



Dial Peer Features and Configuration

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.



Note The example configurations in this section show voice over IP (VoIP) dial peers; the same concepts also apply to voice over Frame Relay (VoFR) and voice over ATM (VoATM) dial peers.

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Common Practices

The following three sections cover the bare essential configuration steps necessary to support voice transmission and reception on a typical voice gateway router in your network:

Voice Ports

Your dial peer configuration cannot function until you have logically assigned a voice port to one or more dial peers. Assigning voice ports to dial peers identifies the physical hardware in the router that will be employed to complete voice communication to and from associated voice network endpoints.

Assigning Voice Ports

The purpose of this task is to assign a voice port to a plain old telephone system (POTS) dial peer.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **dial-peer voice** *number* **pots**
4. **port** *string*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> pots Example: Router(config)# dial-peer voice 864 pots	Enters dial-peer voice configuration mode and defines a local POTS dial peer. <ul style="list-style-type: none"> • The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	port <i>string</i> Example: Router(config-dialpeer)# port 1/0:0	Specifies the voice port associated with the given dial peer. The port command syntax is platform-specific. For more information about the syntax of this command, refer to the port command in the <i>Cisco IOS Voice, Video, and Fax Command Reference</i> .

What to do next

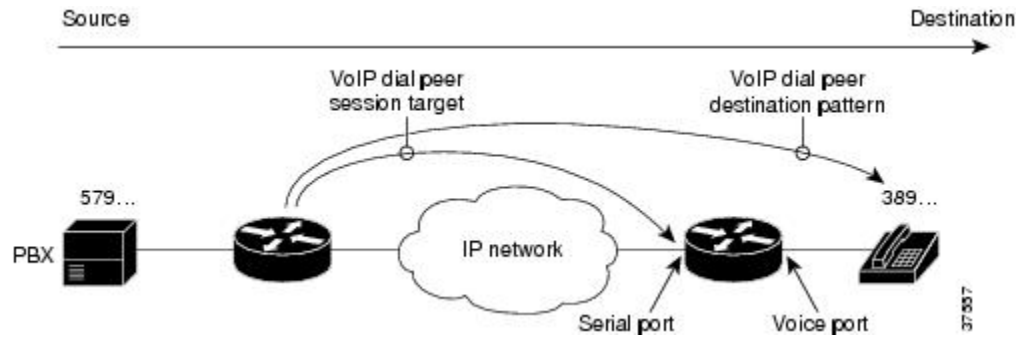


Note Voice port assignments are configured for POTS dial peers only.

Session Targets

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. The figure below illustrates the relationship between the destination pattern and the session target, as shown from the perspective of the originating router.

Figure 1: 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, host name 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 data link connection identifier (DLCI)
- VoATM--Interface number, and ATM virtual circuit
- MMoIP--E-mail address



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

Configuring Session Targets

The purpose of this task is to assign a session target to a voice-network dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip** | **vofr** | **voatm**
4. **session-target** *ip-address*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example:	Enters global configuration mode.

	Command or Action	Purpose
	Router# configure terminal	
Step 3	dial-peer voice <i>number</i> voip vofr voatm Example: Router(config)# dial-peer voice 864 voip	Enters dial-peer voice configuration mode and defines a local dial peer. <ul style="list-style-type: none"> The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	session-target <i>ip-address</i> Example: Router(config-dialpeer)# session-target 10.45.44.43	Defines the IP address identifying the next-hop location of the voice network component associated with this dial peer.

Destination Patterns

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 public switched telephone network (PSTN). You must configure a destination pattern for each POTS and voice-network dial peer that you define on the router.

Configuring Destination Patterns

The purpose of this task is to configure a destination pattern for a dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **pots** | **voip** | **vofr** | **voatm**
4. **destination-pattern** +] *string*[T]

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	dial-peer voice <i>number</i> pots voip vofr voatm Example: <pre>Router(config)# dial-peer voice 123 voip</pre>	Enters dial-peer voice configuration mode and defines a local dial peer. <ul style="list-style-type: none"> • The <i>number</i> argument identifies the dial peer. Valid entries are from 1-to 2147483647.
Step 4	destination-pattern +] <i>string</i> [T] Example: <pre>Router(config-dialpeer)# destination-pattern 5551234</pre>	Defines the phone number that identifies the destination pattern associated with this dial peer. The keywords and argument are as follows: <ul style="list-style-type: none"> • + --(Optional) Character indicating an E.164 standard number. <p>Note With CSCub65380, behavior of dial peers with destination-patterns configured with + symbol was rectified. The + symbol is no longer dropped from the dial peer and matching occurs as expected.</p> • <i>string</i> --A series of digits specifying the E.164 or private dial plan phone number. Valid entries are as follows: <ul style="list-style-type: none"> • Digits 0 through 9, letters A through F, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit. • T --(Optional) Control character indicating that the answer-address value is a variable-length dial string.

Digit Manipulation

The router may need to manipulate digits in a dial string before it passes the dial string to the telephony device. Which 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.

Wildcards

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 (.), several other symbols 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.

The table below shows the wildcard characters that are supported in the destination pattern.

Table 1: 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. Note Using a range of digits is the same as keying in each individual number in the dialed string.
()	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.

The table below 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 2: Dial Peer Matching Examples Using Wildcard Symbols

Destination Pattern	Dial String Translation	String After Stripping 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
408555.%	408555, followed by zero or more wildcard digits. This pattern implies that the string must contain at least 408555.	1234
408555+	40855, followed by 5 repeated one or more times.	1234
408555%	40855, followed by 5 repeated one or more times. Any explicitly matching digit before the % symbol is not stripped off.	51234

Destination Pattern	Dial String Translation	String After Stripping ¹
408555?	40855, followed by 5. Any explicitly matching digit before the ? symbol is not stripped off.	51234
40855[5-7].+	40855, followed by 5, 6, or 7, plus any digit repeated one or more times.	51234
40855[5-7].%	40855, followed by 5, 6, or 7, plus any digit repeated one or more times.	51234
40855[5-7]+1234	40855, followed by 5, 6, or 7 repeated one or more times, followed by 1234.	51234
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 direct inward dialing (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 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. It must be used at the end of the dial string.



Note If you have an exact match that needs preference or demands preference on the dial-peer, you can add a \$ at the end of that dial string. This method allows the exact match preferred over the other random wildcard, creating a consistent result wherein there are two dial-peers with an 'overlap' of destinations and one dial-peer being a range wildcard and the other being an exact match.

- Circumflex symbol (^)--When used within brackets, allows you to eliminate a digit from consideration for dial peer matching purposes. For example, a destination pattern including [^7] would not match any string beginning with 7.

Multiple digits can also be called out within brackets to eliminate more than one initial digit from dial peer matching. For example, a destination pattern including [^4^6^8] would not match any digit string beginning with 4, 6, or 8.



Note A destination pattern including [^752] would allow matching only for digit strings beginning with 5 or 2, but would not match any digit strings beginning with 7. This destination pattern entry essentially behaves the same way as if you had simply included [52] in the destination pattern.

To eliminate a multiple digit string from dial peer matching consideration, you must represent each digit in the string as a succession of individual exceptions. For example, if you wanted to eliminate matching any digit string beginning with 537 from consideration for dial peer matching, you must ensure that your destination pattern includes `[^5][^3][^7]`.

The same destination pattern can be shared across multiple dial peers to form hunt groups.

Digit Stripping and Prefixes

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 1 second between dialing the "9" and dialing the "2" to allow for a secondary dial tone.

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.

Some telephony interfaces require that any digits stripped from the dial string be recovered to support a particular dial plan. You can strip these digits either by using the **no digit-strip** dial-peer voice configuration command to disable the default digit-stripping behavior or by using the **prefix** dial-peer voice configuration command to add digits to the beginning 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 beginning 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
```

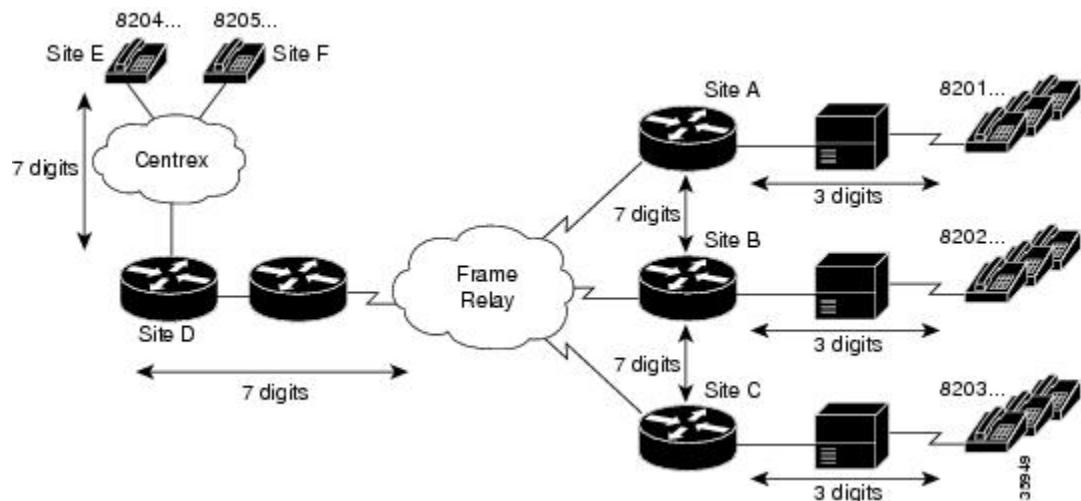
A prefix can also include commas (.). Each comma indicates a 1-second pause in dialing. For example, consider a telephone whose E.164 called number is 1(408)555-1234; to reach this device, you must dial "9." In this case, you might configure "1408....." as the destination pattern, and "9" as the prefix. 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 1 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 1 second per comma, allowing for a dial tone to occur before the router transmits the remaining digits.

The figure below 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), and 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 2: 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
<pre>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 !</pre>	<pre>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 !</pre>

Forwarding 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.

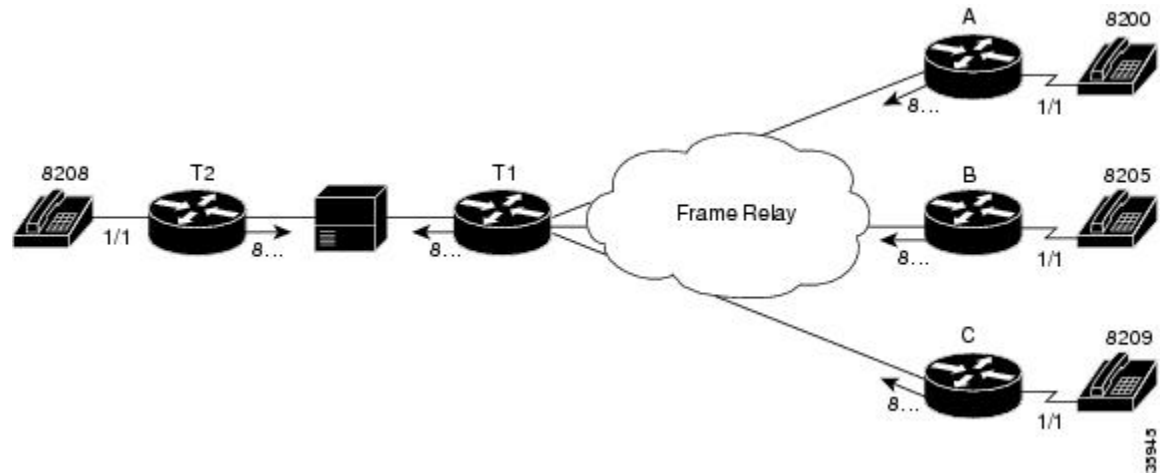
For example, the **forward-digits 4** command tells the router to forward the last four digits in the dialed string. The **forward-digits all** command instructs the router to forward the full dialed string. If the length of the dialed string is longer than the length of the destination pattern, the **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.



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

The figure below 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 3: 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 transmits 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
<pre>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</pre>	<pre>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</pre>

```

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 voice 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. Automatically prepending digits in the dial peer configuration 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 user A 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; user A's extension number is 2123. Suppose that every employee on user A'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, user A 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 2123

```

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.

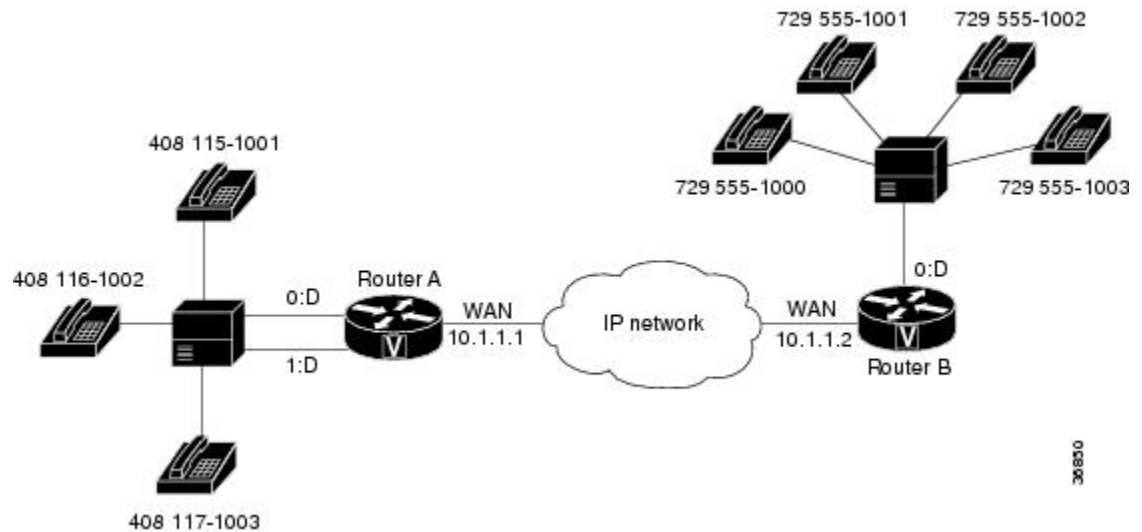
Before you configure the **num-exp** command, it is helpful to map individual telephone extensions to their full E.164 dialed numbers. This task can be done easily by creating a number expansion table.

Creating a Number Expansion Table

The figure below 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 4: VoIP Example for Number Expansion



The table below 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 3: 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

The purpose of this task is to expand an extension number into its full telephone number.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **num-exp** *extension-number* *expanded-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enters privileged EXEC mode.

	Command or Action	Purpose
	Example: Router> enable	<ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	num-exp <i>extension-number</i> <i>expanded-number</i> Example: Router(config)# num-exp 2123 5556611	Configures number expansion globally for all dial peers. The arguments are as follows: <ul style="list-style-type: none"> <i>extension-number</i> --Specifies the extension number to expand into the full telephone number that is specified by the <i>expanded-number</i> argument. <i>expanded-number</i> --Specifies the full telephone number or destination pattern to which the extension number is expanded.

Translation Rules

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 5-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 5-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.

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 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 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.

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

- [CreatingDigitTranslationRules, on page 15](#) (required)

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

- [Applying Translation Rules to Inbound POTS Calls, on page 17](#) (optional)
- [Applying Translation Rules to Inbound VoIP Calls, on page 18](#) (optional)
- [Applying Translation Rules to Outbound Call Legs, on page 19](#) (optional)

CreatingDigitTranslationRules

The purpose of this task is to enter translation-rule configuration mode and specify a set of translation rules.

To create additional individual translation rules to include in the translation-rule set, repeat Step 4 .



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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **translation-rule** *name-tag*
4. **rule** *name-tag input-matched-pattern substituted-pattern [match-type substituted-type]*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	translation-rule <i>name-tag</i> Example: <pre>Router(config)# translation-rule 1</pre>	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. <ul style="list-style-type: none"> • The <i>name-tag</i> argument specifies a unique number that identifies the set of translation rules. Valid entries are from 1 to 2147483647.
Step 4	rule <i>name-tag input-matched-pattern substituted-pattern [match-type substituted-type]</i> Example: <pre>Router(config-translate) # rule 0 ^910 0</pre>	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 arguments are as follows: <ul style="list-style-type: none"> • <i>name-tag</i> --Specifies a unique number that identifies this individual translation rule. Valid entries are from 0 to 10. • <i>input-matched-pattern</i> --Specifies the digit string that must be matched, and then replaced with the <i>substituted-pattern</i> value. • <i>substituted-pattern</i> --Specifies the digit string that replaces the <i>input-matched-pattern</i> value. • <i>match-type</i> --(Optional) Specifies the numbering type that you want to replace with the numbering type defined in the <i>substituted-type</i> value. Enter <i>any</i> for the <i>match-type</i> if you want to match on any numbering type.

	Command or Action	Purpose
		<ul style="list-style-type: none"> • Otherwise, enter one of the following keywords for each of these arguments: <ul style="list-style-type: none"> • abbreviated • international • national • network • reserved • subscriber • unknown

Applying Translation Rules to Inbound POTS Calls

The purpose of this task is to apply a translation rule set to all inbound POTS calls that terminate on the same voice port.



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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *location*
4. **translate** {**called** | **calling**} *name-tag*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	voice-port <i>location</i> Example: <pre>Router(config)# voice-port 1/0:1</pre>	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, refer to the <i>Voice Port Configuration Guide</i> .

	Command or Action	Purpose
Step 4	translate {called calling} <i>name-tag</i> Example: <pre>Router(config-voiceport)# translate called 4</pre>	Specifies the translation rule set to apply to the called number or calling number. The keywords and argument are as follows: <ul style="list-style-type: none"> • called --Applies the translation rule to the called party number. • calling --Applies the translation rule to the calling party number. • <i>name-tag</i> --Specifies the reference number of the translation rule. Valid entries are 1 through 2147483647.

Applying Translation Rules to Inbound VoIP Calls

The purpose of this task is to apply a translation rule set to all inbound VoIP calls that originate at an H.323 gateway.



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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voip-incoming translation-rule** {called | calling} *name-tag*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	voip-incoming translation-rule {called calling} <i>name-tag</i> Example: <pre>Router (config) # voip-incoming translation-rule called 5</pre>	Specifies the translation rule set to apply to all inbound VoIP call legs that originate from an H.323 gateway. The keywords and argument are as follows: <ul style="list-style-type: none"> • called --Applies the translation rule to the called party number.

	Command or Action	Purpose
		<ul style="list-style-type: none"> • calling --Applies the translation rule to the calling party number. • name-tag --Specifies the reference number of the translation rule. Valid entries are 1 through 2147483647.

Applying Translation Rules to Outbound Call Legs

The purpose of this task is to apply a translation rule set to an outbound VoIP or POTS call leg.



Note Translation rules that are configured in a dial peer using the **translate-outgoing** command are not applied to inbound call legs. When two-stage dialing is used, 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.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **pots** | **voip** | **vofr** | **voatm**
4. **translate-outgoing** {**called** | **calling**} *name-tag*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> pots voip vofr voatm Example: Router(config)# dial-peer voice 345 pots	Enters dial-peer voice configuration mode and defines a local dial peer. <ul style="list-style-type: none"> • The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.

	Command or Action	Purpose
Step 4	translate-outgoing {called calling} <i>name-tag</i> Example: <pre>Router(config-dialpeer)# translate-outgoing called 6</pre>	Specifies the translation rule set to apply to the calling number or called number. The keywords and argument are as follows: <ul style="list-style-type: none"> • called --Applies the translation rule to the called party number. • calling --Applies the translation rule to the calling party number. • name-tag --Specifies the reference number of the translation rule. Valid entries are 1 through 2147483647.

Data Dial Peers

In addition to standard voice-network and POTS dial peers, a newer type of dial peer has been introduced to service modem calls over POTS lines with automatic dial peer matching and priority assignment. These new dial peers are called data dial peers.

Traditionally, if a modem call came over a POTS line connected to a voice-network gateway, a procession of matching criteria was required to determine the nature of the incoming call. Only after it was determined that an incoming call was not a voice call could it then be assumed that the transmission was, in fact, a data-based modem call.

Now, however, you have the ability to specify particular dial peers as data dial peers and even assign them priority in relation to other dial peers in the system.

Configuring Data Dial Peers

The purpose of this task is to configure a POTS dial peer to be a data dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer data** *tag* **pots**
4. **incoming called-number** *string*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.

	Command or Action	Purpose
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	dial-peer data tag pots Example: <pre>Router(config)# dial-peer data 2001 pots</pre>	Specifies a dial peer for data calls and enters dial-peer voice configuration mode. The keyword and argument are as follows: <ul style="list-style-type: none"> • tag --Specifies the dial peer identifier. The valid range is from 1 to 2147483647. <p>Note You cannot have a data dial peer and a voice dial peer that are assigned to the same <i>tag</i> number. The <i>tag</i> must be unique for all dial peers.</p> <ul style="list-style-type: none"> • pots --Specifies the dial peer as POTS.
Step 4	incoming called-number string Example: <pre>Router(dial-peer)# incoming called-number 4085551212</pre>	Specifies the incoming called number that is associated with the data dial peer. <ul style="list-style-type: none"> • The <i>string</i> argument specifies the number.

Configuring a Search for Dial Peers by Type

The purpose of this task is to configure a search for dial peers by type.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer search type {data voice | voice data | none}**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.

	Command or Action	Purpose
Step 3	dial-peer search type {data voice voice data none} Example: <pre>Router(config)# dial-peer search type data voice</pre>	Specifies the dial-peer search functionality. The keywords are as follows: <ul style="list-style-type: none"> • data --Searches for data dial peers. • voice --Searches for voice dial peers. • none --Searches for all dial peers with the same preference based on the input order. <p>Note The default is data and voice.</p>

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. The first figure below illustrates call legs from the perspective of the originating router; the second figure below illustrates call legs from the perspective of the terminating router.



Note The figures below 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.

Figure 5: Call Legs from the Perspective of the Originating Router

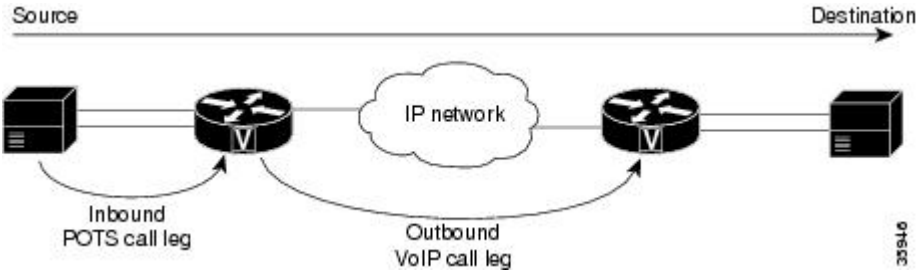
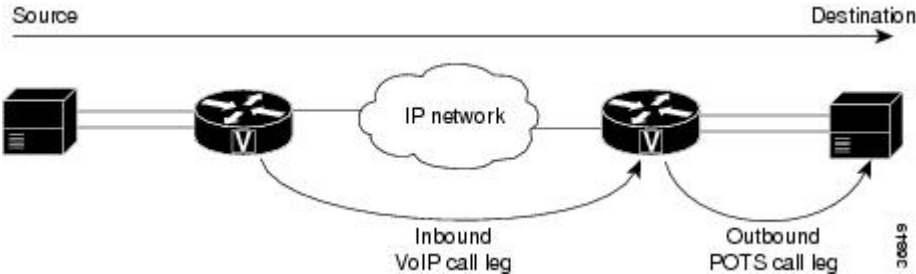


Figure 6: Call Legs from the Perspective of the Terminating Router



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.

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.

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 channel associated signaling (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 five 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 voice configuration command in POTS or multimedia mail over IP (MMoIP) dial peers.
- Answer address--A string representing the calling number or ANI. It is configured by using the **answer-address** dial-peer voice configuration command in POTS or VoIP dial peers and is used only for inbound calls from the IP network.
- Destination pattern--A string representing the calling number or ANI. It is configured by using the **destination-pattern** dial-peer voice configuration command in POTS or voice-network dial peers.
- Application--A string representing the predefined application that you wish to enable on the dial peer. It is configured by using the **application** dial-peer voice configuration command on inbound POTS dial peers.
- 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 the **incoming called-number** command

2. Calling number with the **answer-address** command
3. Calling number with the **destination-pattern** command
4. Incoming voice port with the 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 the **destination-pattern** command is used, the voice port or session target must also be configured.

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 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.

Configuring the incoming called-number Command

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.

The purpose of this task is to identify the service type of a call as voice.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **pots** | **voip** | **vofr** | **voatm**
4. **incoming called-number** *number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> pots voip vofr voatm Example:	Enters dial-peer voice configuration mode and defines a local dial peer.

	Command or Action	Purpose
	Router(config)# dial-peer voice 345 pots	<ul style="list-style-type: none"> The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	incoming called-number <i>number</i> Example: Router(config-dialpeer)# incoming called-number 5551212	Defines the telephone number that identifies voice calls associated with this dial peer.

answer-address Command

The purpose of this task is to specify the answer address for this dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* {pots | voip | vofr| voatm}
4. **answer-address** [+]*string* [T]

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> {pots voip vofr voatm} Example: Router(config)# dial-peer voice 123 pots	Enters dial-peer voice configuration mode and defines a local dial peer. <ul style="list-style-type: none"> The <i>number</i> argument identifies the dial peer. Valid entries are 1–2147483647.
Step 4	answer-address [+] <i>string</i> [T] Example: Router(config-dialpeer)# answer-address 55534..	Defines the phone number that identifies voice calls associated with this dial peer. The keywords and argument are as follows: <ul style="list-style-type: none"> + --(Optional) Character indicating an E.164 standard number. <i>string</i> --A series of digits specifying the E.164 or private dial plan phone number. Valid entries are as follows:

	Command or Action	Purpose
		<ul style="list-style-type: none"> • Digits 0 through 9, letters A through F, pound sign (#), and asterisk (*), which represent specific digits that can be entered. • Comma (,), which inserts a pause between digits. • Period (.), which matches any entered digit. • T --(Optional) Control character indicating that the answer-address value is a variable-length dial string.

Configuring the destination-pattern Command

See the [Configuring Destination Patterns, on page 4](#) for information on configuring the destination pattern for a dial peer.

Configuring the port Command

See the [Assigning Voice Ports, on page 1](#) for information on associating a voice port with a dial peer.

Matching Outbound Dial Peers

The method a router uses to select an outbound dial peer depends on whether ISDN 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.

Using Default Routes

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 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.

Additional Features

One Stage and Two Stage Dialing

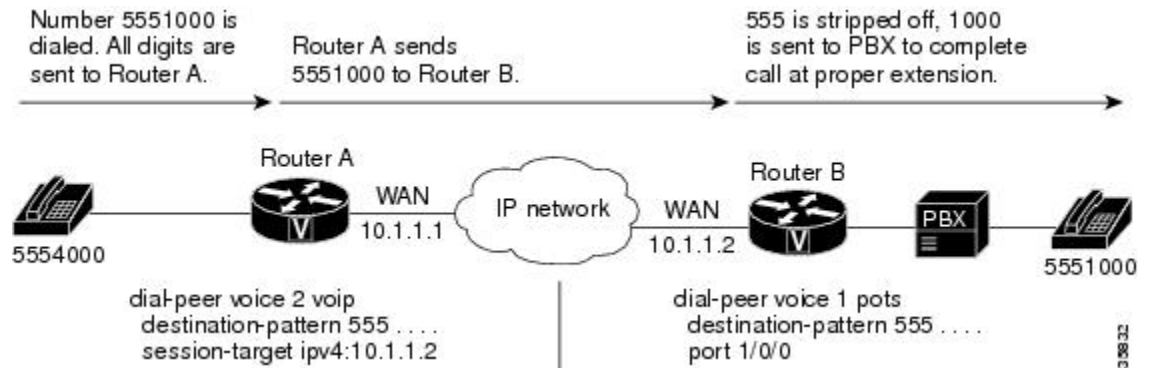
With two-stage dialing, when a voice call enters the network, the originating router collects dialed digits until it can match an outbound dial peer. As soon as the router matches a dial peer, it immediately places the call and forwards the associated dial string. No additional dialed digits are collected. 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.

The figure below illustrates how the originating router collects a dial string and the terminating router forwards the digits to the telephony device.

Figure 7: Collecting and Forwarding Dialed Digits



The examples in the table below demonstrate how the originating router collects dialed digits for a given destination pattern in the outbound voice-network dial peer.

Table 4: Digit Collection Based on Destination Pattern

Dialed Digits	Destination Pattern	Dial String Collected ²
5551234	5.....	5551234
5551234	555....	5551234
5551234	555	555
555123499	555....	5551234

² 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.

Direct Inward Dialing

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.

The 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.



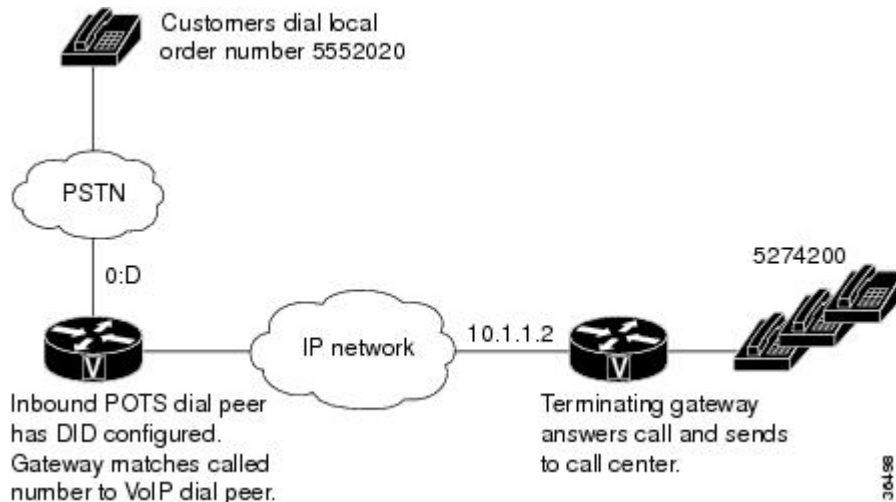
Note DID for POTS dial peers, as described here, is for ISDN connections only. It is not the same as analog DID for Cisco routers, which supports analog DID trunk service. For more information about analog DID see Analog Direct Inward Dialing .

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*.

The figure below shows a call scenario using DID.

Figure 8: VoIP Call Using DID



In the figure above, the POTS dial peer that matches the incoming called-number has DID 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 5274200 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.

Configuring Direct Inward Dialing

The purpose of this task is to configure a POTS dial peer for DID.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **pots**
4. **direct-inward-dial**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice <i>number</i> pots Example: Router(config)# dial-peer voice 234 pots	Enters dial-peer voice configuration mode and defines a local dial peer that will connect to the POTS network. <ul style="list-style-type: none"> • The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	direct-inward-dial Example: Router(config-dialpeer)# direct-inward-dial	Specifies DID for this POTS dial peer.

What to do next



Note DID is configured for inbound POTS dial peers only.

Hunt Groups

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)
<pre>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</pre>	<pre>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</pre>

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.

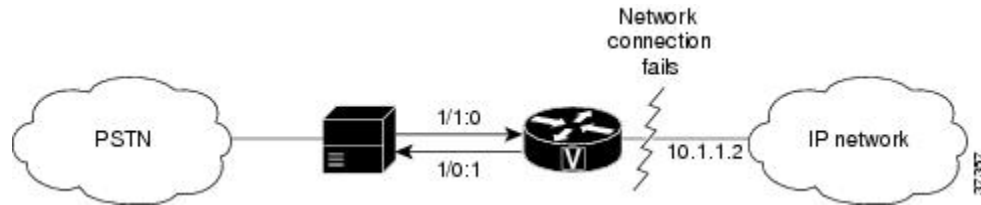
By default, dial peers in a hunt group are selected according to the following criteria, in the order listed:

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 criterion 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. Mixing dial peer types 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 the figure below.

Figure 9: 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. The purpose of this task is to disable dial peer hunting on an individual dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* {pots | voip | vofr | voatm}
4. **huntstop**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	dial-peer voice <i>number</i> {pots voip vofr voatm} Example: Router(config)# dial-peer voice 345 pots	Enters dial-peer voice configuration mode and defines a local dial peer. <ul style="list-style-type: none"> The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	huntstop Example: Router(config-dialpeer)# 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.

Configuring Dial Peer Hunting Options

Use the **no huntstop** command to enable dial peer hunting if it has been disabled.

The purpose of this task is to configure dial peer hunting options for all dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer hunt** *hunt-order-number*
4. **voice hunt** {user-busy | invalid-number | unassigned-number}

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	dial-peer hunt <i>hunt-order-number</i> Example: Router(config)# dial-peer hunt 2	(Optional) Specifies the hunt selection order for dial peers in a hunt group. Valid entries are 0 through 7. The default is 0. The allowable values are as follows: <ul style="list-style-type: none"> 0--Specifies longest match in phone number, explicit preference, random selection. 1--Specifies longest match in phone number, explicit preference, least recent use. 2--Specifies explicit preference, longest match in phone number, random selection.

	Command or Action	Purpose
		<ul style="list-style-type: none"> • 3--Specifies explicit preference, longest match in phone number, least recent use. • 4--Specifies least recent use, longest match in phone number, explicit preference. • 5--Specifies least recent use, explicit preference, longest match in phone number. • 6--Specifies random selection. • 7--Specifies least recent use.
Step 4	<p>voice hunt {user-busy invalid-number unassigned-number}</p> <p>Example:</p> <pre>Router(config)# voice hunt user-busy</pre>	<p>(Optional) Defines how the originating or tandem router handles rotary dial peer hunting if it receives a disconnect cause code from the terminating router. The keywords are as follows:</p> <ul style="list-style-type: none"> • user-busy --Instructs the router to continue dial peer hunting if it receives a user-busy disconnect cause code from a destination router. • invalid-number --Instructs the router to stop dial peer hunting if it receives a an invalid-number disconnect cause code from a destination router. • unassigned-number --Instructs the router to stop dial peer hunting if it receives an unassigned-number disconnect cause code from a destination router.

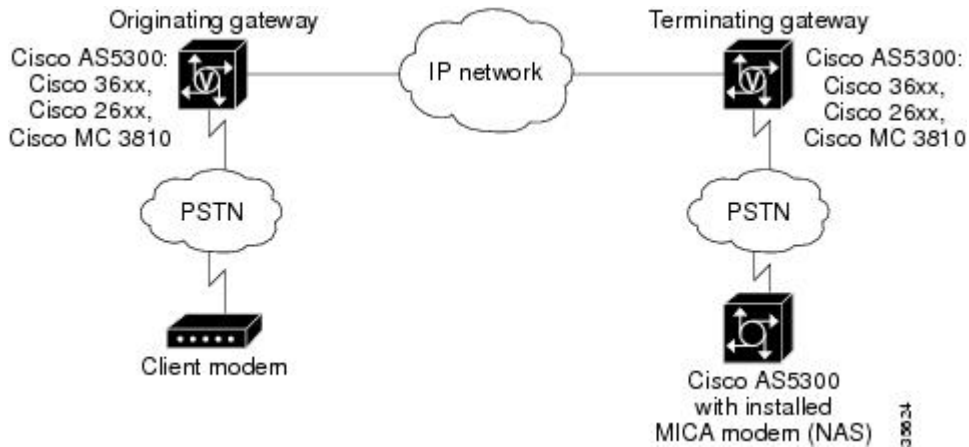
Modem Pass Through

Like T.38 Fax Relay and Modem Relay, Modem Pass Through functionality can be enabled and configured on a per-dial peer basis. Modem Pass Through behavior enables you to take advantage of features such as the following:

- Repressing bandwidth- and resource-consuming functions like compression, echo cancellation, high-pass filtering, and voice activity detection (VAD).
- Automatically sending redundant packets to minimize the possibility of packet loss.
- Employing automatic static jitter buffers to protect against clock skew.
- Identifying signals that are for modem calls versus voice or fax calls.

The figure below illustrates a network featuring Modem Pass Through capability.

Figure 10: Modem Pass Through Connection Example



When a call over the network is identified as a modem call, both the originating and terminating voice gateway routers automatically "roll over" to using the G.711 codec for the duration of the modem call. Once the modem call has ceased, the digital signal processors (DSPs) in both the originating and terminating voice gateways revert to default operation, enabling fax and voice calls to be placed and received using those DSPs. The version of the G.711 codec you use (either a-law or u-law) is determined by the type of network on which your voice gateways are operating and the configuration you specify using the **modem passthrough** command in dial-peer voice configuration mode.

Configuring Modem Pass Through Capability for Individual Dial Peers

The purpose of this task is to configure Modem Pass Through capability for individual dial peers.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* {pots | voip | vofr | voatm}
4. **modem passthrough** {system | nse [payload-type *number*] codec {g711ulaw | g711alaw}[redundancy]}

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<p>dial-peer voice <i>number</i> {pots voip vofr voatm}</p> <p>Example:</p> <pre>Router(config)# dial-peer voice 123 pots</pre>	<p>Enters dial-peer voice configuration mode and defines a local dial peer.</p> <ul style="list-style-type: none"> The <i>number</i> argument identifies the dial peer. Valid entries are from 1 to 2147483647.
Step 4	<p>modem passthrough {system nse [payload-type <i>number</i>] codec {g711ulaw g711alaw}[redundancy]}</p> <p>Example:</p> <pre>Router (config-dialpeer)# modem passthrough codec g711ulaw</pre>	<p>Configures the Modem Pass Through feature for a specific dial peer. The keywords and argument are as follows:</p> <ul style="list-style-type: none"> system --Defaults to the global configuration. <p>Note When the system keyword is used, the nse, payload-type, codec, and redundancy keywords are not valid. Instead, the values from the global configuration are used.</p> <ul style="list-style-type: none"> nse --Named signaling event. payload-type --(Optional) NSE payload type. The <i>number</i> argument specifies the value of the payload type. Valid range is from 96 to 119, inclusive. The default value is 100. <p>Note When the payload type is 100, and you use the show running-config command, the payload-type parameter does not appear in the output.</p> <ul style="list-style-type: none"> codec --Voice compression for speech or audio signals. Codec selections for upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there is only voice traffic. g711ulaw --Codec G.711 u-law 64000 bps for T1. g711alaw --Codec G.711 a-law 64000 bps for E1. <p>Note Be sure to use the same codec type for both the originating gateway and the terminating gateway. The g711ulaw codec is required for T1, and the g711alaw codec is required for E1.</p> <ul style="list-style-type: none"> redundancy --(Optional) Enables packet redundancy (RFC 2198) for modem traffic. The default behavior for Modem Pass Through behavior in dial-peer voice configuration mode is modem passthrough system.

Dual Tone Multifrequency Relay

Dual tone multifrequency (DTMF) tones are generated when a button on a touch-tone phone is pressed. When the tone is generated, it is compressed, transported to the other party, and then decompressed. If a low-bandwidth codec, such as G.729 or G.723, is used without a DTMF relay method, the tone may be distorted during compression and decompression.

DTMF relay sends DTMF tones out of band, or separately from the voice stream. Cisco gateways currently support the following methods of DTMF relay:

- Using a Cisco-proprietary Real-Time Transport Protocol (RTP). DTMF tones are sent in the same RTP channel as voice data. However, the DTMF tones are encoded differently from the voice samples and are identified by a different RTP payload type code. Use of this method accurately transports DTMF tones, but because it is proprietary, it requires the use of Cisco gateways at both the originating and terminating endpoints of the H.323 call.
- Using either the H.245 signal or H.245 alphanumeric method. These methods separate DTMF digits from the voice stream and send them through the H.245 signaling channel instead of through the RTP channel. The tones are transported in H.245 User Input Indication messages. The H.245 signaling channel is a reliable channel, so the packets that transport the DTMF tones are guaranteed to be delivered. However, because of the overhead of using a reliable protocol, and depending on network congestion conditions, the DTMF tones may be slightly delayed. All H.323 version 2-compliant systems are required to support the "h245-alphanumeric" method: support of the "h245-signal" method is optional.
- Using Named Telephone Events (NTEs). Using NTE to relay DTMF tones provides a standardized means of transporting DTMF tones in RTP packets according to section 3 of *RFC 2833, RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, developed by the IETF Audio/Video Transport (AVT) working group. RFC 2833 defines formats of NTE RTP packets used to transport DTMF digits, hookflash, and other telephony events between two peer endpoints. With the NTE method, the endpoints perform per-call negotiation of the DTMF relay method. They also negotiate to determine the payload type value for the NTE RTP packets.

Cisco H.323 gateways advertise capabilities using the H.245 capabilities messages. By default, they advertise that they can receive all DTMF relay modes. If the capabilities of the remote gateway do not match, the Cisco H.323 gateway sends DTMF tones as in-band voice. Configuring DTMF relay on the Cisco H.323 gateway sets preferences for how the gateway handles DTMF transmission. If multiple methods are configured, the priority is as follows:

- Cisco RTP
- RTP NTE
- H.245 signal
- H.245 alphanumeric

In addition to support for NTE, DTMF Relay provides support for asymmetrical payload types. Payload types can differ between local and remote endpoints. Therefore, the Cisco gateway can transmit one payload type value and receive a different payload type value.

Configuring DTMF Relay and Payload Type

The purpose of this task is to configure DTMF Relay capability and the respective payload type for individual dial peers.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `dial-peer voice number {voip | vofr}`
4. `dtmf-relay [cisco-rtp][h245-alphanumeric] [h245-signal]`
5. `rtp payload-type nte number`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>enable</code></p> <p>Example:</p> <pre>Router> enable</pre>	<p>Enters privileged EXEC mode.</p> <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<p><code>configure terminal</code></p> <p>Example:</p> <pre>Router# configure terminal</pre>	<p>Enters global configuration mode.</p>
Step 3	<p><code>dial-peer voice number {voip vofr}</code></p> <p>Example:</p> <pre>Router(config)# dial-peer voice 123 voip</pre>	<p>Enters dial-peer voice configuration mode and defines a remote VoIP dial peer. The keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • The <i>number</i> argument is one or more digits that identify the dial peer. Valid entries are from 1 to 2147483647. • The voip keyword indicates a VoIP peer that uses voice encapsulation on the IP network.
Step 4	<p><code>dtmf-relay [cisco-rtp][h245-alphanumeric] [h245-signal]</code></p> <p>Example:</p> <pre>Router(config-dialpeer)# dtmf-relay cisco-rtp</pre>	<p>Allows DTMF relay using Cisco proprietary RTP packets. DTMF tones are encoded in the Cisco proprietary format and transported in the same RTP channel as the voice.</p>
Step 5	<p><code>rtp payload-type nte number</code></p> <p>Example:</p> <pre>Router(config-dialpeer)# rtp payload-type nte 100</pre>	<p>Identifies the payload type of an RTP packet.</p> <ul style="list-style-type: none"> • The nte keyword identifies the payload type as an NTE. Number values are 96 through 127. The default value is 101. • The following numbers have preassigned values: <ul style="list-style-type: none"> • 96 • 97 • 100 • 121 to 123 • 125 to 127

	Command or Action	Purpose
		<ul style="list-style-type: none"> If you use these values for the NTE payload type, the command fails. You must first reassign the value in use to a different unassigned number, for example: <pre>rtp payload-type nse 105 rtp payload-type nte 100</pre>

Connection PLAR

You can configure dial peers on your Cisco voice gateway router to take advantage of one-way and two-way private-line automatic ringdown (PLAR) functionality. By using the **connection plar** command you can enhance your voice network to offer a number of useful features including the following:

- Providing an off-premises extension (OPX) from a PBX, thus simulating direct connections between FXS port users on a voice gateway router and the PBX.
- Providing dial-tone from a remote PBX in order to offer toll-bypass functionality. Instead of relying on the gateway routers in your voice network to provide dial-tone, you can employ PLAR behavior to enable remote sites to behave as though they have a direct connection to a PBX.
- Connection PLAR behavior eliminates the need for user dialing, because both of the endpoints for the VoIP call are statically configured.

In addition to the features described, connection PLAR behavior does not dedicate bandwidth to a call unless one or the other of the privately associated endpoints goes off-hook.



Note

The Connection PLAR and Connection Trunk feature behavior is configured on a per-voice-port basis. Therefore, you cannot employ the same voice port for both Connection PLAR or trunk mode and collect-dialed-digits mode.

Configuring Connection PLAR

The purpose of this task is to enable Connection PLAR on the specified dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port-number*
4. **connection plar** *number*
5. **exit**
6. **dial-peer voice** *number* {**voip** | **pots**}
7. **port** *port-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	voice-port <i>port-number</i> Example: <pre>Router(config)# voice-port 1/0/0</pre>	Enters voice-port configuration mode for the voice port specified.
Step 4	connection plar <i>number</i> Example: <pre>Router(config-voiceport)# connection plar 51234</pre>	Configures the specified voice port to use the Connection PLAR feature to automatically dial the specified digits.
Step 5	exit Example: <pre>Router(config-voiceport)# exit</pre>	Returns to global configuration mode.
Step 6	dial-peer voice <i>number</i> { voip pots } Example: <pre>Router(config)# dial-peer voice 123 voip</pre>	Enters dial-peer voice configuration mode to configure a VoIP or POTS dial peer.
Step 7	port <i>port-number</i> Example: <pre>Router(config-dial-peer)# port 1/0/0</pre>	Configures the dial peer from the previous step to contact the VoIP network using the specified voice port.

Connection PLAR Design Considerations

Take the following items into consideration when planning to configure Connection PLAR behavior on your VoIP network:

- Because Connection PLAR is a switched VoIP call (similar to a switched virtual circuit), calls are set up and torn down as needed--bandwidth is taken up only when a call is initiated.
- Connection PLAR will operate between any types of signaling endpoints--E&M, FXO, and FXS--and between any combination of analog and digital interfaces.

- Connection PLAR does not collect digits from the connected telephony device, so you can configure Connection PLAR without any subsequent changes to your dial plan.
- Connection PLAR can be enabled on one or both of the statically configured endpoints, thus allowing you the ability to use one-way or two-way Connection PLAR.

**Note**

Because automatically forwarded digits can become distorted over the compressed VoIP audio path when low bit rate codecs like G.729 and G.723.1 are used, you can use the **dtmf relay** command to transport dual tone multifrequency (DTMF) tones out-of-band (separate from the compressed VoIP audio path) to avoid this potential problem. For more information regarding DTMF relay configuration, refer to [Dual Tone Multifrequency Relay, on page 38](#).

Connection Trunk

In addition to configuring Connection PLAR, you can configure your dial peers to employ the Connection Trunk feature using the **connection trunk** command. Connection Trunk functionality offers some of the same advantages as Connection PLAR, such as eliminating the need for user dialing because both of the endpoints for the VoIP call are statically configured. In addition, Connection Trunk behavior offers you the ability to pass supplemental call signaling capability like hookflash and point-to-point Hoot-n-Holler between endpoints on the trunk.

However, unlike Connection PLAR, Connection Trunk endpoints are always active (or off-hook) as far as the voice network is concerned, so bandwidth is always being allocated to a trunk that you have configured.

**Note**

The Connection PLAR and Connection Trunk feature behavior is configured on a per-voice-port basis. Therefore, you cannot employ the same voice port for both Connection PLAR or trunk mode and collect-dialed-digits mode.

Configuring Connection Trunk

The purpose of this task is to enable Connection Trunk behavior on the specified dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port-number*
4. **connection trunk** *number*
5. **exit**
6. **dial-peer voice** *number* {**voip** | **pots**}
7. **port** *port-number*

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enters privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	voice-port <i>port-number</i> Example: <pre>Router(config)# voice-port 1/0/0</pre>	Enters voice-port configuration mode for the voice port specified.
Step 4	connection trunk <i>number</i> Example: <pre>Router(config-voiceport)# connection trunk 51234</pre>	Configures the specified voice port to use the Connection Trunk feature to automatically dial the specified digits.
Step 5	exit Example: <pre>Router(config-voiceport)# exit</pre>	Returns to global configuration mode.
Step 6	dial-peer voice <i>number</i> { voip pots } Example: <pre>Router(config)# dial-peer voice 123 voip</pre>	Enters dial-peer configuration mode to configure a VoIP or POTS dial peer.
Step 7	port <i>port-number</i> Example: <pre>Router(config-dial-peer)# port 1/0/0</pre>	Configures the dial peer from the previous step to contact the VoIP network using the specified voice port.

What to do next



Note Because automatically forwarded digits can become distorted over the compressed VoIP audio path when low bit rate codecs like G.729 and G.723.1 are used, you can use the **dtmf relay** command to transport DTMF tones out-of-band (separate from the compressed VoIP audio path) to avoid this potential problem. For more information regarding DTMF relay configuration, refer to [Dual Tone Multifrequency Relay, on page 38](#).

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
```



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

SUMMARY STEPS

1. Router(config)# **dial-peer cor custom**
2. Router(config-dp-cor)# **name** *class-name*
3. Router(config-dp-cor)# **exit**
4. Router(config)# **dial-peer cor list** *list-name*
5. Router(config-dp-corlist)# **member** *class-name*
6. Router(config-dp-corlist)# **exit**
7. Router(config)# **dial-peer voice** *number* {**pots** | **voip**}
8. Router(config-dial-peer)# **corlist incoming** *cor-list-name*
9. Router(config-dial-peer)# **corlist outgoing** *cor-list-name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	Router(config)# dial-peer cor custom	Enters COR configuration mode to specify that named class of restrictions apply to dial peers.
Step 2	Router(config-dp-cor)# name <i>class-name</i>	Specifies a name for a custom class of restrictions. Note Enter the name (dial peer cor custom) command for additional class names, as needed. You can define a maximum of 64 COR names. 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 <i>list-name</i>	Defines a COR list name.
Step 5	Router(config-dp-corlist)# member <i>class-name</i>	Adds a COR class to this list of restrictions. The member is a class named in Step 2. Note Enter the dial-peer cor list command and member (dial peer cor list) command (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.

	Command or Action	Purpose
Step 7	Router(config)# dial-peer voice <i>number</i> { pots voip }	Enters dial-peer configuration mode and defines a dial peer.
Step 8	Router(config-dial-peer)# corlist incoming <i>cor-list-name</i>	Specifies the COR list to be used when this is the incoming dial peer.
Step 9	Router(config-dial-peer)# corlist outgoing <i>cor-list-name</i>	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 an iLBC Codec

This section includes the following tasks:

Configuring an iLBC Codec on a Dial Peer

The internet Low Bit-rate Codec (iLBC) is intended for packet-based communication. Perform the following steps to configure the iLBC codec on a dial peer.

Before you begin

iLBC is supported on the following:

- Cisco AS5350XM and Cisco AS5400XM Universal Gateways with Voice Feature Cards (VFCs)
- IP-to-IP gateways with no transcoding and conferencing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **rtp payload-type** **cisco-codec-ilbc** [*number*]
5. **codec ilbc** [*mode frame_size*[*bytes payload_size*]]
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: <pre>Router> enable</pre>	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: <pre>Router# configure terminal</pre>	Enters global configuration mode.
Step 3	dial-peer voice <i>tag</i> voip Example: <pre>Router(config)# dial-peer voice 10 voip</pre>	Enters dial-peer configuration mode for the VoIP dial peer designated by <i>tag</i> .
Step 4	rtp payload-type cisco-codec-ilbc [<i>number</i>] Example: <pre>Router(config-dial-peer)# rtp payload-type cisco-codec-ilbc 100</pre>	Identifies the payload type of a Real-Time Transport Protocol (RTP) packet. Keyword and argument are as follows: <ul style="list-style-type: none"> • cisco-codec-ilbc [<i>number</i>] --Payload type is for internet Low Bit Rate Codec (iLBC). Range: 96 to 127. Default: 116.

	Command or Action	Purpose
		<p>Note Do not use the following numbers because they have preassigned values: 96, 97, 100, 117, 121 to 123, and 125 to 127. If you use these values, the command will fail. You must first reassign the value in use to a different unassigned number, for example:</p> <pre>rtp payload-type nse 105 rtp payload-type cisco-codec-ilbc 100</pre>
Step 5	<p>codec ilbc [<i>mode frame_size</i>[<i>bytes payload_size</i>]]</p> <p>Example:</p> <pre>Router(config-dial-peer)# codec ilbc mode 30 bytes 200</pre>	<p>Specifies the voice coder rate of speech for a dial peer. Keywords and arguments are as follows:</p> <ul style="list-style-type: none"> • mode <i>frame_size</i> --The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20. • bytes <i>payload_size</i> --Number of bytes in an RTP packet. For mode 20, valid values are 38 (default), 76, 114, 152, 190, and 228. For mode 30, valid values are 50(default), 100, 150, and 200.
Step 6	<p>exit</p> <p>Example:</p> <pre>Router(config-dial-peer)# exit</pre>	Exits the current mode.

Troubleshooting Tips

You can use the following commands to check iLBC status:

- **show voice call summary**
- **show voice call status**
- **show voice dsmp stream**
- **show call active voice**
- **show call history voice**
- **show voice dsp** and its extensions
- **show dial-peer voice**
- **show voice dsp channel operational-status**

What to Do Next

To set the codec preference, see the "Configuring Multiple Codecs" section in the *Cisco IOS H.323 Configuration Guide*.

Configuring an iLBC Codec in the Voice Class

When using multiple codecs, you must create a voice class in which you define a selection order for codecs; then, you can apply the voice class to VoIP dial peers. The **voice class codec** global configuration command allows you to define the voice class that contains the codec selection order. Then, use the **voice-class codec** dial-peer configuration command to apply the class to individual dial peers.

To configure an iLBC codec in the voice class for multiple-codec selection order, perform the following steps.

Follow these procedures to create a voice class. For the complete dial-peer configuration procedure, see the [Common Practices, on page 1](#).

You can configure more than one voice class codec list for your network. Configure the codec lists and apply them to one or more dial peers based on which codecs (and the order) you want supported for the dial peers. Define a selection order if you want more than one codec supported for a given dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class codec** *tag*
4. **codec preference** *value ilbc [mode frame_size] [bytes payload_size]*
5. **exit**
6. **dial-peer voice** *tag voip*
7. **voice-class codec** *tag*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enters privileged EXEC mode. Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice class codec <i>tag</i> Example: Router(config)# voice class codec 99	Enters voice-class configuration mode and assigns an identification tag number for a codec voice class. The argument is as follows: <ul style="list-style-type: none"> • <i>tag</i> --Unique identifier on the router. Range: 1 to 10000.
Step 4	codec preference <i>value ilbc [mode frame_size] [bytes payload_size]</i> Example:	Specifies a list of preferred codecs to use on a dial peer. Keywords and arguments are as follows: <ul style="list-style-type: none"> • <i>value</i> --Order of preference, with 1 being the most preferred and 14 being the least preferred.

	Command or Action	Purpose
	<pre>Router(config-voice-class)# codec preference 1 ilbc 30 200</pre>	<ul style="list-style-type: none"> • mode <i>frame_size</i> --The iLBC operating frame mode that will be encapsulated in each packet. Valid entries are 20 (20ms frames for 15.2kbps bit rate) or 30 (30ms frames for 13.33 kbps bit rate). Default is 20. • bytes <i>payload_size</i> --Number of bytes in an RTP packet. For mode 20, valid values are 38 (default), 76, 114, 152, 190, and 228. For mode 30, valid values are 50(default), 100, 150, and 200.
Step 5	<p>exit</p> <p>Example:</p> <pre>Router(config-voice-class)# exit</pre>	Exits the current mode.
Step 6	<p>dial-peer voice <i>tag</i> voip</p> <p>Example:</p> <pre>Router(config)# dial-peer voice 16 voip</pre>	Enters dial-peer configuration mode for the specified VoIP dial peer.
Step 7	<p>voice-class codec <i>tag</i></p> <p>Example:</p> <pre>Router(config-dial-peer)# voice-class codec 99</pre>	<p>Assigns a previously configured codec selection preference list (the codec voice class that you defined in step 3) to the specified VoIP dial peer.</p> <p>Note The voice-class codec command in dial-peer configuration mode contains a hyphen. The voice class command in global configuration mode does not contain a hyphen.</p>
Step 8	<p>exit</p> <p>Example:</p> <pre>Router(config-dial-peer)# exit</pre>	Exits the current mode.

Where to Go for Dial Peer Troubleshooting Information

Dial peer troubleshooting consists of a vast array of techniques applicable to a variety of dial peer configuration practices. This document does not cover the techniques necessary to effectively troubleshoot potential pitfalls relating to dial peer configuration. Instead, refer to the TAC Web Site at <http://www.cisco.com/public/support/tac/technologies.shtml>.