take through the network. Attributes within the dial peer determine which dialed digits the router collects and forwards to telephony devices.

Note   If you are using Media Gateway Control Protocol (MGCP) or Simple Gateway Control Protocol (SGCP) on your call agent, you do not need to configure static dial peers. See the chapter "Configuring MGCP and Related Protocols" for more information.

The following sections provide an overview of basic dial peer concepts:

Note   The illustrations and sample configurations in this section use VoIP; the same concepts also apply to Voice over Frame Relay (VoFR) and Voice over ATM (VoATM) networks.

Dial Peer Overview

Configuring dial peers is the key to setting up dial plans and implementing voice over a packet network. Dial peers are used to identify call source and destination endpoints and to define the characteristics applied to each call leg in the call connection.

A traditional voice call over the PSTN uses a dedicated 64K circuit end to end. In contrast, a voice call over the packet network is made up of discrete segments or call legs. A call leg is a logical connection between two routers or between a router and a telephony device. A voice call comprises four call legs, two from the perspective of the originating router and two from the perspective of the terminating router, as shown in Figure 22.

Figure 22   Dial Peer Call Legs
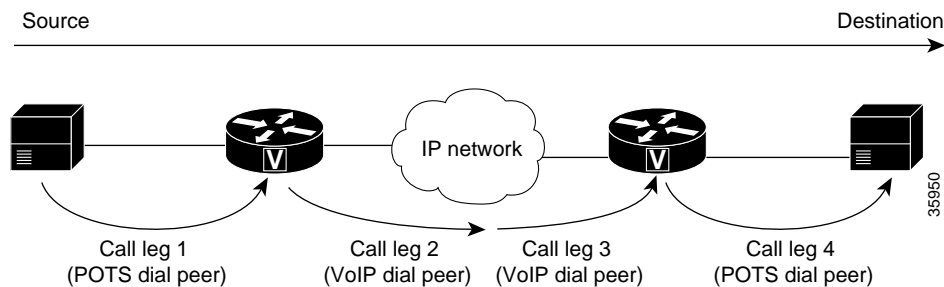
A dial peer is associated with each call leg. Attributes that are defined in a dial peer and applied to the call leg include codec, Quality of Service (QoS), voice activity detection (VAD), and fax rate. To complete a voice call, you must configure a dial peer for each of the four call legs in the call connection.

Depending on the call leg, a call is routed using one of the two types of dial peers:

- POTS—Dial peer that defines the characteristics of a traditional telephony network connection. POTS dial peers map a dialed string to a specific voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.

- Voice-network—Dial peer that defines the characteristics of a packet network connection. Voice-network dial peers map a dialed string to a remote network device, such as the destination router that is connected to the remote telephony device.

  The specific type of voice-network dial peer depends on the packet network technology:

  - VoIP (Voice over IP)—Points to the IP address of the destination router that terminates the call.

  - VoFR (Voice over Frame Relay)—Points to the data-link connection identifier (DLCI) of the interface from which the call exits the router.

  - VoATM (Voice over ATM)—Points to the ATM virtual circuit for the interface from which the call exits the router.

  - MMoIP (Multimedia Mail over IP)—Points to the e-mail address of the SMTP server. This type of dial peer is used only for fax traffic. For more information, see the chapter "Configuring Fax Applications."

Both POTS and voice-network dial peers are needed to establish voice connections over a packet network.

## Inbound and Outbound Dial Peers

Dial peers are used for both inbound and outbound call legs. It is important to remember that these terms are defined from the perspective of the router. An inbound call leg originates when an incoming call comes *to* the router. An outbound call leg originates when an outgoing call is placed *from* the router. Figure 23 illustrates call legs from the perspective of the originating router; Figure 24 illustrates call legs from the perspective of the terminating router.

Note   Figure 23 and Figure 24 apply to voice calls that are being sent across the packet network. If the originating and terminating POTS interfaces share the same router or if the call requires hairpinning, then two POTS call legs are sufficient. See Figure 29 on page 126 for more information.

Figure 23   Call Legs from the Perspective of the Originating Router
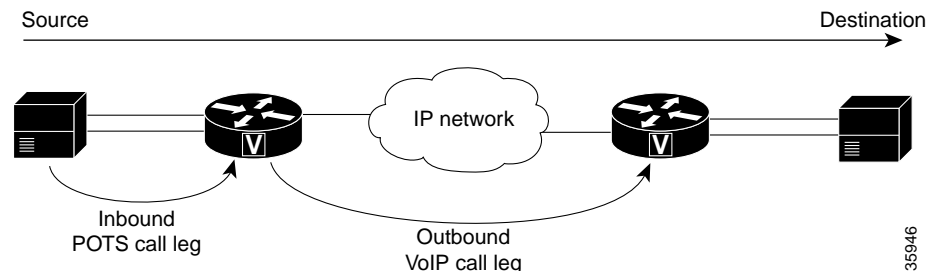
Figure 24   Call Legs from the Perspective of the Terminating Router
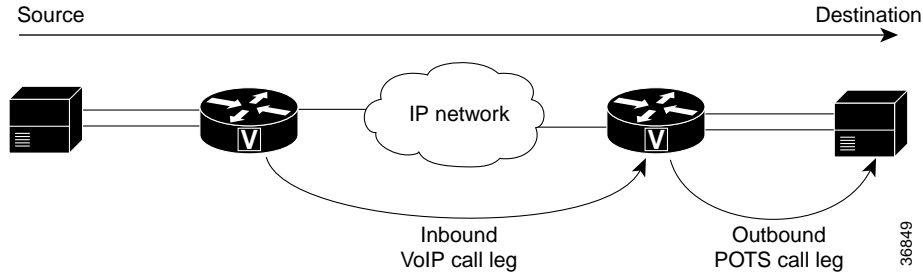
For inbound calls from a POTS interface that are destined for the packet network, the router matches a POTS dial peer for the inbound call leg and a voice-network dial peer, such as VoIP or VoFR, for the outbound leg. For inbound calls from the packet network, the router matches a POTS dial peer to terminate the call and a voice-network dial peer to apply features such as codec or QoS.

For inbound POTS call legs going to outbound voice-network dial peers, the router forwards all digits that it collects. On outbound POTS call legs, the router strips off explicitly matching digits and forwards any excess digits out the designated port. For specific information about how the router handles excess digits, see the "Two-Stage Dialing" section on page 137.

The following examples show basic configurations for POTS and VoIP dial peers:

```
dial-peer voice 1 pots
 destination-pattern 555....
 port 1/0:1

dial-peer voice 2 voip
 destination-pattern 555....
 session target ipv4:192.168.1.1
```

The router selects a dial peer for a call leg by matching the string that is defined by using the **answer-address**, **destination-pattern**, or **incoming called-number** command in the dial peer configuration. For specific information about how the router matches dial peers, see the "Dial Peer Overview" section on page 137.

Destination Pattern

The destination pattern associates a dialed string with a specific telephony device. It is configured in a dial peer by using the **destination-pattern** command. If the dialed string matches the destination pattern, the call is routed according to the voice port in POTS dial peers, or the session target in voice-network dial peers. For outbound voice-network dial peers, the destination pattern may also determine the dialed digits that the router collects and then forwards to the remote telephony interface, such as a PBX, a telephone, or the PSTN. You must configure a destination pattern for each POTS and voice-network dial peer that you define on the router.

The destination pattern can be either a complete telephone number or a partial telephone number with wildcard digits, represented by a period (.) character. Each "." represents a wildcard for an individual digit that the originating router expects to match. For example, if the destination pattern for a dial peer is defined as "555....", then any dialed string beginning with 555, plus at least four additional digits, matches this dial peer.

In addition to the period (.), there are several other symbols that can be used as wildcard characters in the destination pattern. These symbols provide additional flexibility in implementing dial plans and decrease the need for multiple dial peers in configuring telephone number ranges.

Table 11 shows the wildcard characters that are supported in the destination pattern.

Table 11   Wildcard Symbols Used in Destination Patterns

Symbol	Description
.	Indicates a single-digit placeholder. For example, 555.... matches any dialed string beginning with 555, plus at least four additional digits.
[ ]	Indicates a range of digits. A consecutive range is indicated with a hyphen (-); for example, [5-7]. A nonconsecutive range is indicated with a comma (,); for example, [5,8]. Hyphens and commas can be used in combination; for example, [5-7,9]. **Note** Only single-digit ranges are supported. For example, [98-102] is invalid.
( )	Indicates a pattern; for example, 408(555). It is used in conjunction with the symbol ?, %, or +.
?	Indicates that the preceding digit occurred zero or one time. Enter **ctrl-v** before entering ? from your keyboard.
%	Indicates that the preceding digit occurred zero or more times. This functions the same as the "*" used in regular expression.
+	Indicates that the preceding digit occurred one or more times.
T	Indicates the interdigit timeout. The router pauses to collect additional dialed digits.

Note   The period (.) is the only wildcard character that is supported for dial strings that are configured using the **answer-address** or **incoming called-number** commands.

Table 12 shows some examples of how these wildcard symbols are applied to the destination pattern and the dial string that results when dial string 4085551234 is matched to an outbound POTS dial peer. The wildcard symbols follow regular expression rules.

Table 12   Dial Peer Matching Examples Using Wildcard Symbols

	Destination Pattern	Dial String Translation	String After Stripping[1]
1	408555.+	408555, followed by one or more wildcard digits. This pattern implies that the string must contain at least 7 digits starting with 408555.	1234
2	408555.%	408555, followed by zero or more wildcard digits. This pattern implies that the string must contain at least 408555.	1234
3	408555+	40855, followed by 5 repeated one or more times.	1234
4	408555%	40855, followed by 5 repeated zero or more times. Any explicitly matching digit before the % symbol is not stripped off.	51234
5	408555?	40855, followed by 5 repeated zero or one time. Any explicitly matching digit before the ? symbol is not stripped off.	51234
6	40855[5-7].+	40855, followed by 5, 6, or 7, plus any digit repeated one or more times.	51234

Table 12   Dial Peer Matching Examples Using Wildcard Symbols (continued)

	Destination Pattern	Dial String Translation	String After Stripping[1]
7	40855[5-7].%	40855, followed by 5, 6, or 7, plus any digit repeated zero or more times.	51234
8	40855[5-7]+1234	40855, followed by 5, 6, or 7 repeated one or more times, followed by 1234.	51234
9	408(555)+1234	408, followed by 555, which may repeat one or more times, followed by 1234.	5551234

1. These examples apply only to one-stage dialing, where DID is enabled on the inbound POTS dial peer. If the router is using two-stage dialing and collecting digits one at a time as dialed, then the call is routed immediately after a dial peer is matched and any subsequent dialed digits are lost.

In addition to wildcard characters, the following characters can also be used in the destination pattern:

- Asterisk (*) and pound sign (#)—These characters on standard touch-tone dial pads can be used anywhere in the pattern. They can be used as the leading character (for example, *650), except on the Cisco 3600 series.

- Dollar sign ($)—Disables variable-length matching. Must be used at the end of the dial string.

The same destination pattern can be shared across multiple dial peers to form hunt groups. For information on building hunt groups, see the "Hunt Groups and Preferences" section on page 144.

For information on how the terminating router strips off digits after matching a destination pattern, see the "Digit Stripping on Outbound POTS Dial Peers" section on page 124.

Fixed- and Variable-Length Dial Plans

Fixed-length dialing plans, in which all the dial-peer destination patterns have a fixed length, are sufficient for most voice networks because the telephone number strings are of known lengths. Some voice networks, however, require variable-length dial plans, particularly for international calls, which use telephone numbers of different lengths.

If you enter the timeout T-indicator at the end of the destination pattern in an outbound voice-network dial peer, the router accepts a fixed-length dial string and then waits for additional dialed digits. The timeout character must be an uppercase T. The following dial-peer configuration shows how the T-indicator is set to allow variable-length dial strings:

```
dial-peer voice 1 voip
 destination-pattern 2222T
 session target ipv4:10.10.1.1
```

In the example above, the router accepts the digits 2222, and then waits for an unspecified number of additional digits. The router can collect up to 31 additional digits, as long as the interdigit timeout has not expired. When the interdigit timeout expires, the router places the call.

The default value for the interdigit timeout is 10 seconds. Unless the default value is changed, using the T-indicator adds 10 seconds to each call setup because the call is not attempted until the timer has expired (unless the # character is used as a terminator). You should therefore reduce the voice-port interdigit timeout value if you use variable-length dial plans. You can change the interdigit timeout by using the **timeouts inter-digit** voice-port command.

The calling party can immediately terminate the interdigit timeout by entering the # character. If the # character is entered while the router is waiting for additional digits, the # character is treated as a terminator; it is not treated as part of the dial string or sent across the network. But if the # character is entered before the router begins waiting for additional digits (meaning that the # is entered as part of the fixed-length destination pattern), then the # character is treated as a dialed digit.

For example, if the destination pattern is configured as 2222...T, then the entire dialed string of 2222#9999 is collected, but if the dialed string is 2222#99#99, the #99 at the end of the dialed digits is not collected because the final # character is treated as a terminator. You can change the termination character by using the **dial-peer terminator** command.

Note In most cases, you must configure the T-indicator only when the router uses two-stage dialing. If Direct Inward Dialing (DID) is configured in the inbound POTS dial peer, the router uses one-stage dialing, which means that the full dialed string is used to match outbound dial peers. The only exception is when the **ISDN overlap-receiving** command is configured; the ISDN overlap-receiving feature requires the T-indicator.

Session Target

The session target is the network address of the remote router to which you want to send a call once a local voice-network dial peer is matched. It is configured in voice-network dial peers by using the **session target** command. For outbound dial peers, the destination pattern is the telephone number of the remote voice device that you want to reach. The session target represents the path to the remote router that is connected to that voice device. Figure 25 illustrates the relationship between the destination pattern and the session target, as shown from the perspective of the originating router.

Figure 25 Relationship Between Destination Pattern and Session Target
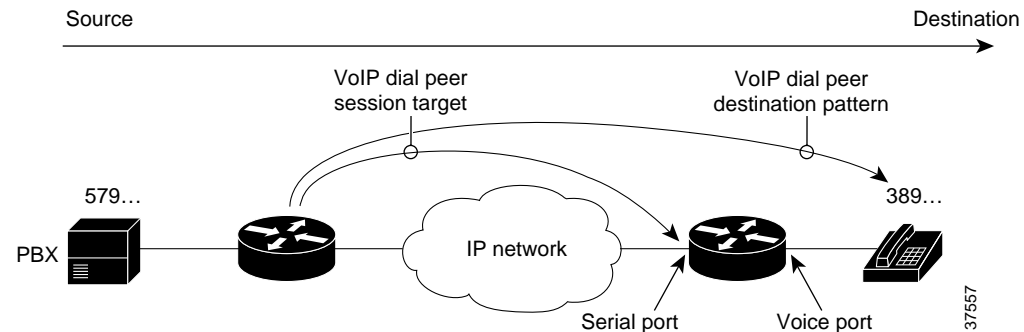
The address format of the session target depends on the type of voice-network dial peer:

- VoIP—IP address, hostname of the Domain Name System (DNS) server that resolves the IP address, **ras** for registration, admission, and status (RAS) if an H.323 gatekeeper resolves the IP address, or **settlement** if the settlement server resolves the IP address

- VoFR—Interface type and number and the DLCI

- VoATM—Interface number, and ATM virtual circuit

- MMoIP—E-mail address

Note For inbound dial peers, the session target is ignored.

Digit Stripping on Outbound POTS Dial Peers

When a terminating router receives a voice call, it selects an outbound POTS dial peer by comparing the called number (the full E.164 telephone number) in the call information with the number configured as the destination pattern in the POTS dial peer. The access server or router then strips off the left-justified digits that match the destination pattern. If you have configured a prefix, the prefix is added to the front of the remaining digits, creating a dial string, which the router then dials. If all numbers in the destination pattern are stripped out, the user receives a dial tone.

For example, consider a voice call whose E.164 called number is 1(408) 555-2222. If you configure a destination-pattern of "1408555" and a prefix of "9," the router strips off "1408555" from the E.164 telephone number, leaving the extension number of "2222." It then appends the prefix, "9," to the front of the remaining numbers, so that the actual numbers dialed are "9, 2222." The comma in this example means that the router will pause for one second between dialing the "9" and dialing the "2" to allow for a secondary dial tone.

For detailed information about digit stripping and the prefix command, see the "Digit Stripping and Prefixes" section on page 151.

Configuring Dial Peers

This section describes how to configure dial peers:

Note The example configurations in this section show VoIP dial peers; the same concepts also apply to VoFR and VoATM dial peers.

Establishing voice communication over a packet network is similar to configuring a static route: you are establishing a specific voice connection between two defined endpoints. Call legs define the discrete segments that lie between two points in the call connection. A voice call over the packet network comprises four call legs, two on the originating router and two on the terminating router; a dial peer is associated with each of these four call legs.

Configuring Dial Peers for Call Legs

When a voice call comes into the router, the router must match dial peers to route the call. For inbound calls from a POTS interface that are being sent over the packet network, the router matches a POTS dial peer for the inbound call leg and a voice-network dial peer for the outbound call leg. For calls coming into the router from the packet network, the router matches an outbound POTS dial peer to terminate the call and an inbound voice-network dial peer for features such as codec, VAD, and QoS.

Figure 26 shows the call legs and associated dial peers necessary to complete a voice call.

Figure 26   Matching Call Legs to Dial Peers
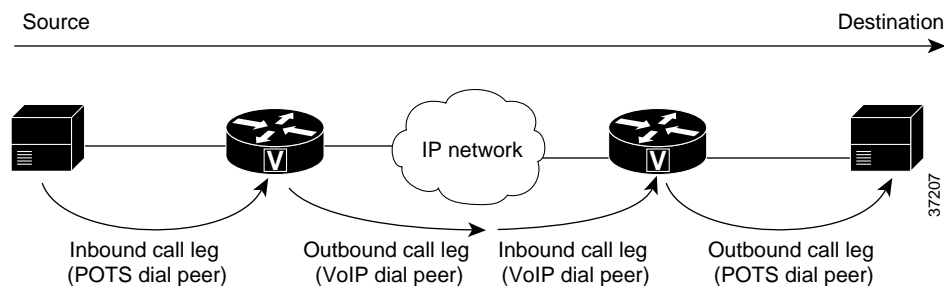
The following configurations show an example of a call being made from 4085554000 to 3105551000. Figure 27 shows the inbound POTS dial peer and the outbound VoIP dial peer that are configured on the originating router. The POTS dial peer establishes the source of the call (via the calling number or voice port), and the voice-network dial peer establishes the destination by associating the dialed number with the network address of the remote router.

Figure 27   Dial Peers from the Perspective of the Originating Router
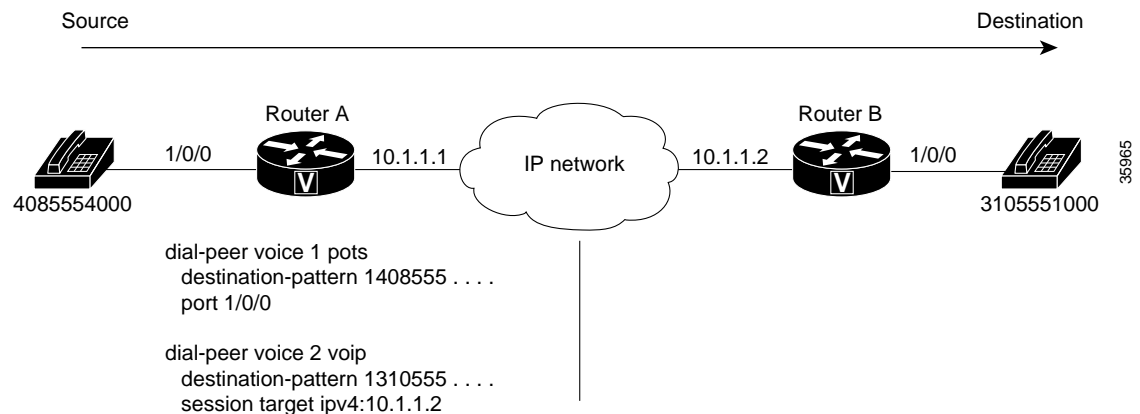
In this example, the dial string 14085554000 maps to telephone number 555-4000, with the digit 1 plus the area code 408 preceding the number. When you configure the destination pattern, set the string to match the local dialing conventions.

Figure 28 shows the inbound VoIP dial peer and outbound POTS dial peer that are configured on the terminating router to complete the call. Dial peers are of local significance only.

Figure 28   Dial Peers from the Perspective of the Terminating Router
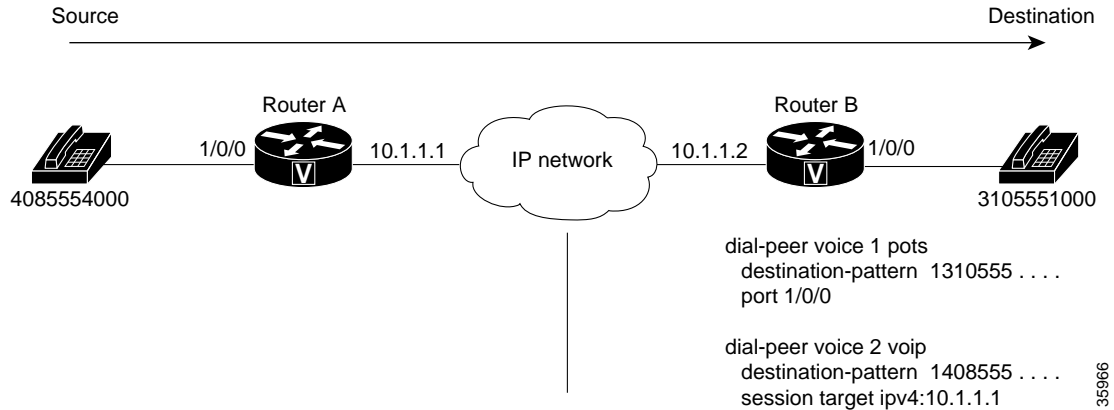

In the previous configuration examples, the last four digits in the VoIP dial peer's destination pattern were replaced with wildcards. This means that from Router A, calling any telephone number that begins with the digits "1310555" will result in a connection to Router B. This implies that Router B services all numbers beginning with those digits. From Router B, calling any telephone number that begins with the digits "1408555" will result in a connection to Router A. This implies that Router A services all numbers beginning with those digits.

> **Note**   It is not always necessary to configure the inbound dial peers. If the router is unable to match a configured dial peer for the inbound call leg, it uses an internally defined default POTS or voice-network dial peer to match inbound voice calls. In the example shown in Figure 28, dial peer 2 is only required when making a call from Router B to Router A.

The only exception to the previous example occurs when both POTS dial peers share the same router, as shown in Figure 29. In this circumstance, you do not need to configure a voice-network dial peer.

Figure 29   Communication Between Dial Peers Sharing the Same Router
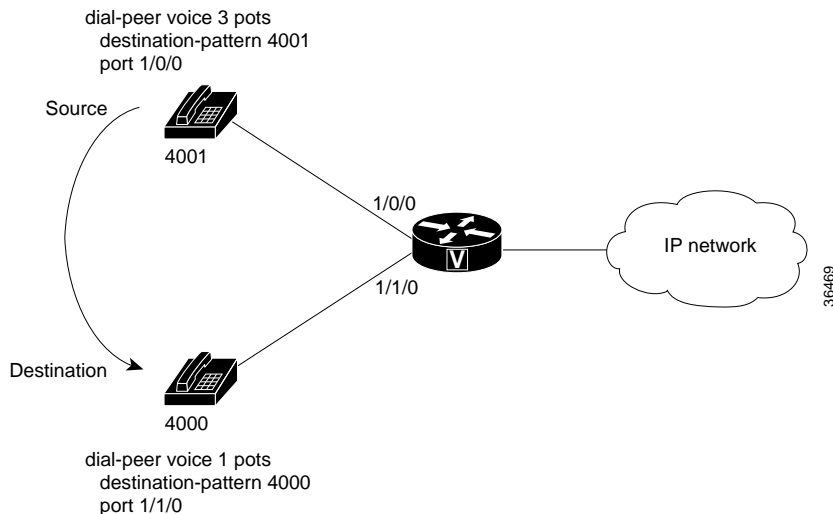
This type of configuration is similar to the configuration used for hairpinning, which occurs when a voice call destined for the packet network is instead routed back over the PSTN because the packet network is unavailable. For more information about the hairpinning feature, see the "Hunt Groups and Preferences" section on page 144.

Creating a Dial Peer Configuration Table

Before you can configure dial peers, you must obtain specific information about your network. One way to identify this information is to create a dial peer configuration table. This table should contain all the telephone numbers and access codes for each router that is carrying telephone traffic in the network. Because most installations require integrating equipment into an existing voice network, the telephone dial plans are usually preset.

Figure 30 shows an example of a network in which Router A, with an IP address of 10.1.1.1, connects a small sales branch office to the main office through Router B, with an IP address of 10.1.1.2.

Note The example in Figure 30 shows a VoIP configuration. The same concepts also apply to VoFR and VoATM applications. The only change is in the format of the session target.

Figure 30   Sample VoIP Network
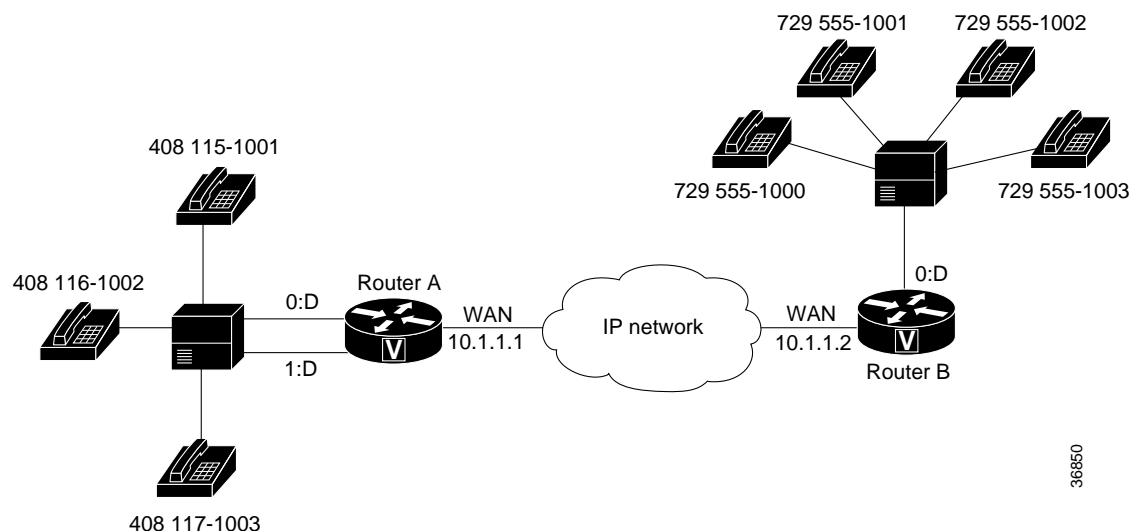
There are three telephone numbers in the sales branch office that need dial peers configured for them. Router B is the primary gateway to the main office; as such, it needs to be connected to the company's PBX. There are four devices that need dial peers configured for them in the main office, all of which are connected to the PBX.

Table 13 shows the peer configuration table for the example in Figure 30.

Table 13   Dial Peer Configuration Table for Sample Voice over IP Network

Dial Peer	Extension	Prefix	Destination Pattern	Type	Voice Port	Session Target
Router A						
1	51001	5	1408115....	POTS	0:D	—
2	61002	6	1408116....	POTS	0:D	—
3	71003	7	1408117....	POTS	0:D	—
10	—	—	1729555....	VoIP	—	10.1.1.2
Router B						
1	1000, 1001, 1002, 1003	—	1729555....	POTS	0:D	—
10	—	—	1408.......	VoIP	—	10.1.1.1

Configuring POTS Dial Peers

To configure a POTS dial peer, you must do the following:

- Identify the dial peer by assigning it a unique tag number
- Define its destination telephone number or range of telephone numbers
- Associate it with a voice port through which calls are established

Under most circumstances, the default values for the remaining dial peer configuration commands are sufficient to establish connections.

To configure a POTS dial peer, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer voice** *number* **pots**	Enters dial-peer configuration mode and defines a local dial peer that connects to a POTS interface.
		The *number* argument is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647.
		The **pots** keyword indicates a dial peer using basic telephone service.
Step 2	Router(config-dial-peer)# **destination-pattern** *string* [**T**]	Matches dialed digits to a telephony device.
		The *string* argument is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the numbers 0 through 9 and the letters A through D.
		You can also enter the following special characters:
		• The asterisk (*) or pound sign (#) on standard touch-tone dial pads can be used anywhere in the pattern.
		• The period (.) acts as a wildcard character.
		For a list of additional wildcard characters, see Table 11 on page 121.
		When the timer (T) character is included at the end of the destination pattern, the router collects dialed digits until the interdigit timer expires (10 seconds, by default) or until you dial the termination character (the default is #). The timer character must be a capital T.
Step 3	Router(config-dial-peer)# **port** *location*	Maps the dial peer to a specific logical interface.
		The **port** command syntax is platform-specific. For more information about the syntax of this command, see the chapter "Configuring Voice Ports" in this document.

Configuring Dial Plan Options for POTS Dial Peers

When you configure a dial plan, you have different options, depending on how the dial plan is designed. To configure optional dial plan features for POTS dial peers, use one or more of the following commands in dial-peer configuration mode:

Command	Purpose
`Router(config-dial-peer)# answer-address string`	(Optional) Selects the inbound dial peer based on the calling number.
`Router(config-dial-peer)# incoming called-number string`	(Optional) Selects the inbound dial peer based on the called number to identify voice and modem calls.
`Router(config-dial-peer)# direct-inward-dial string`	(Optional) Enables the Direct Inward Dialing (DID) call treatment for the incoming called number. For more information, see the "DID for POTS Dial Peers" section on page 142.
`Router(config-dial-peer)# forward-digits {num-digit | all | extra}`	(Optional) Configures the digit-forwarding method used by the dial peer. The valid range for the number of digits forwarded (*num-digit*) is 0 through 32. For more information, see the "Forward Digits" section on page 154.
`Router(config-dial-peer)# max-conn number`	(Optional) Specifies the maximum number of allowed connections to and from the POTS dial peer. The valid range is 1 through 2147483647.
`Router(config-dial-peer)# numbering-type {abbreviated | international | national | network | reserved | subscriber | unknown}`	(Optional) Specifies the numbering type to match, as defined by the ITU Q.931 specification. For more information, see the "Numbering Type Matching" section on page 147.
`Router(config-dial-peer)# preference value`	(Optional) Configures a preference for the POTS dial peer. The valid range is 0 through 10, where the lower the number, the higher the preference. For more information, see the "Hunt Groups and Preferences" section on page 144.
`Router(config-dial-peer)# prefix string`	(Optional) Includes a prefix that the system adds automatically to the front of the dial string before passing it to the telephony interface. Valid entries for the *string* argument are 0 through 9 and a comma (,). Use a comma to include a one-second pause between digits to allow for a secondary dial tone. For more information, see the "Digit Stripping and Prefixes" section on page 151.
`Router(config-dial-peer)# translate-outgoing {called | calling} name-tag`	(Optional) Specifies the translation rule set to apply to the calling number or called number. For more information, see the "Digit Translation Rules for VoIP" section on page 157.

Configuring VoIP Dial Peers

VoIP dial peers enable the router to make outbound calls to a particular telephony device. To configure a VoIP dial peer, you must do the following:

- Identify the dial peer by assigning it a unique tag number
- Define its destination telephone number
- Define its destination IP address

As with POTS dial peers, under most circumstances the default values for the remaining dial peer configuration commands are adequate to establish connections.

To configure a VoIP peer, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice number voip`	Enters dial-peer configuration mode and defines a remote VoIP dial peer.
		The *number* argument is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647.
		The **voip** keyword indicates a dial peer using voice encapsulation on the IP network.
Step 2	`Router(config-dial-peer)# destination-pattern string [T]`	Configures the dial peer's destination pattern so that the system can reconcile dialed digits with a telephone number.
		The *string* argument is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the numbers 0 through 9 and the letters A through D. You can also enter the following special characters:
		• The asterisk (*) or pound sign (#) on standard touch-tone dial pads can be used anywhere in the pattern.
		• The period (.) acts as a wildcard character.
		For a list of additional wildcard characters, see Table 11 on page 121.
		When the timer (T) character is included at the end of the destination pattern, the router collects dialed digits until the interdigit timer expires (10 seconds, by default) or until you dial the termination character (the default is #). The timer character must be a capital T.

Command	Purpose
Step 3 `Router(config-dial-peer)# ` **`session target`** `{`**`ipv4:`**`destination-address | ` **`dns:`**`[`**`$s$.`** ` | ` **`$d$.`** ` | ` **`$e$.`** ` | ` **`$u$.`**`] host-name}`	Defines the IP address of the router that is connected to the remote telephony device. The **ipv4:**_destination-address_ keyword and argument indicate the IP address of the remote router. The **dns:**_host-name_ keyword and argument indicate that the domain name server will resolve the name of the IP address. Valid entries for this parameter are characters representing the name of the host device. Wildcards are also available for defining domain names with the keyword by using source, destination, and dialed information in the host name.
Step 4 `Router(config-dialpeer)# ` **`codec`** ` {`**`g711alaw`** ` | ` **`g711ulaw`** ` | ` **`g723ar53`** ` | ` **`g723ar63`** ` | ` **`g723r53`** ` | ` **`g723r63`** ` | ` **`g726r16`** ` | ` **`g726r24`** ` | ` **`g726r32`** ` | ` **`g728`** ` | ` **`g729br8`** ` | ` **`g729r8`** ` [`**`pre-ietf`**`]} [`_bytes_`]`	Defines the codec for the dial peer. The optional _bytes_ parameter sets the number of voice data bytes per frame. Acceptable values are from 10 to 240 in increments of 10 (for example, 10, 20, 30, and so on). Any other value is rounded down (for example, from 236 to 230). The same codec value must be configured in both VoIP dial peers on either side of the connection. If you specify **g729r8**, then IETF bit-ordering is used. For interoperability with a Cisco 2600 series, Cisco 3600 series, or Cisco AS5300 running a release earlier than Cisco IOS Release 12.0(5)T or 12.0(4)XH, you _must_ specify the additional keyword **pre-ietf** after **g729r8**. The **codec** command syntax is platform- and release-specific. For more information about the syntax of this command, refer to the _Cisco IOS Voice, Video, and Fax Command Reference_.

If you have used the **codec complexity** voice-card interface configuration command, the **codec** command sets the codec options that are available. If you do not set codec complexity, **g729r8** with IETF bit-ordering is used. For more information about the **codec complexity** command, see the "Configuring Voice Ports" chapter.

Configuring Codec Selection Order

You can create a voice class in which you define a selection order for codecs, and then apply the voice class to VoIP dial peers. The **voice class codec** global configuration command allows you to define the voice class containing the codec selection order. Then you use the **voice-class codec** dial-peer configuration command to apply the class to individual dial peers.

To configure codec selection order, perform the tasks described in the following sections:

- Creating a Voice Class to Define Codec Selection Order
- Applying Codec Selection Order to a VoIP Dial Peer

Creating a Voice Class to Define Codec Selection Order

To create a voice class to define the order of preference for selecting a codec when the router negotiates with a destination router, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	router(config)# **voice class codec** *tag*	Creates a voice class for a codec preference list. The range for the *tag* number is from 1 through 10000. The *tag* number must be unique on the router.
Step 2	router(config-voice-class)# **codec preference** *priority codec* [**bytes** *payload-size*]	Configures the order of preference for selecting a codec. Repeat this command to specify the preferred selection order for additional codecs, if required.

Applying Codec Selection Order to a VoIP Dial Peer

To apply voice-class codec attributes to a VoIP dial peer, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	router(config)# **dial-peer voice** *tag* **voip**	Defines a VoIP dial peer and enters dial-peer configuration mode. The *tag* is a number that identifies the dial peer and must be unique on the router.
Step 2	router(config-dialpeer)# **voice-class codec** *tag*	Assigns to the dial peer the voice class that you created in the "Creating a Voice Class to Define Codec Selection Order" section. The **voice-class** command in dial-peer configuration mode is entered with a hyphen. The **voice class** command in global configuration mode is entered without the hyphen.

✎
Note   You cannot assign voice-class codec attributes to POTS dial peers.

Configuring Dial Plan Options for VoIP Dial Peers

When you configure a dial plan, you have different options, depending on how the dial plan is designed. To configure optional dial plan features, use the following commands in dial-peer configuration mode:

	Command	Purpose
Step 1	Router(config-dial-peer)# **answer-address** *string*	(Optional) Selects the inbound dial peer based on the calling number.
Step 2	Router(config-dial-peer)# **incoming called-number** *string*	(Optional) Selects the inbound dial peer based on the called number to identify voice and modem calls.

Step 3	Router(config-dial-peer)# **dtmf-relay** [**cisco-rtp**] [**h245-signal**] [**h245-alphanumeric**]	(Optional) Configures the tone that sounds in response to a keypress on a touch-tone telephone. Dual tone multifrequency (DTMF) tones are compressed at one end of a call and decompressed at the other end.
		If a low-bandwidth codec such as G.729 or G.723 is used, the tones can sound distorted. The **dtmf-relay** command transports DTMF tones generated after call establishment out-of-band by using a method that sends with greater fidelity than is possible in-band for most low-bandwidth codecs. Without DTMF Relay, calls established with low-bandwidth codecs can have trouble accessing automated telephone menu systems such as voice mail and interactive voice response (IVR) systems.
		A signaling method is supplied only if the remote end supports it. Options are Cisco proprietary (**cisco-rtp**), standard H.323 (**h245-alphanumeric**), and H.323 standard with signal duration (**h245-signal**).
Step 4	Router(config-dial-peer)# **fax rate** {**2400** | **4800** | **7200** | **9600** | **12000** | **14400** | **disable** | **voice**}	(Optional) Specifies the transmission speed of a fax to be sent to this dial peer. The **disable** keyword turns off fax transmission capability. The **voice** keyword, which is the default, specifies the highest possible transmission speed supported by the voice rate.
Step 5	Router(config-dial-peer)# **numbering-type** {**abbreviated** | **international** | **national** | **network** | **reserved** | **subscriber** | **unknown**}	(Optional) Specifies the numbering type to match, as defined by the ITU Q.931 specification. For more information, see the "Numbering Type Matching" section on page 147.
Step 6	Router(config-dial-peer)# **playout-delay mode** {**adaptive** | **fixed**}	(Optional) Specifies the type of jitter buffer playout delay to use.
Step 7	Router(config-dial-peer)# **playout-delay** {**maximum** *value* | **nominal** *value* | **minimum** {**default** | **low** | **high**}}	(Optional) Specifies the amount of time that a packet is held in the jitter buffer before it is played out on the audio path. For detailed information, see the chapter "Quality of Service" in this document.
Step 8	Router(config-dial-peer)# **preference** *value*	(Optional) Configures a preference for the VoIP dial peer. The value is a number from 0 through 10, where the lower the number, the higher the preference. For more information, see the "Hunt Groups and Preferences" section on page 144.
Step 9	Router(config-dial-peer)# **tech-prefix** *number*	(Optional) Specifies that a particular technology prefix be prepended to the destination pattern of this dial peer.

Step 10	`Router(config-dial-peer)# translate-outgoing {called | calling} name-tag`	(Optional) Specifies the translation rule set to apply to the calling number or called number. For more information, see the "Digit Translation Rules for VoIP" section on page 157.
Step 11	`Router(config-dial-peer)# vad`	(Optional) Enables voice activity detection (VAD) by disabling the transmission of packets during periods of silence. VAD is enabled by default.
		The minimum silence detection time for VAD can be modified by using the **voice vad-time** global configuration command.

The default for the **vad** command is enabled, which is normally the preferred configuration. If you are operating on a high-bandwidth network and voice quality is of the highest importance, you should disable VAD by using the **no vad** command. This results in better voice quality, but also requires higher bandwidth for voice. For example, a broad industry average for VAD savings on links T1 and up is from 30 to 35 percent of the overall bandwidth.

Note   The music threshold that is configured by using the **music-threshold** voice-port command can affect VAD performance.

Some codecs come with built-in VAD algorithms (specifically, G.729 Annex B and G.723.1 symmetric). VAD can be used with all other codecs.

Configuring VoFR Dial Peers

To configure VoFR dial peers, see the "Configuring Voice over Frame Relay" chapter.

Configuring VoATM Dial Peers

To configure VoATM dial peers, see the "Configuring Voice over ATM" chapter.

Verifying POTS and VoIP Dial Peer Configurations

You can check the validity of your dial peer configuration by performing the following tasks:

*   If you have relatively few dial peers configured, you can use the **show dial-peer voice** command to verify that the configuration is correct. To display a specific dial peer or to display all configured dial peers, use this command. The following is sample output from the **show dial-peer voice** command for a specific VoIP dial peer:

```
router# show dial-peer voice 10

VoiceOverIpPeer10
        tag = 10, dest-pat = \Q',
        incall-number = \Q+14087',
        group = 0, Admin state is up, Operation state is down
        Permission is Answer,
        type = voip, session-target = \Q',
        sess-proto = cisco, req-qos = bestEffort,
        acc-qos = bestEffort,
```

```
                     fax-rate = voice, codec = g729r8,
                     Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
                     Connect Time = 0, Charged Units = 0
                     Successful Calls = 0, Failed Calls = 0
                     Accepted Calls = 0, Refused Calls = 0
                     Last Disconnect Cause is ""
                     Last Disconnect Text is ""
                     Last Setup Time = 0
```

- To show the dial peer that matches a particular number (destination pattern), use the **show dialplan number** command. The following example displays the VoIP dial peer associated with the destination pattern 51234:

```
router# show dialplan number 51234

Macro Exp.: 14085551234
VoiceOverIpPeer1004
        tag = 1004, destination-pattern = \Q+1408555....',
        answer-address = \Q',
        group = 1004, Admin state is up, Operation state is up
        type = voip, session-target = \Qipv4:1.13.24.0',
        ip precedence: 0        UDP checksum = disabled
        session-protocol = cisco, req-qos = best-effort,
        acc-qos = best-effort,
        fax-rate = voice, codec = g729r8,
        Expect factor = 10, Icpif = 30,
        VAD = enabled, Poor QOV Trap = disabled
        Connect Time = 0, Charged Units = 0
        Successful Calls = 0, Failed Calls = 0
        Accepted Calls = 0, Refused Calls = 0
        Last Disconnect Cause is ""
        Last Disconnect Text is ""
        Last Setup Time = 0
Matched: +14085551234   Digits: 7
Target: ipv4:172.13.24.0
```

Troubleshooting Tips

You can troubleshoot your dial peer configurations by performing the following tasks:

- Ping the associated IP address to confirm connectivity. If you cannot successfully ping your destination, refer to the *Cisco IOS IP Configuration Guide*.

- To verify that the operational status and administrative status of the dial peer is up, use the **show dial-peer voice** command.

> ✎
>
> **Note** To activate a dial peer, the **answer-address**, **incoming called-number**, or **destination-pattern** with **port** or **session-target** command must be configured in the dial peer.

- To verify that the data is configured correctly on both routers, use the **show dialplan number** command on the local and remote routers.

- If you have configured number expansion, use the **show num-exp** command to check that the partial number on the local router maps to the correct full E.164 telephone number on the remote router.

- If you have configured translation rules, use the **test translation-rule** command to verify digit manipulation.

- If you have configured a codec value, make sure that the same codec value is configured in both VoIP dial peers on either side of the connection. You can verify the configured codec value by using the **show dial-peer voice** or **show dialplan number** command.

- To verify that the output string the router dials is correct, use the **debug voip ccapi inout** command.

- To check Real-Time Transport Protocol (RTP) packets, use the **debug cch323 rtp** command.

- To check logical channel negotiation, use the **debug cch323 h245** command.

- To check the call setup, use the **debug cch323 h225** command.

Dial Peer Overview

Before setting up a dial plan, you should understand how the router matches dialed strings to inbound and outbound dial peers. How the router matches dialed strings directly affects the digits that your users have to dial, in addition to the digits that are collected and then forwarded or played out to the telephony interface, such as a PBX, key system, or PSTN.

The following sections describe basic concepts on how the router selects a matching dial peer:

Note Unless otherwise noted, the concepts described in this section apply to VoIP, VoFR, and VoATM dial peers.

Two-Stage Dialing

With two-stage dialing, when a voice call enters the network, the originating router collects dialed digits until it can match an outbound dial peer. As soon as the router matches a dial peer, it immediately places the call and forwards the associated dial string. No additional dialed digits are collected. The digits and wildcards that are defined in the destination pattern determine how many digits the originating router collects before matching the dial peer. Any digits dialed after the first dial peer is matched are dropped.

For example, if the dialed string is "1234599" and the originating router matches a dial peer with a destination pattern of 123. ., then the digits "99" are not collected. The call is placed immediately after the digit "5" is dialed, and the dial string "12345" is forwarded to the next call leg.

On the terminating router, the left-justified digits that explicitly match the terminating POTS dial peer are stripped off. Any trailing wildcard digits are considered excess digits. The terminating router forwards these excess digits to the telephony interface. For example, if the dial string "1234599" is matched on a terminating router to a destination pattern of "123. .," the digits "4599" are excess digits and are forwarded to the telephony interface.

Figure 31 illustrates how the originating router collects a dial string and the terminating router forwards the digits to the telephony device.

Figure 31 Collecting and Forwarding Dialed Digits
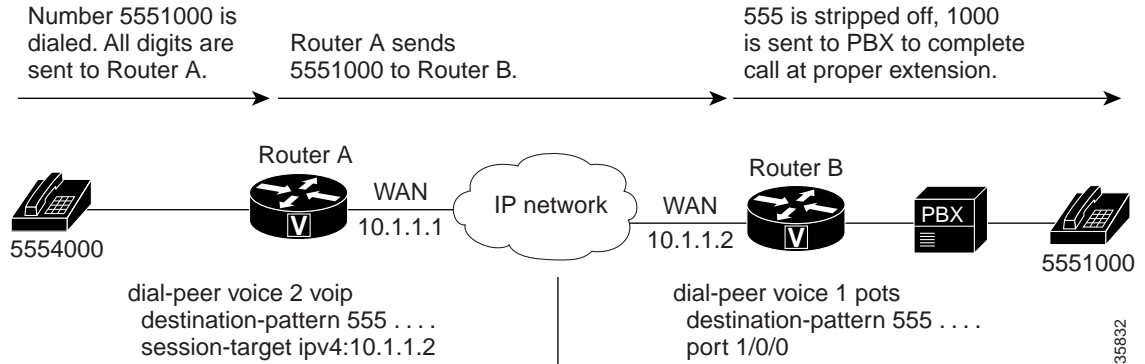
The examples in Table 14 demonstrate how the originating router collects dialed digits for a given destination pattern in the outbound voice-network dial peer.

Table 14 Digit Collection Based on Destination Pattern

Dialed Digits	Destination Pattern	Dial String Collected[1]
5551234	5......	5551234
5551234	555....	5551234
5551234	555	555
555123499	555....	5551234

1. These examples apply only to two-stage dialing, in which the router collects the dialed string digit by digit. If DID is enabled in the inbound POTS dial peer, the router performs one-stage dialing, which means that the full dialed string is used regardless of the destination pattern that is matched.

The router defaults to two-stage dialing unless you configure DID. For information on configuring DID, see the "DID for POTS Dial Peers" section on page 142.

Variable-Length Matching

When matching dial peers, the router defaults to variable-length matching, which means that as long as the left-justified digits in the dial string match the configured pattern in the dial peer, any digits beyond the configured pattern are ignored for the purposes of matching. For example, dial string 5551212 would match both of the following dial peers:

```
dial-peer voice 1 voip
 destination-pattern 555
 session target ipv4:10.10.1.1

dial-peer voice 2 voip
 destination-pattern 5551212
 session target ipv4:10.10.1.2
```

To disable variable-length matching for a dial peer, add the dollar sign ($) to the end of the destination pattern, as shown:

```
dial-peer voice 1 voip
 destination-pattern 555$
 session target ipv4:10.10.1.1
```

The $ character in the above configuration prevents this dial peer from being matched for dial string 5551212 because the extra digits beyond 555 are considered in the matching.

With two-stage dialing, the router collects the dialed string digit by digit. It attempts to match a dial peer after each digit is received. As soon as it finds a match, it immediately routes the call. For example, given the following configurations, the router would immediately match dial string 5551212 to dial peer 1.

```
dial-peer voice 1 voip
 destination-pattern 555
 session target ipv4:10.10.1.1

dial-peer voice 2 voip
 destination-pattern 5551212
 session target ipv4:10.10.1.2
```

If the router is performing two-stage dialing and you want to make sure that the full dial string is collected before a dial peer is matched, you can use the timeout T-indicator as in variable-length dial plans. For example, after the router waits until the full dial string is collected, dial string 5551212 would match both of the following dial peers:

```
dial-peer voice 1 voip
 destination-pattern 555T
 session target ipv4:10.10.1.1

dial-peer voice 2 voip
 destination-pattern 5551212T
 session target ipv4:10.10.1.2
```

How the router selects a dial peer also depends on whether the dial peer is being matched for the inbound or outbound call leg. For more information, see the and the .

Matching Inbound Dial Peers

To match inbound call legs to dial peers, the router uses three information elements in the call setup message and four configurable dial peer attributes. The three call setup elements are:

- Called number or dialed number identification service (DNIS)—A set of numbers representing the destination, which is derived from the ISDN setup message or CAS DNIS.
- Calling number or automatic number identification (ANI)—A set of numbers representing the origin, which is derived from the ISDN setup message or CAS ANI.
- Voice port—The voice port carrying the call.

The four configurable dial peer attributes are:

- Incoming called-number—A string representing the called number or DNIS. It is configured by using the **incoming called-number** dial-peer configuration command in POTS or MMoIP dial peers. For more information, see the .
- Answer address—A string representing the calling number or ANI. It is configured by using the **answer-address** dial-peer configuration command in POTS or VoIP dial peers and is used only for inbound calls from the IP network. For more information, see the .

- Destination pattern—A string representing the calling number or ANI. It is configured by using the **destination-pattern** dial-peer configuration command in POTS or voice-network dial peers. For more information, see the "Destination Pattern" section on page 120.

- Port—The voice port through which calls to this dial peer are placed.

The router selects an inbound dial peer by matching the information elements in the setup message with the dial peer attributes. The router attempts to match these items in the following order:

1. Called number with **incoming called-number**

2. Calling number with **answer-address**

3. Calling number with **destination-pattern**

4. Incoming voice port with configured voice port

The router must match only one of these conditions. It is not necessary for all the attributes to be configured in the dial peer or that every attribute match the call setup information; only one condition must be met for the router to select a dial peer. The router stops searching as soon as one dial peer is matched and the call is routed according to the configured dial peer attributes. Even if there are other dial peers that would match, only the first match is used.

Note For a dial peer to be matched, its administrative state must be up. The dial peer administrative state is up by default when it is configured with at least one of these commands: **incoming called-number**, **answer-address**, or **destination-pattern**. If **destination-pattern** is used, the voice port or session target must also be configured.

Inbound Dial Peers for IVR Applications

To identify an interactive voice response (IVR) application to handle inbound calls, the originating router must match a POTS dial peer. You configure which IVR application handles incoming voice calls by using the **application** dial-peer configuration command. If the router is unable to match an inbound dial peer, or if the inbound dial peer does not specify an application, the default application handles the call. The following configuration shows an example of specifying an IVR application for an inbound POTS call leg:

```
dial-peer voice 571 pots
 application tr6
 destination-pattern 5714954
 port 0:D
```

Matching Outbound Dial Peers

How the router selects an outbound dial peer depends on whether DID is configured in the inbound POTS dial peer. If DID is not configured in the inbound POTS dial peer, the router collects the incoming dialed string digit by digit. As soon as one dial peer is matched, the router immediately places the call using the configured attributes in the matching dial peer.

If DID is configured in the inbound POTS dial peer, the router uses the full incoming dial string to match the destination pattern in the outbound dial peer. With DID, the setup message contains all the digits necessary to route the call; no additional digit collection is required. If more than one dial peer matches the dial string, all of the matching dial peers are used to form a rotary group. The router attempts to place the outbound call leg using all of the dial peers in the rotary group until one is successful. For more information on rotary groups, see the "Hunt Groups and Preferences" section on page 144.

For information on configuring DID, see the "DID for POTS Dial Peers" section on page 142.

Default Routes for Outbound Call Legs

Default routes reduce the number of dial peers that must be configured when calls that are not terminated by other dial peers are sent to a central router, usually for forwarding to a PBX. A default route is a dial peer that automatically matches any call that is not terminated by other dial peers. For example, in the following configuration, the destination pattern 8... is a voice default route because all voice calls with a dialed string that starts with 8 followed by at least three additional digits will either match on 8208 or end up with 8..., which is the last-resort voice route used by the router if no other dial peer is matched.

```
dial-peer voice 8 pots
 destination-pattern 8208
 port 1/1
!
dial-peer voice 1000 pots
 destination-pattern 8...
 port 1/1
```

A default route could also be defined by using a single wildcard character with the timeout T-indicator in the destination pattern, as shown in the following example:

```
dial-peer voice 1000 voip
 destination-pattern .T
 session-target ipv4:10.10.1.2
```

You should be careful, however, when using the T-indicator for default routes. Remember, when matching dial peers for outbound call legs, the router places the call as soon as it finds the first matching dial peer. The router could match on this dial peer immediately even if there were another dial peer with a more explicit match and a more desirable route.

> **Note**   The timeout T-indicator is appropriate only for two-stage dialing. If the router is configured for one-stage dialing, which means that DID is configured in the inbound POTS dial peer, then the timeout T-indicator is unnecessary.

Configuring Dial Peer Matching Features

You can define the attributes that the router uses to match dial peers by configuring specific dial peer features. These dial peer matching features are described in the following sections:

- Answer Address for VoIP, page 142
- DID for POTS Dial Peers, page 142
- Identifying Voice and Modem Calls, page 144
- Hunt Groups and Preferences, page 144

Note Unless otherwise noted, the concepts described in this section apply to VoIP, VoFR, and VoATM dial peers.

Answer Address for VoIP

The **answer-address** command can be used to select the inbound dial peer for VoIP calls, instead of using the destination pattern. If the **answer-address** command is configured in VoIP or POTS dial peers, the router attempts to match the calling number to the string configured as the answer address before attempting to match a destination pattern in any dial peer. The following dial peer would match any inbound VoIP call that had a calling number of 5551212.

```
dial-peer voice 2 voip
 answer-address 5551212
 session target ipv4:192.168.1.1
```

For more information, see the "Matching Inbound Dial Peers" section on page 139.

Note The **answer-address** command is not supported for VoFR or VoATM dial peers.

DID for POTS Dial Peers

The Direct Inward Dialing (DID) feature in dial peers enables the router to use the called number (DNIS) to directly match an outbound dial peer when receiving an inbound call from a POTS interface. When DID is configured on the inbound POTS dial peer, the called number (DNIS) is automatically used to match the destination pattern for the outbound call leg.

Unless otherwise configured, when a voice call comes into the router, the router presents a dial tone to the caller and collects digits until it can identify an outbound dial peer. This process is called *two-stage dialing*. After the outbound dial peer is identified, the router forwards the call through to the destination as configured in the dial peer.

You may prefer that the router use the called number (DNIS) to find a dial peer for the outbound call leg—for example, if the switch connecting the call to the router has already collected all the dialed digits. DID enables the router to match the called number to a dial peer and then directly place the outbound call. With DID, the router does not present a dial tone to the caller and does not collect digits; it forwards the call directly to the configured destination. This is called *one-stage dialing*.

Figure 32 shows a call scenario using DID.

Figure 32   VoIP Call Using DID
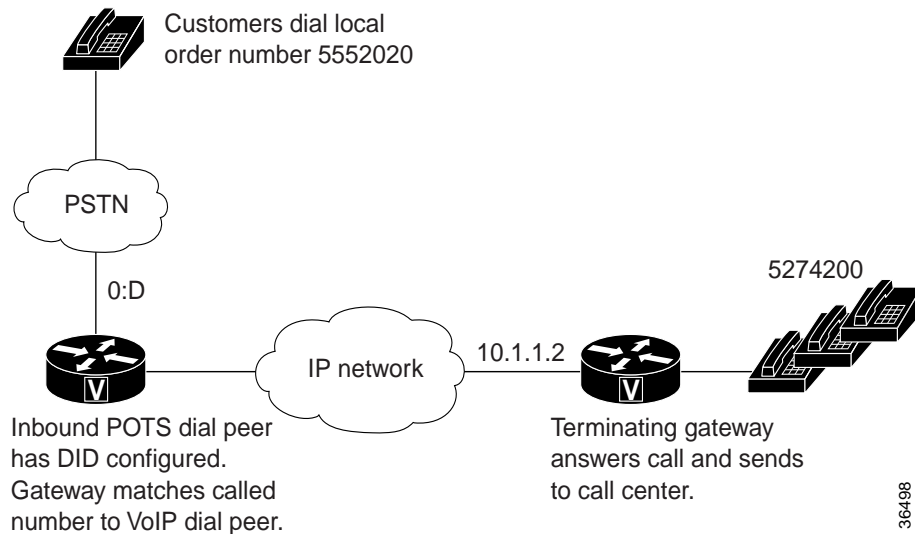
In Figure 32, the POTS dial peer that matches the incoming called-number has direct-inward-dial configured:

```
dial-peer voice 100 pots
 incoming called-number 5552020
 direct-inward-dial
 port 0:D
```

The **direct-inward-dial** command in the POTS dial peer tells the gateway to look for a destination pattern in a dial peer that matches the DNIS. For example, if the dialed number is 5552020, the gateway matches the following VoIP dial peer for the outbound call leg:

```
dial-peer voice 101 voip
 destination-pattern 5552020
 session target ipv4:10.1.1.2
```

The call is made across the IP network to 10.1.1.2, and a match is found in that terminating gateway:

```
dial-peer voice 555 pots
 destination-pattern 5552020
 port 0:D
 prefix 5274200
```

This dial peer matches on the dialed number and changes that number to 52744200 with the **prefix** command. The result is that the user dials a number, gets connected, and never knows that the number reached is different from the number dialed.

✎

Note   DID for POTS dial peers is not the same as analog DID for Cisco routers which enables DID trunk service from the PSTN.

To configure a POTS dial peer for DID, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer voice** *number* **pots**	Enters dial-peer configuration mode and defines a local dial peer that will connect to the POTS network. The *number* is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647.
Step 2	Router(config-dial-peer)# **direct-inward-dial**	Specifies DID for this POTS dial peer.

Note DID is configured for inbound POTS dial peers only.

Identifying Voice and Modem Calls

When a Cisco router is handling both modem and voice calls, it needs to identify the service type of the call—that is, whether the incoming call to the router is a modem or a voice call. When the router handles only modem calls, the service type identification is handled through modem pools. Modem pools associate calls with modem resources based on the called number (DNIS). In a mixed environment, where the router receives both modem and voice calls, you need to identify the service type of a call by using the **incoming called-number** command.

If the **incoming called-number** command is not configured, the router attempts to resolve whether an incoming call is a modem or voice call on the basis of the interface over which the call comes. If the call comes in over an interface associated with a modem pool, the call is assumed to be a modem call; if a call comes in over a voice port associated with a POTS dial peer, the call is assumed to be a voice call.

To identify the service type of a call as voice, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer voice** *number* {**pots** | **voip** | **vofr** | **voatm**}	Enters dial peer configuration mode.
Step 2	Router(config-dial-peer)# **incoming called-number** *number*	Defines the telephone number that identifies voice calls associated with this dial peer.

Hunt Groups and Preferences

The router supports the concept of *hunt groups*, sometimes called *rotary groups*, in which multiple dial peers are configured with the same destination pattern. Because the destination of each POTS dial peer is a single voice port to a telephony interface, hunt groups help ensure that calls get through even when a specific voice port is busy. If the router is configured to hunt, it can forward a call to another voice port when one voice port is busy.

For example, in the following configuration for Router A, four POTS dial peers are configured with different destination patterns. Because each dial peer has a different destination pattern, no backup is available if the voice port mapped to a particular dial peer is busy with another call.

With a hunt group, if a voice port is busy, the router hunts for another voice port until it finds one that is available. In the following example for Router B, each dial peer is configured using the same destination pattern of 3000, forming a dial pool to that destination pattern.

Router A (Without Hunt Groups)	Router B (With Hunt Groups and Preferences)
| ```
dial-peer voice 1 pots
 destination-pattern 3001
 port 1/1
!
dial-peer voice 2 pots
 destination-pattern 3002
 port 1/2
!
dial-peer voice 3 pots
 destination-pattern 3003
 port 1/3
!
dial-peer voice 4 pots
 destination-pattern 3004
 port 1/4
``` | ```
dial-peer voice 1 pots
 destination pattern 3000
 port 1/1
 preference 0
!
dial-peer voice 2 pots
 destination pattern 3000
 port 1/2
 preference 1
!
dial-peer voice 3 pots
 destination pattern 3000
 port 1/3
 preference 2
!
dial-peer voice 4 pots
 destination pattern 3000
 port 1/4
 preference 3
``` |

To give specific dial peers in the pool a preference over other dial peers, you can configure the preference order for each dial peer by using the **preference** command. The router attempts to place a call to the dial peer with the highest preference. The configuration example given for Router B shows that all dial peers have the same destination pattern, but different preference orders.

The lower the preference number, the higher the priority. The highest priority is given to the dial peer with preference order 0. If the same preference is defined in multiple dial peers with the same destination pattern, a dial peer is selected randomly.
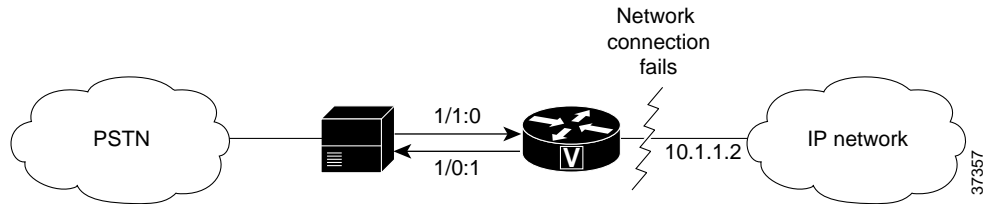
By default, dial peers in a hunt group are selected according to the following criteria, in the order listed:

1. Longest match in phone number—Destination pattern that matches the greatest number of dialed digits. For example, if one dial peer is configured with a dial string of 345 . . . . and a second dial peer is configured with 3456789, the router would first select 3456789 because it has the longest explicit match of the two dial peers.

2. Explicit preference—Priority configured by using the **preference** dial peer command.

3. Random selection—All destination patterns weighted equally.

You can change this default selection order or choose different methods for hunting dial peers by using the **dial-peer hunt** global configuration command. An additional selection criteria is "least recent use," which selects the destination pattern that has waited the longest since being selected.

You can mix POTS and voice-network dial peers when creating hunt groups. This can be useful if you want incoming calls to be sent over the packet network, except that if network connectivity fails, you want to reroute the calls back through the PBX to the PSTN. This type of configuration is sometimes referred to as *hairpinning*. Hairpinning is illustrated in Figure 33.

Figure 33   Voice Call Using Hairpinning

The following configuration shows an example of sending calls to the PSTN if the IP network fails:

```
dial-peer voice 101 voip
 destination-pattern 472....
 session target ipv4:192.168.100.1
 preference 0
!
dial-peer voice 102 pots
 destination-pattern 472....
 prefix 472
 port 1/0:1
 preference 1
```

You cannot use the same preference numbers for POTS and voice-network dial peers within a hunt group. You can set a separate preference order for each dial peer type, but the preference order does not work on both at the same time. For example, you can configure preference order 0, 1, and 2 for POTS dial peers, and you can configure preference order 0, 1, and 2 for the voice-network dial peers, but the two preference orders are separate. The system resolves preference orders among POTS dial peers first.

Configuring Dial-Peer Hunting Options

Dial-peer hunting is enabled by default. To disable dial-peer hunting on a dial peer, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer voice** *number* {**pots** | **vofr** | **voip**}	Enters dial-peer configuration mode for the specified dial peer.
Step 2	Router(config-dial-peer)# **huntstop**	(Optional) Disables dial-peer hunting on the dial peer. Once you enter this command, no further hunting is allowed if a call fails on the selected dial peer.

Use the **no huntsto**p command to reenable dial-peer hunting if it has been disabled.

To configure dial peer hunting options for all dial peers, use the following commands in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer hunt` *hunt-order-number*	(Optional) Specifies the hunt selection order for dial peers in a hunt group. Valid entries are 0 through 7. The default is 0. • 0—Longest match in phone number, explicit preference, random selection • 1—Longest match in phone number, explicit preference, least recent use • 2—Explicit preference, longest match in phone number, random selection • 3—Explicit preference, longest match in phone number, least recent use • 4—Least recent use, longest match in phone number, explicit preference • 5—Least recent use, explicit preference, longest match in phone number • 6—Random selection • 7—Least recent use
Step 2	`Router(config)# voice hunt {user-busy	invalid-number	unassigned-number}`	(Optional) Defines how the originating or tandem router handles rotary dial-peer hunting if it receives a disconnect cause code from the terminating router. • **user-busy** sets the router to continue dial-peer hunting if it receives a user-busy disconnect cause code from a destination router. • **invalid-number** sets the router to stop dial-peer hunting if it receives a an invalid-number disconnect cause code from a destination router. • **unassigned-number** sets the router to stop dial-peer hunting if it receives an unassigned-number disconnect cause code from a destination router.

Numbering Type Matching

A dial peer can be selected according to the type of number field in the called party number or calling party number information element, in addition to matching the dial peer based on the configured destination pattern, answer address, or incoming called number. The type of number value is selected by using the **numbering-type** dial-peer configuration command.

For example, in the following configuration, the dialed string "4085559999" would match this dial peer if the type of number field for the called party number is "national."

```
dial-peer voice 408 voip
 numbering-type national
 destination-pattern 408.......
 session target ipv4:10.1.1.2
```

The following numbering types can be used:

- Abbreviated—Abbreviated representation of the complete number as supported by this network
- International—Number called to reach a subscriber in another country
- National—Number called to reach a subscriber in the same country, but outside the local network
- Network—Administrative or service number specific to the serving network
- Reserved—Reserved for extension
- Subscriber—Number called to reach a subscriber in the same local network
- Unknown—Type of number is unknown by the network

For detailed information about these numbering types, see ITU-T Recommendation Q.931

Configuring Numbering-Type Matching

To configure numbering-type matching for a dial peer call leg, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice` *number* `{pots	voip	vofr	voatm}`	Enters dial peer configuration mode.
Step 2	`Router(config-dial-peer)# numbering-type {abbreviated	international	national	network	reserved	subscriber	unknown}`	Specifies the numbering type to match, as defined by the ITU Q.931 specification.

Note To match a dial peer using the **numbering-type** command, you must also configure the **destination-pattern**, **answer-address**, or **incoming called-number** command.

Class of Restrictions

The Class of Restrictions (COR) feature provides the ability to deny certain call attempts based on the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, to 900 numbers), and applies different restrictions to call attempts from different originators.

COR is used to specify which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list. The incoming COR list indicates the capability of the dial peer to initiate certain classes of calls. The outgoing COR list indicates the capability required for an incoming dial peer to deliver a call via this outgoing dial peer. If the capabilities of the incoming dial peer are not the same or a superset of the capabilities required by the outgoing dial peer, the call cannot be completed using this outgoing dial peer.

A typical application of COR is to define a COR name for the number that an outgoing dial peer serves, then define a list that contains only that COR name, and assign that list as **corlist outgoing** for this outgoing dial peer. For example, dial peer with destination pattern 5T can have a **corlist outgoing** that contains COR 5x, as shown in the following configuration.

The next step, in the typical application, is to determine how many call permission groups are needed, and define a COR list for each group. For example, group A is allowed to call 5x and 6x, and group B is allowed to call 5x, 6x, and 1900x. Then, for each incoming dial peer, we can assign a group for it, which defines what number an incoming dial peer can call. Assigning a group means assigning a **corlist incoming** to this incoming dial peer.

```
dial-peer cor custom
 name 5x
 name 6x
 name 1900x
!
dial-peer cor list listA
 member 5x
 member 6x
!
dial-peer cor list listB
 member 5x
 member 6x
 member 1900x
!
dial-peer cor list list5x
 member 5x
!
dial-peer cor list list6x
 member 6x
!
dial-peer cor list list1900x
 member 1900x

! outgoing dialpeer 100, 200, 300
dial-peer voice 100 pots
 destination-pattern 5T
 corlist outgoing list5x
dial-peer voice 200 pots
 destination-pattern 6T
 corlist outgoing list6x
dial-peer voice 300 pots
 destination-pattern 1900T
 corlist outgoing list1900x
!
! incoming dialpeer 400, 500
dial-peer voice 400 pots
 answer-address 525....
 corlist incoming listA
dial-peer voice 500 pots
 answer-address 526
 corlist incoming listB
```

Configuring Classes of Restrictions

To configure classes of restrictions for dial peers, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer cor custom**	Enters COR configuration mode to specify classes of restrictions to apply to dial peers.
Step 2	Router(config-dp-cor)# **name** *class-name*	Provides a name for a class of restrictions.
		Note Repeat this step for additional class names, as needed. These class names are used to define the COR lists configured in Step 4 and Step 5.
Step 3	Router(config-dp-cor)# **exit**	Exits COR configuration mode.
Step 4	Router(config)# **dial-peer cor list** *list-name*	Provides a name for a list of restrictions.
Step 5	Router(config-dp-corlist)# **member** *class-name*	Adds a COR class to this list of restrictions.
		The member is a class named in Step 2.
		Note Repeat Step 4 and Step 5 to define another list and its membership, as needed.
Step 6	Router(config-dp-corlist)# **exit**	Exits COR-list configuration mode.
Step 7	Router(config)# **dial-peer voice** *number* {**pots** | **voip**}	Enters dial-peer configuration mode and defines a dial peer.
Step 8	Router(config-dial-peer)# **corlist incoming** *cor-list-name*	Specifies the COR list to be used when this is the incoming dial peer.
Step 9	Router(config-dial-peer)# **corlist outgoing** *cor-list-name*	Specifies the COR list to be used when this is the outgoing dial peer.
		Note Repeat Step 7 through Step 9 for additional dial peers, as needed.

Verifying Classes of Restrictions

To check the validity of your classes of restrictions configuration, perform the following tasks:

- Enter the **show dial-peer voice** command to learn whether the COR list fields are set as desired on a dial peer:

```
Router# show dial-peer voice 210

VoiceEncapPeer210
        information type = voice,
        tag = 210, destination-pattern = `221',
        answer-address = `', preference=0,
        numbering Type = `unknown'
        group = 210, Admin state is up, Operation state is up,
        incoming called-number = `221', connections/maximum = 4/unlimited,
        DTMF Relay = disabled,
        Modem = system passthrough ,
        huntstop = disabled,
        application associated:
        permission :both
        incoming COR list:maximum capability
        outgoing COR list:minimum requirement
```

```
                    type = pots, prefix = `221',
                    forward-digits default
                    session-target = `', voice-port = `1/0/8:D',
                    direct-inward-dial = enabled,
                    digit_strip = enabled,
```

- Enter the **show dial-peer cor** command to display the COR names and lists you defined:

```
Router# show dial-peer cor

Class of Restriction
  name:900block
  name:800_call
  name:Catchall

COR list <list1>
  member:900block
  member:800_call

COR list <list2>
  member:900block

COR list <list3>
  member:900block
  member:800_call
  member:Catchall
```

Configuring Digit Manipulation

The router may need to manipulate digits in a dial string before it passes the dial string to the telephony device. This can be necessary, for instance, when calling PBXs with different capabilities to accept digits, or for PSTN and international calls. You may need to consider different strategies for configuring digit manipulation within your dial peers depending on your existing dial plan, the digits users are expected to dial, and the capabilities of your PBX or key system unit (KSU). These digit-manipulation options, in conjunction with the destination pattern, determine the dial string that the router forwards to the telephony device.

The following dial peer digit-manipulation options are described in this section:

Note   Unless otherwise noted, these concepts apply to VoIP, VoFR, and VoATM networks.

Digit Stripping and Prefixes

When the terminating router matches a dial string to an outbound POTS dial peer, by default the router strips off the left-justified digits that explicitly match the destination pattern. Any remaining digits, called *excess digits,* are forwarded to the telephony interface, such as a PBX or the PSTN. For more information about excess digits, see the "Two-Stage Dialing" section on page 137.

Some telephony interfaces require that any digits that are stripped from the dial string must be recovered to support a particular dial plan. You can accomplish this either by using the **no digit-strip** dial-peer configuration command to disable the default digit-stripping behavior or by using the **prefix** dial-peer configuration command to add digits to the front of the dial string before it is forwarded to the telephony interface. These commands are supported only in POTS dial peers.

The **no digit-strip** command disables the automatic digit-stripping function so that matching digits are not stripped from the dialed string before it is passed to the telephony interface. For example, in the following dial peer configuration, the entire seven-digit dialed string is passed to the telephony interface:

```
dial-peer voice 100 pots
 destination-pattern 555....
 no digit-strip
 port 1/0:1
```

Disabling digit stripping is useful when the telephony interface requires the full dialed string. With some dial plans, however, the dialed digits must be manipulated according to specific rules. The **prefix** command can be used to add specific digits to the front of the dialed string before it is forwarded to the telephony interface.

For example, consider a telephone whose E.164 called number is 1(408)555-1234. This telephone can be reached within the company by dialing its extension number, 51234. If you configure a destination pattern of "1408555 . . . ." (the periods represent wildcards) for the associated outbound POTS dial peer, the terminating gateway will strip off the digits "1408555" when it receives a call for 1(408)555-1234. For the terminating gateway to forward the call to the appropriate destination, the digit "5" needs to be prepended to the remaining digits. In this case, you would configure a prefix of 5, as shown in the following dial peer configuration.
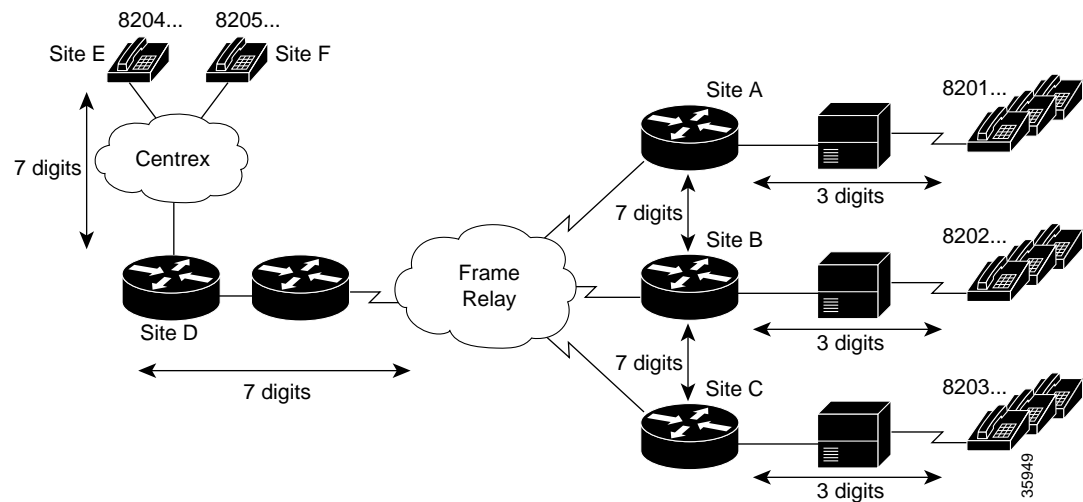
```
dial-peer voice 100 pots
 destination-pattern 1408555....
 prefix 5
 port 1/0:1
```

A prefix can also include commas (,). Each comma indicates a one-second pause in dialing. For example, consider a telephone whose E.164 called number is 1(408)555-1234; to reach this device, you must dial "9." In this case, you might configure "1408 . . . ." as the destination pattern, and "9" as the prefix. In this example, the terminating router will strip the digits "1408" from the called number and append the digit "9" to the front of the remaining digits, so that the actual number dialed is" 9,5551234." The router pauses for one second between dialing the "9" and the "5551234" to allow for a secondary dial tone. In this example, you would configure the router as follows:

```
dial-peer voice 100 pots
 destination-pattern 1408.......
 prefix 9,
 port 1/0:1
```

Using a comma with the **prefix** command is useful when the router must allow for a secondary dial tone; otherwise the router does not wait for the dial tone before playing out excess digits. Putting commas in the prefix makes the router pause one second per comma, allowing for a dial tone to occur before the router plays out the digits.

Figure 34 shows an example of a network using the **no digit-strip** command. In this example, a central site (Site D) is connected to remote sites through routers (Sites A, B, and C), as well as through a Centrex system for sites still using the PSTN (Sites E and F). The Centrex service requires the full 7-digit dial string to complete calls. The dial peers are configured with a fixed-length 7-digit dial plan.

Figure 34   Network with Digit Stripping Disabled or Prefixes Enabled

When Site E (8204 . . .) dials 8201999, the full 7-digit dialed string is passed through the Centrex to the router at Site D. Router D matches the destination pattern 8201 . . . and forwards the 7-digit dial string to Router A. Router A matches the destination pattern 8201 . . . , strips off the matching 8201, and forwards the remaining 3-digit dial string to the PBX. The PBX matches the correct station and completes the call to the proper extension.

Calls in the reverse direction are handled similarly, but because the Centrex service requires the full 7-digit dial string to complete calls, the POTS dial peer at Router D is configured with digit stripping disabled. Alternatively, digit stripping could be enabled and the dial peer could instead be configured with a 4-digit prefix, in this case 8204, which would result in forwarding the full dial string to the Centrex service.

Router A	Router D
| ```
dial-peer voice 1 pots
 destination-pattern 8201...
 port 1/0:1
!
dial-peer voice 4 vofr
 destination-pattern 8204...
 session target s0 2
!
dial-peer voice 5 vofr
 destination-pattern 8205...
 session target s0 2
!
``` | ```
dial-peer voice 4 pots
 destination-pattern 8204...
 no digit-strip
 port 1/0:1
!
dial-peer voice 5 pots
 destination-pattern 8205...
 no digit-strip
 port 1/0:1
!
dial-peer voice 1 vofr
 destination-pattern 8201...
 session target s0 1
!
``` |
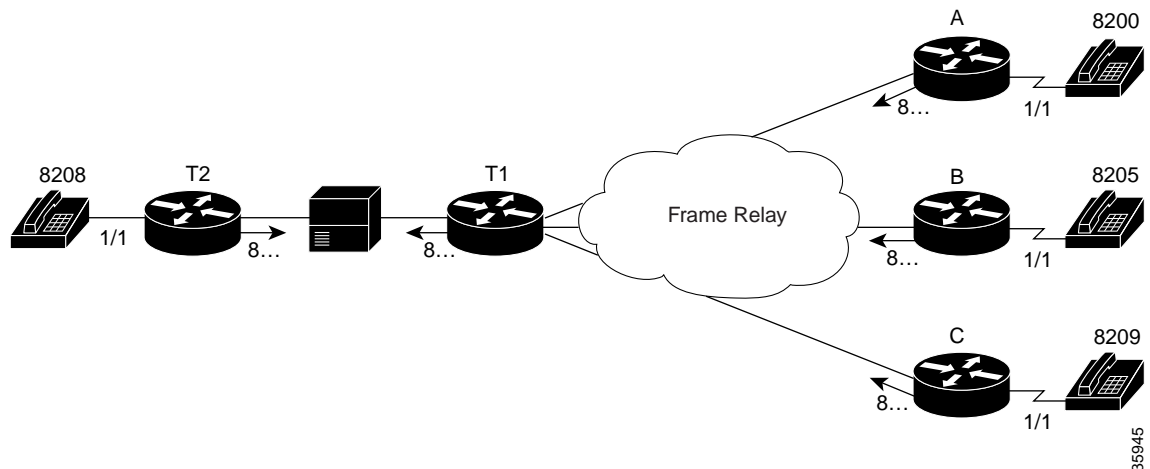
Forward Digits

The **forward-digits** command controls the number of digits that are stripped before the dialed string is passed to the telephony interface. On outbound POTS dial peers, the terminating router normally strips off all digits that explicitly match the destination pattern in the terminating POTS dial peer. Only digits matched by the wildcard pattern are forwarded. The **forward-digits** command can be used to forward a fixed number of dialed digits, or all dialed digits, regardless of the number of digits that explicitly match the destination pattern.

For example, the **forward-digits 4** command tells the router to forward the last four digits in the dialed string. The **forward-digits all** command instructs the router to forward the full dialed string. If the length of the dialed string is longer than the length of the destination pattern, the **forward-digits extra** command forwards the extra trailing digits. Extra digits are not forwarded, however, if the dial-peer destination pattern is variable length; for example, 123T, 123 . . . T.

The **forward-digits** command is supported only in POTS dial peers.

Figure 35 shows an example of routing voice calls through a PBX using forward digits. In this configuration, Routers T1 and T2 are tandem nodes that must support forward digits so that calls from Routers A, B, or C can make a call to extension 8208.

Figure 35   Routing Voice Calls Through a PBX Using Forward Digits

In this example, all digits matched with destination 8 . . . are forwarded to the appropriate port. For a call from Router A to reach extension 8208, the call first terminates at Router T1, which plays out the digits 8208 to the voice port connected to the PBX. The PBX then routes the voice call to Router T2. The **forward-digits all** command is used here, but the **forward-digits 4** command could also be used in this example.

The following dial peer configurations are required on each router for this example:

Router T1	Router T2
| ```
dial-peer voice 1 vofr
 destination-pattern 8200
 session-target s0 1
!
dial-peer voice 6 vofr
 destination-pattern 8205
 session-target s0 6
!
dial-peer voice 10 vofr
 destination-pattern 8209
 session-target s0 10
!
dial-peer voice 1 pots
 destination-pattern 8...
 forward-digits all
 port 1/1
``` | ```
dial-peer voice 8 pots
 destination-pattern 8208
 port 1/1
!
dial-peer voice 1000 pots
 destination-pattern 8...
 forward-digits all
 port 1/1
!
dial-peer voice 9999 pots
 destination-pattern ....
 forward-digits all
 port 1/1
``` |

Router A
| ```
dial-peer voice 1 pots
 destination-pattern 8200
 port 1/1
!
dial-peer voice 1000 vofr
 destination-pattern 8...
 session-target s0 1
``` |

Number Expansion

In most corporate environments, the telephone network is configured so that you can reach a destination by dialing only a portion (an extension number) of the full E.164 telephone number. You can define an extension number as the destination pattern for a dial peer. The router can be configured to recognize the extension number and expand it into its full E.164 dialed number when the **num-exp** global configuration command is used with the **destination-pattern** dial-peer configuration command.

Number expansion is a globally applied rule that enables you to define a set of digits for the router to prepend to the beginning of a dialed string before passing it to the remote telephony device. This reduces the number of digits that a user must dial to reach a remote location. Number expansion is similar to using a prefix, except that number expansion is applied globally to all dial peers.

Using a simple telephony-based example, suppose that John works in a company where employees extensions are reached by dialing the last four digits of the full E.164 telephone number. The E.164 telephone number is 555-2123; John's extension number is 2123. Suppose that every employee on John's floor has a telephone number that begins with the same first four digits: 5552. You could define each dial peer's destination pattern using each extension number, and then use number expansion to prepend the first four digits onto the extension. In this example, the router could be configured as follows:

```
num-exp 2... 5552...
!
dial peer voice 1 pots
 destination pattern 2123
```

Number expansion can also be used to replace a dialed number with another number, as in the case of call forwarding. Suppose that for some reason, John needs to have all of his telephone calls forwarded to another number, 555-6611. In this example, you would configure the router as follows:

```
num-exp 2123 5556611
!
dial peer voice 1 pots
 destination pattern 5556611
```

In this example, every time the device receives a call for extension 2123, the dialed digits will be replaced with 555-6611 and the call will be forwarded to that telephone.

Before you configure the **num-exp** command, it is helpful to map individual telephone extensions to their full E.164 dialed numbers. This task can be done easily by creating a number expansion table.

Creating a Number Expansion Table

Figure 36 shows a network for a small company that wants to use VoIP to integrate its telephony network with its existing IP network. The destination patterns (or expanded telephone numbers) associated with Router A are 408 115-xxxx, 408 116-xxxx, and 408 117-xxxx, where xxxx identifies the individual dial peers by extension. The destination pattern (or expanded telephone number) associated with Router B is 729 555-xxxx.

Figure 36 VoIP Example for Number Expansion

Table 15 shows the number expansion table for this scenario. The information included in this example must be configured on both Router A and Router B.

Table 15   Sample Number Expansion Table

Extension	Destination Pattern	Num-Exp Command Entry
5....	408115....	num-exp 5.... 408115....
6....	408116....	num-exp 6.... 408116....
7....	408117....	num-exp 7.... 408117....
1...	729555....	num-exp 1... 729555....

The period (.) character represents wildcards (such as extension numbers) in a telephone number.

Configuring Number Expansion

To expand an extension number into its full telephone number, use the following command in global configuration mode:

Command	Purpose
Router(config)# **num-exp** *extension-number expanded-number*	Configures number expansion globally for all dial peers.
	The *extension-number* argument defines the extension number to expand into the full telephone number that is specified by the *expanded-number* argument.
	The *expanded-number* argument defines the full telephone number or destination pattern to which the extension number is expanded.

Verifying Number Expansion

You can check the validity of your number expansion configuration by performing the following tasks:

- Enter the **show num-exp** command to confirm that you have mapped the telephone numbers correctly.
- Enter the **show dialplan number** command to see how a telephone number maps to a dial peer.

Digit Translation Rules for VoIP

Digit translation rules are used to manipulate the calling number (ANI) or called number (DNIS) digits for a voice call, or to change the numbering type of a call. Translation rules are used to convert a telephone number into a different number before the call is matched to an inbound dial peer or before the call is forwarded by the outbound dial peer. For example, within your company you may dial a five-digit extension to reach an employee at another site. If the call is routed through the PSTN to reach the other site, the originating gateway must use translation rules to convert the five-digit extension into the 10-digit format that is recognized by the central office switch.

Translation rules are defined by using the **translation-rule** command. After you define a set of translation rules, you can apply the rules to all inbound VoIP calls, to all inbound calls that terminate at a specific voice port, and to individual inbound or outbound call legs according to the dial peer.

Note Digit translation rules are not supported for inbound SIP calls.

The following example shows a dial peer that is configured to use translation-rule set 1, which contains ten translation rules. The first rule defined is rule 0, in which 910 is the pattern that must be matched and replaced, and 0 is the pattern that is substituted for 910.

```
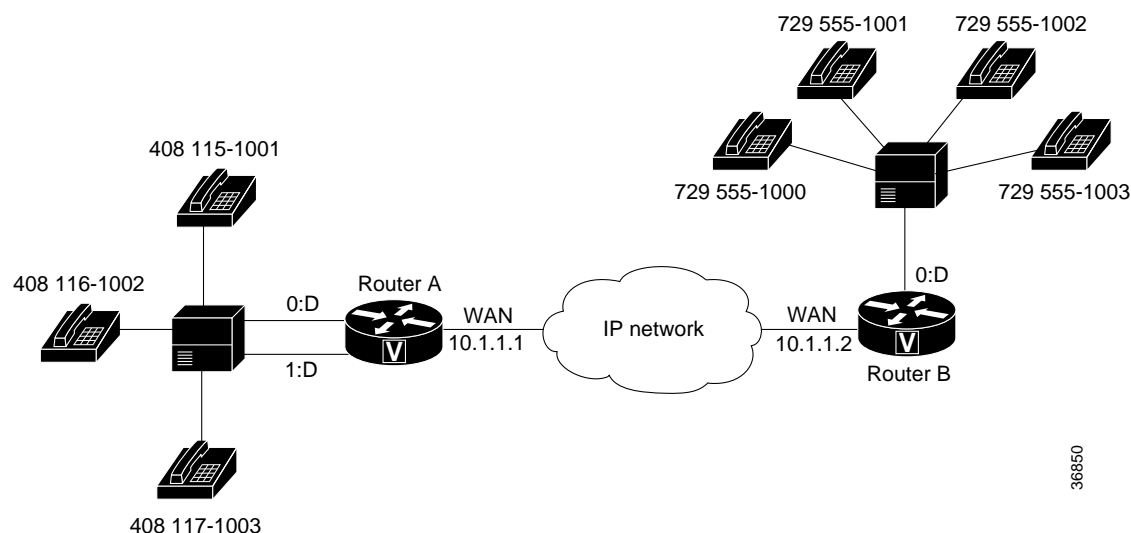translation-rule 1
 rule 0 ^910 0
 rule 1 ^911 1
 rule 2 ^912 2
 rule 3 ^913 3
 rule 4 ^914 4
 rule 5 ^915 5
 rule 6 ^916 6
 rule 7 ^917 7
 rule 8 ^918 8
 rule 9 ^919 9
!
!
dial-peer voice 2 voip
 destination-pattern 91..........
 translate-outgoing called 1
 session target ras
```

The configuration above results in the stripping of the leading digits 91 from any called number that begins with 91 before the number is forwarded by the outbound VoIP dial peer. Use the caret (^) symbol to specify that the matched digits must occur at the start of a dial string.

Note Wildcard symbols such as the period (.), asterisk (*), percent sign (%), plus sign (+), and question mark (?) are not valid in translation rules. The router simply ignores these symbols when converting a number if they are used in a translation rule.

Translation rules can also be used to change the numbering type for a call. For example, some gateways may tag any number with more than 11 digits as an international number, even when the user must dial a 9 to reach an outside line. The following example shows a translation rule that converts any called number that starts with 91, and that is tagged as an international number, into a national number without the 9 before sending it to the PSTN.

```
translation-rule 20
 rule 1 91 1 international national
!
!
dial-peer voice 10 pots
 destination-pattern 91..........
 translate-outgoing called 20
 port 1:D
!
```

Note Using digit translation rules with the **num-exp** or **prefix** command is not recommended unless it is the only way to minimize confusion.

Configuring Digit Translation Rules

To create digit translation rules, perform the tasks in the following procedure:

- Creating Digit Translation Rules (Required)

To apply digit translation rules to VoIP calls, perform one or more of the following procedures:

- Applying Translation Rules to Inbound POTS Calls (Optional)
- Applying Translation Rules to Inbound VoIP Calls (Optional)
- Applying Translation Rules to Outbound Call Legs (Optional)

Creating Digit Translation Rules

To enter translation-rule configuration mode and specify a set of translation rules, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **translation-rule** *name-tag*	Defines a digit translation-rule set and enters translation-rule configuration mode. All subsequent commands that you enter in this mode before you exit will apply to this translation-rule set.
		The *name-tag* argument represents a unique number that identifies the set of translation rules. Valid entries are from 1 to 2147483647.
Step 2	Router(config-translate)# **rule** *name-tag input-matched-pattern substituted-pattern* [*match-type substituted-type*]	Defines an individual translation rule. This command can be entered up to 11 times to add an individual translation rule to the translation rule set defined in Step 1.
		The *name-tag* argument represents a unique number that identifies this individual translation rule. Valid entries are from 0 to 10.
		The *input-matched-pattern* argument defines the digit string that must be matched, and then replaced with the *substituted-pattern*. The *substituted-pattern* argument defines the digit string that replaces the *input-matched-pattern*.
		The optional *match-type* argument defines the numbering-type that you want to replace with the numbering-type defined in *substituted-type*. Enter **any** for the *match-type* if you want to match on any numbering-type.

Command	Purpose
	Otherwise, enter one of the following keywords for each of these arguments: • **abbreviated** • **international** • **national** • **network** • **reserved** • **subscriber** • **unknown** For a description of these numbering-types, see the "Numbering Type Matching" section on page 147.

To create additional individual translation rules to include in the translation-rule set, repeat Step 2.

> **Note** Applying translation rules to more than one call leg in an end-to-end call is not recommended.

Applying Translation Rules to Inbound POTS Calls

To apply a translation rule set to all inbound POTS calls that terminate on the same voice port, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **voice-port** *location*	Specifies the voice port through which the call enters the router. The **voice-port** command syntax is platform-specific. For more information about the syntax of this command, see the "Voice Port Configuration" chapter.
Step 2	Router(config-voiceport)# **translate** {**called** | **calling**} *name-tag*	Specifies the translation rule set to apply to the called number or calling number. The **called** keyword applies the translation rule to the called party number. The **calling** keyword applies the translation rule to the calling party number. The *name-tag* argument is the reference number of the translation rule. Valid entries are 1 through 2147483647.

> **Note** When this method is used, the digit translation rules are executed first before the inbound POTS dial peer is matched.

Applying Translation Rules to Inbound VoIP Calls

To apply a translation rule set to all inbound VoIP calls that originate at an H.323 gateway, use the following command in global configuration mode:

Command	Purpose
Router(config)# **voip-incoming translation-rule** {**called** | **calling**} *name-tag*	Specifies the translation rule set to apply to all inbound VoIP call legs that originate from an H.323 gateway.
	The **called** keyword applies the translation rule to the called party number. The **calling** keyword applies the translation rule to the calling party number.
	The *name-tag* argument is the reference number of the translation rule. Valid entries are 1 through 2147483647.

Note When using this method, the digit translation rules are executed first before the inbound VoIP dial peer is matched.

Note Digit translation rules are not supported for inbound session initiation protocol (SIP) calls.

Applying Translation Rules to Outbound Call Legs

To apply a translation rule set to an outbound VoIP or POTS call leg, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# **dial-peer voice** *number* **voip**	Enters dial-peer configuration mode to configure a VoIP dial peer.
	or	
	Router(config)# **dial-peer voice** *number* **pots**	Enters dial-peer configuration mode to configure a POTS dial peer.
Step 2	Router(config-dial-peer)# **translate-outgoing** {**called** | **calling**} *name-tag*	Specifies the translation rule set to apply to the calling number or called number.
		The **called** keyword applies the translation rule to the called party number. The **calling** keyword applies the translation rule to the calling party number.
		The *name-tag* argument is the reference number of the translation rule. Valid entries are 1 through 2147483647.

> **Note**
> Translation rules that are configured in a dial peer using the **translate-outgoing** command are not applied to inbound call legs. When using two-stage dialing, the translation rules that are configured in the voice port using the **translate** command are applied twice; after the inbound dial peer is matched, and again after the digits are collected.

> **Note**
> If the **prefix** command is also configured in the dial peer, the **translate-outgoing** command is executed first.

Verifying Digit Translation

To verify the configuration of a digit translation rule, enter the **show translation-rule** EXEC command. The following example shows output for a specific translation rule:

```
Router# show translation-rule 10

Translation rule address: 0x62C4F4B0
Tag name: 10
Translation rule in_used 1
**** Xrule rule table *******
        Rule : 1
        in_used state: 1
        Match pattern: 555.%
        Sub  pattern: 1408555
        Match  type: subscriber
        Sub type: international
**** Xrule rule table *******
        Rule : 2
        in_used state: 1
        Match pattern: 91.%
        Sub  pattern: 1
        Match  type: international
        Sub type: national
**** Xrule rule table *******
        Rule : 3
        in_used state: 1
        Match pattern: 527.%
        Sub  pattern: 1408527
        Match  type: subscriber
        Sub type: international
```

To verify whether a digit translation rule functions as expected, enter the **test translation-rule** EXEC command. The following example shows that when translation rule 10 is used, the number 5551212 is converted to 14085551212:

```
Router# test translation-rule 10 5551212

The replaced number: 14085551212
```