Configuring Dialing Plans

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This chapter describes features that enable Cisco Unified Communications Manager Express (Cisco Unified CME) to expand or manipulate internal extension numbers so that they conform to numbering plans used by external systems.

Finding Feature Information in This Module

Your Cisco Unified CME version may not support all of the features documented in this module. For a list of the versions in which each feature is supported, see the "Feature Information for Dialing Plan Features" section on page 408.

Contents

Information About Dialing Plans

To design and configure dialing plans, you should understand the following concepts:

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Phone Number Plan

If you install a Cisco Unified CME system to replace an older telephony system that had an established telephone number plan, you can retain the old number plan. Cisco Unified CME supports flexible extension number lengths and can provide automatic conversion between extension dialing and E.164 public telephone number dialing.

When a router receives a voice call, it selects an outbound 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 for the POTS dial peer. The router then strips out the left-justified numbers corresponding to the destination pattern matching the called number. If you have configured a prefix, the prefix will be put in front of the remaining numbers, creating a dial string, which the router will then dial. If all numbers in the destination pattern are stripped-out, the user will receive (depending on the attached equipment) a dial tone.

A successful Cisco Unified CME system requires a telephone numbering plan that supports future expansion. The numbering plan also must not overlap or conflict with other numbers that are on the same VoIP network or are part of a centralized voice mail system.

Cisco Unified CME supports shared lines and multiple lines configured with the same extension number. This means that you can set up several phones to share an extension number to provide coverage for that number. You can also assign several line buttons on a single phone to the same extension number to create a small hunt group. For more information about types of line configurations, see "Configuring Phones to Make Basic Calls" on page 189.

If you are configuring more than one Cisco Unified CME site, you need to decide how calls between the sites will be handled. Calls between Cisco Unified CME phones can be routed either through the PSTN or over VoIP. If you are routing calls over VoIP, you must decide among the following three choices:

- You can route calls using a global pool of fixed-length extension numbers. For example, all sites have unique extension numbers in the range 5000 to 5999, and routing is managed by a gatekeeper. If you select this method, assign a subrange of extension numbers to each site so that duplicate number assignment does not result. You will have to keep careful records of which Cisco Unified CME system is assigned which number range.
- You can route calls using a local extension number plus a special prefix for each Cisco Unified CME site. This choice allows you to use the same extension numbers at more than one site.
- You can use an E.164 PSTN phone number to route calls over VoIP between Cisco Unified CME sites. In this case, intersite callers use the PSTN area code and local prefix to route calls between Cisco Unified CME systems.

If you choose to have a gatekeeper route calls among multiple Cisco Unified CME systems, you may face additional restrictions on the extension number formats that you use. For example, you might be able to register only PSTN-formatted numbers with the gatekeeper. The gatekeeper might not allow the registration of duplicate telephone numbers in different Cisco Unified CME systems, but you might be able to overcome this limitation. Cisco Unified CME allows the selective registration of either 2- to 5-digit extension numbers or 7- to 10-digit PSTN numbers, so registering only PSTN numbers might prevent the gatekeeper from sensing duplicate extensions.

Mapping of public telephone numbers to internal extension numbers is not restricted to simple truncation of the digit string. Digit substitutions can be made by defining dial-plan patterns to be matched. For information about dial plans, see the "Dial-Plan Patterns" section on page 383. More sophisticated number manipulations can be managed with voice translation rules and voice translation profiles, which are described in the "Voice Translation Rules and Profiles" section on page 384.

In addition, your selection of a numbering scheme for phones that can be directly dialed from the PSTN is limited by your need to use the range of extensions that are assigned to you by the telephone company that provides your connection to the PSTN. For example, if your telephone company assigns you a range from 408 555-0100 to 408 555-0199, you may assign extension numbers only in the range 100 to 199 if those extensions are going to have Direct Inward Dialing (DID) access. For more information about DID, see the "Direct Inward Dialing Trunk Lines" section on page 384.

Dial-Plan Patterns

A dial-plan pattern enables abbreviated extensions to be expanded into fully qualified E.164 numbers. Use dial-plan patterns when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single router, you do not need to use dial-plan patterns.

.When you define a directory number for an SCCP phone, the Cisco Unified CME system automatically creates a POTS dial peer with the ephone-dn endpoint as a destination. For SIP phones connected directly into Cisco Unified CME, the dial peer is automatically created when the phone registers. By default, Cisco Unified CME creates a single POTS dial peer for each directory number.

For example, when the ephone-dn with the number 1001 was defined, the following POTS dial peer was automatically created for it:

```
dial-peer voice 20001 pots
 destination-pattern 1001
 voice-port 50/0/2
```

A dial-plan pattern builds additional dial peers for the expanded numbers it creates. If a dialplan pattern is configured and it matches against a directory number, two POTS dial peers are created, one for the abbreviated number and one for the complete E.164 direct-dial telephone number.

For example, if you then define a dial-plan pattern that 1001 will match, such as 40855500.., a second dial peer is created so that calls to both the 0001 and 4085550001 numbers are completed. In this example, the additional dial peer that is automatically created looks like the following:

```
dial-peer voice 20002 pots
 destination-pattern 40855510001
 voice-port 50/0/2
```

In networks with multiple routers, you may need to use dial-plan patterns to expand extensions to E.164 numbers because local extension numbering schemes can overlap each other. Networks with multiple routers have authorities such as gatekeepers that route calls through the network. These authorities require E.164 numbers so that all numbers in the network are unique. Define dial-plan patterns to expand extension numbers into unique E.164 numbers for registering with a gatekeeper. For more information on E.164 numbers, see "E.164 Enhancements" section on page 385.

If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with the lowest numbered dial-plan pattern tag first. Once a pattern matches an extension number, the pattern is used to generate an expanded number. If additional patterns subsequently match the extension number, they are not used.

Direct Inward Dialing Trunk Lines

Direct Inward Dialing (DID), is a one-way incoming trunking mechanism, that allows an external caller to directly reach a specific extension without the call being served by an attendant or other intervention.

It is a service offered in which the last few (typically three or four) digits dialed by the caller are forwarded to the called party on a special DID trunk. For example, all the phone numbers from 555-0000 to 555-0999 could be assigned to a company with 20 DID trunks. When a caller dials any number in this range, the call is forwarded on any available trunk. If the caller dialed 555-0234, then the digits 2, 3, and 4 are forwarded. These DID trunks could be terminated on a PBX, so that the extension 234 gets the call without operator assistance. This makes it look as though 555-0234 and the other 999 lines all have direct outside lines, while only requiring 20 trunks to service the 1,000 telephone extensions. Using DID, a company can offer its customers individual phone numbers for each person or workstation within the company without requiring a physical line into the PBX for each possible connection. Compared to regular PBX service, DID saves the cost of a switchboard operator. Calls go through faster, and callers feel they are calling a person rather than a company.

Dial-plan patterns are required to enable calls to DID numbers. When the PSTN connects a DID call for "4085550234" to the Cisco Unified CME system, it also forwards the extension digits "234" to allow the system to route the call.

Voice Translation Rules and Profiles

Translation rules manipulate dialed numbers to conform to internal or external numbering schemes. Voice translation profiles allow you to group translation rules together and apply them to the following types of numbers:

- Called numbers (DNIS)
- Calling numbers (ANI)
- Redirected called numbers
- Redirected target numbers—These are transfer-to numbers and call-forwarding final destination numbers. Supported by SIP phones in Cisco Unified CME 4.1 and later versions.

After you define a set of translation rules and assign them to a translation profile, you can apply the rules to incoming and outgoing call legs to and from the Cisco Unified CME router based on the directory number. Translation rules can perform regular expression matches and replace substrings. A translation rule replaces a substring of the input number if the number matches the match pattern, number plan, and type present in the rule.

For configuration information, see the "Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions" section on page 393.

For examples of voice translation rules and profiles, see the "*Voice Translation Rules*" technical note and the "*Number Translation using Voice Translation Profiles*" technical note.

Secondary Dial Tone

A secondary dial tone is available for Cisco Unified IP phones connected to Cisco Unified CME. The secondary dial tone is generated when a phone user dials a predefined PSTN access prefix and terminates when additional digits are dialed. An example is when a secondary dial tone is heard after a PSTN access prefix, such as the number 9, is dialed to reach an outside line. For configuration information, see the "Activating a Secondary Dial Tone" section on page 401.

E.164 Enhancements

Cisco Unified CME 8.5 allows you to present a phone number in + E.164 telephone numbering format. E.164 is an International Telecommunication Union (ITU-T) recommendation that defines the international public telecommunication numbering plan used in the PSTN and other data networks. E.164 defines the format of telephone numbers. A leading + E.164 telephone number can have a maximum of 15 digits and is usually written with a '+' prefix defining the international access code. To dial such numbers from a normal fixed line phone, the appropriate international call prefix must be used.

The leading +E.164 number is unique number specified to a phone or a device. Callers from around the world dial the leading + E.164 phone number to reach a phone or a device without the need to know local or international prefix. The leading + E.164 feature also reduces the overall telephony configuration process by eliminating the need to further translate the telephone numbers.

Phone Registration with Leading + E164 Number

In Cisco Unified CME, phones register using the leading '+' dialing plan in two ways. Phones can either register with the extension number or with leading + E.164 number.

When phones are registered with extension number, the phones will have a dial peer association with the extension number. The **dialplan-pattern** command is enhanced to allow you to configure leading + phone numbers on the dialplan pattern. Once dialplan-pattern is configured, there could be an E.164 number dialpeer associated with the same phone.

For example, phones registered with extension number 1111 can also be reached by dialing +13332221111. This phone registration method is beneficial in two ways, that is, locally, phones are able to reach each other by just dialing the extension numbers and, remotely, phones can dial abbreviated numbers which are translated as an E.164 number at the outgoing dial-peer. See Example 1 (CME1), page 385 for more information.

When phones are registered with a leading + E.164 number, there is only one leading + E.164 number associated with the phone. The **demote** option in the **dialplan-pattern** command allows the phone to have two dialpeers associated with the same phone. For more information on configuring the dialplan-patterns, see, *How to Configure Dialing Plans*.

For example, a phone registered with + E.164 phone number +12223331111 will have two dialpeers associated with the same phone that is, +122233331111 and 1111. See Example 2 (CME2), page 387.

Example 1 (CME1)

In the following example, phones are registered with extension number but they can be reached by either dialing the 5-digit extension number, or a leading + E.164 number. When the dial-peer pattern and extension number is configured, phones can also be reached by dialing its + E.164 number. In this example, phone number 41236 (configured in CME 2 Example) can reach phone number +12223331234 by dialing the abbreviated phone number because the translation profile has the abbreviated rule configured

The phones can reach each other by dialing either the 5-digit extension number or the + E.164 number because the IPv4 address (172.1.1.188) of the phone in CME 2 example is configured in the dial-peer session target for the phone number 41236 in CME 1 example.

```
!
dial-peer voice 333 voip
  destination-pattern +1222333....
  session target ipv4:172.1.1.188
!
voice translation-rule 1
  rule 2 /^3/ /+12223333/
!
voice translation-rule 2
  rule 1 /^01555/ /+1555/
!
voice translation-profile abbreviated-rule-1
  translate called 1
  translate redirect-target 1
!
voice translation-profile callback-rule-2
  translate callback-number 2
!
ephone-dn  1
  number 41236
  translation-profile incoming abbreviated-rule-1
  translation-profile outgoing callback-rule-1
!
!
ephone 1
  button 1:1
!
!
telephony-service
  dialplan-pattern 1 +1333444.... extension-pattern 5
!
voice register dn 1
  number 41237
  translation-profile incoming abbreviated-rule-1
  translation-profile outgoing callback-rule-1
!
!
voice register pool 1
  number 1 dn 1
!
voice register global
  dialplan-pattern 1 +1333444.... extension-pattern 5
```

Example 2 (CME2)

In the following example, phones are registered with leading + E.164 number and the phones can be reached by dialing either the 5-digit extension number or the + E.164 number. In this example, phone number +12223331234 can reach the phone number 41236 (configured in CME 2 Example). The phone number +12223331234 can reach the phone number 41236 by dialing either the 5-digit extension number or the + E.164 number because the IPv4 address (172.1.1.187) of the phone number 41236 is configured in the dial-peer session target in the CME 2 example.

```
!

dial-peer voice 333 voip
  destination-pattern +1333444....
  session target ipv4:172.1.1.187
!
voice translation-rule 1
  rule 1 /^4/ /+13334444/
!
voice translation-rule 2
  rule 1 /^01555/ /+1555/
!
!
voice translation-profile abbreviated-rule-2
  translate called 1  translate redirect-target 1
!
!
voice translation-profile callback-rule-2
  translate callback-number 2
!
ephone-dn  1
  number +12223331234
  translation-profile incoming abbreviated-rule-2
  translation-profile outgoing callback-rule-2
!
!
ephone 1
  button 1:1
!
telephony-service
  dialplan-pattern 1 +1222333.... extension-pattern 4 demote
!
voice register dn 1
  number +12223331235
  translation-profile incoming abbreviated-rule-2
  translation-profile outgoing callback-rule-2
!
!
voice register pool 1
  number 1 dn 1
!
voice register global
  dialplan-pattern 1 +1222333.... extension-pattern 4 demote
```

Because the legacy phone does not have a '+' button, you can configure dialplan-pattern or translation profile and dial 5 digits.

Let us assume that we have a calling number from PSTN 015556667777 calling to any phone, we know for a fact that the phone number can be translated to a leading + E.164 number as, +15556667777. Then, by applying the translate callback-number above, you can use **Local Services** or **Missed Calls** to callback to +15556667777 instead of dialing 015556667777, which is a not universally known number.

Callback and Calling Number Display

In earlier versions of Cisco Unified CME and Cisco Unified SRST, the calling number (number from an incoming call ringing on your phone) was used for both callback (number displayed under Missed Calls in your local phone directory number) and calling numbers. The + E.164 feature in Cisco Unified CME 8.5, allows you to display both calling number and callback numbers in appropriate format so that you are not required to edit the phone numbers before placing a call. The calling number is displayed on the phone when you configure the **translation-profile outgoing** command in ephone-dn or voice register dn mode.

The **translate callback-number** configuration in voice translation-profile allows you to translate the callback number and display it in E.164 format. The **translate callback number** configuration is only applicable for outgoing calls on SIP and SCCP IP phones. When **translate callback number** is configured, the extra callback field is displayed and if the number matches the translation rule, it is translated. For more information see, "Defining Translation Rules for Callback-Number" section on page 402.

Similarly, in Cisco Unified SRST 8.5, you can configure **translate calling** under **voice translation-profile** mode to display the calling number. You can configure **translation-profile outgoing** in **call-manager-fallback** mode or **voice register pool** to display the callback number. You can use **translate called** command in **translation-profile** and **call-manager-fallback** or **voice register pool** will try to match the called number to do the translation. See *Enabling Translation Profiles* for more information.

The leading '+' in the E.164 number is stripped from the called and calling numbers if the called endpoint or gateway, such as H323 or QSIG gateway, does not support the leading '+' sign in the E.164 number translation. You can strip the leading '+' sign from the number you are calling or a called number using the **translation-profile incoming** or **translation-profile outgoing** commands**.**

How to Configure Dialing Plans

This section contains the following tasks:

Dial-Plan Patterns

- SCCP: Configuring Dial-Plan Patterns, page 389 (required)
- SIP: Configuring Dial-Plan Patterns, page 390 (required)
- Verifying Dial-Plan Patterns, page 392 (optional)

Voice Translation Rules

- Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions, page 393 (required)
- SCCP: Applying Voice Translation Rules in Cisco CME 3.2 and Later Versions, page 395 (required)
- SCCP: Applying Translation Rules Before Cisco CME 3.2, page 396 (required)
- SIP: Applying Voice Translation Rules in Cisco Unified CME 4.1 and Later, page 398 (required)
- SIP: Applying Voice Translation Rules before Cisco Unified CME 4.1, page 399 (required)
- Verifying Voice Translation Rules and Profiles, page 400 (optional)

Secondary Dial Tone

- Activating a Secondary Dial Tone, page 401 (optional)

E.164 Enhacements

- Defining Translation Rules for Callback-Number, page 402

SCCP: Configuring Dial-Plan Patterns

To define a dial-plan pattern, perform the following steps.

Tip In networks that have a single router, you do not need to define dial-plan patterns.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **dialplan-pattern** *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern* | **no-reg**]
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**dialplan-pattern** *tag pattern* **extension-length** *length* [**extension-pattern** *epattern*] [**no-reg**] **Example:** `Router(config-telephony)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..` **Note** This example maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.	Maps a digit pattern for an abbreviated extension-number prefix to the full E.164 telephone number pattern.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Exits configuration mode and enters privileged EXEC mode.

SIP: Configuring Dial-Plan Patterns

To create and apply a pattern for expanding individual abbreviated SIP extensions into fully qualified E.164 numbers, follow the steps in this section. Dial-plan pattern expansion affects calling numbers and for call forward using B2BUA, redirecting, including originating and last reroute, numbers for SIP extensions in Cisco Unified CME.

Prerequisites

Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**

4. **dialplan-pattern** *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern*] [**no-reg**]
5. **call-forward system redirecting-expanded**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**dialplan-pattern** *tag pattern* **extension-length** *extension-length* [**extension-pattern** *extension-pattern* | **no-reg**] **Example:** `Router(config-register-global)# dialplan-pattern 1 4085550... extension-length 5`	Defines pattern that is used to expand abbreviated extension numbers of SIP calling numbers in Cisco Unified CME into fully qualified E.164 numbers.
Step 5	**call-forward system redirecting-expanded** **Example:** `Router(config-register-global)# call-forward system redirecting-expanded`	Applies dial-plan pattern expansion globally to redirecting, including originating and last reroute, numbers for SIP extensions in Cisco Unified CME for call forward using B2BUA.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

Verifying Dial-Plan Patterns

To verify dial-plan pattern configurations, perform the following steps.

SUMMARY STEPS

1. **show telephony-service**
2. **show telephony-service dial-peer**
   or
   show dial-peer summary

DETAILED STEPS

Step 1 **show telephony-service**

Use this command to verify dial-plan patterns in the configuration.

The following example maps the extension pattern 4.. to the last three digits of the dial-plan pattern 4085550155:

```
telephony-service
 dialplan-pattern 1 4085550155 extension-length 3 extension-pattern 4..
```

Step 2 SCCP: **show telephony-service dial-peer**
or
SIP: **show dial-peer summary**

Use the command to display dial peers that are automatically created by the **dialplan-pattern** command.

Use this command display the configuration for all VoIP and POTS dial peers configured for a router, including dial peers created by using the **dialplan-expansion (voice register)** command.

The following example is output from the **show dial-peer summary** command displaying information for four dial peers, one each for extensions 60001 and 60002 and because the **dialplan-expansion** command is configured to expand 6.... to 4085555...., one each for 4085550001 and 4085550002. The latter two dial peers will not appear in the running configuration.

```
Router# show dial-peer summary
             AD                                     PRE PASS                OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN       FER THRU SESS-TARGET    STATT
20010  pots  up   up             60002$              0                          0
20011  pots  up   up             60001$              0                          9
20012  pots  up   up             5105555001$         0                          9
20013  pots  up   up             5105555002$         0                          0
```

Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions

To define voice translation rules and voice translation profiles, perform the following steps.

Note To configure translation rules for voice calls in Cisco CME 3.1 and earlier versions, see the *"Cisco IOS Voice, Video, and FAX Configuration Guide."*

Prerequisites

- SCCP support—Cisco CME 3.2 or a later version.
- SIP support—Cisco Unified CME 4.1 or a later version.
- To define up to 100 translation rules per translation rule table—Cisco Unified CME 8.6 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice translation-rule** *number*
4. **rule** *precedence* **/***match-pattern***/ /***replace-pattern***/**
5. **exit**
6. **voice translation-profile** *name*
7. **translate** {**called** | **calling** | **redirect-called** | **redirect-target**} *translation-rule-number*
8. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	`voice translation-rule` *number* **Example:** `Router(config)# voice translation-rule 1`	Defines a translation rule for voice calls and enters voice translation-rule configuration mode. • number—Number that identifies the translation rule. Range: 1 to 2147483647.

	Command or Action	Purpose
Step 4	**rule** *precedence* **/***match-pattern***/** **/***replace-pattern***/** **Example:** `Router(cfg-translation-rule)# rule 1 /^9/ //`	Defines a translation rule. • *precedence*—Priority of the translation rule. Range: 1 to 100. **Note** Range limited to 15 maximum rules in CME 8.5 and earlier versions. • *match-pattern*—Stream Editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern. • *replace-pattern*—SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern. •
Step 5	**exit** **Example:** `Router(cfg-translation-rule)# exit`	Exits voice translation-rule configuration mode.
Step 6	**voice translation-profile** *name* **Example:** `Router(config)# voice translation-profile name1`	Defines a translation profile for voice calls. • *name*—Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.
Step 7	**translate** {**called** | **calling** | **redirect-called** | **redirect-target**} *translation-rule-number* **Example:** `Router(cfg-translation-profile)# translate called 1`	Associates a translation rule with a voice translation profile. • **called**—Associates the translation rule with called numbers. • **calling**—Associates the translation rule with calling numbers. • **redirect-called—**Associates the translation rule with redirected called numbers. • **redirect-target**—Associates the translation rule with transfer-to numbers and call-forwarding final destination numbers. This keyword is supported by SIP phones in Cisco Unified CME 4.1 and later versions. • *translation-rule-number*—Reference number of the translation rule configured in Step 3. Range: 1 to 2147483647.
Step 8	**end** **Example:** `Router(cfg-translation-profile)# end`	Returns to privileged EXEC mode.

What to Do Next

- To apply voice translation profiles to SCCP phones connected to Cisco Unified CME 3.2 or a later version, see the "SCCP: Applying Voice Translation Rules in Cisco CME 3.2 and Later Versions" section on page 395.
- To apply voice translation profiles to SIP phones connected to Cisco Unified CME 4.1 or a later version, see the "SIP: Applying Voice Translation Rules in Cisco Unified CME 4.1 and Later" section on page 398.
- To apply voice translation profiles to SIP phones connected to Cisco CME 3.4 or Cisco Unified CME 4.0(x), see the "SIP: Applying Voice Translation Rules before Cisco Unified CME 4.1" section on page 399.

SCCP: Applying Voice Translation Rules in Cisco CME 3.2 and Later Versions

To apply a voice translation profile to incoming or outgoing calls to or from a directory number on a SCCP phone, perform the following steps.

Prerequisites

- Cisco CME 3.2 or a later version.
- Voice translation profile containing voice translation rules to be applied must be already configured. For configuration information, see the "Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions" section on page 393.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *tag*
4. **translation-profile** {**incoming** | **outgoing**} *name*
5. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables 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	**ephone-dn** *tag* **Example:** `Router(config)# ephone-dn 1`	Enters ephone-dn configuration mode to create an extension (ephone-dn) for a Cisco Unified IP phone line, an intercom line, a paging line, a voice-mail port, or a message-waiting indicator (MWI). • *tag*—Unique sequence number that identifies this ephone-dn during configuration tasks. Range is 1 to the maximum number of ephone-dns allowed on the router platform. See the CLI help for the maximum value for this argument.
Step 4	**translation-profile** {**incoming** | **outgoing**} *name* **Example:** `Router(config-ephone-dn)# translation-profile outgoing name1`	Assigns a translation profile for incoming or outgoing call legs to or from Cisco Unified IP phones. • You can also use an ephone-dn template to apply this command to one or more directory numbers. If you use an ephone-dn template to apply a command and you use the same command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.
Step 5	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

What to Do Next

If you are done modifying parameters for phones in Cisco Unified CME, generate a new configuration file and restart the phones. See "Generating Configuration Files for Phones" on page 359.

SCCP: Applying Translation Rules Before Cisco CME 3.2

To apply a translation rule to an individual directory number in Cisco CME 3.1 and earlier versions, perform the following steps.

Prerequisites

Translation rule to be applied must be already configured by using the **translation-rule** and **rule** commands. For configuration information, see the *"Cisco IOS Voice, Video, and FAX Configuration Guide."*

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag*
4. **translate** {**called** | **calling**} *translation-rule-number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone-dn** *tag* **Example:** `Router(config)# ephone-dn 1`	Enters ephone-dn configuration mode to create directory number for a Cisco Unified IP phone line, an intercom line, a paging line, a voice-mail port, or a message-waiting indicator (MWI).
Step 4	**translate** {**called** | **calling**} *translation-rule-tag* **Example:** `Router(config-ephone-dn)# translate called 1`	Specifies rule to be applied to the directory number being configured. • *translation-rule-tag*—Reference number of previously configured translation rule. Range: 1 to 2147483647. • You can use an ephone-dn template to apply this command to one or more directory numbers. If you use an ephone-dn template to apply a command to a directory number and you also use the same command in ephone-dn configuration mode for the same directory number, the value that you set in ephone-dn configuration mode has priority.
Step 5	**end** **Example:** `Router(cfg-translation-profile)# end`	Exits configuration mode and enters privileged EXEC mode.

What to Do Next

If you are done modifying parameters for phones in Cisco Unified CME, generate a new configuration file and restart the phones. See "Generating Configuration Files for Phones" on page 359.

SIP: Applying Voice Translation Rules in Cisco Unified CME 4.1 and Later

To apply a voice translation profile to incoming calls to a directory number on a SIP phone, perform the following steps.

Prerequisites

- Cisco Unified CME 4.1 or a later version.
- Voice translation profile containing voice translation rules to be applied must be already configured. For configuration information, see the "Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions" section on page 393.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **translation-profile incoming** *name*
5. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	**voice register dn** *dn-tag* **Example:** `Router(config)# voice register dn 1`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or a message-waiting indicator (MWI).
Step 4	**translation-profile incoming** *name* **Example:** `Router(config-register-dn)# translation-profile incoming name1`	Assigns a translation profile for incoming call legs to this directory number.

Step 5	`end`
	
	Example:
	`Router(config-register-dn)# end`
		Returns to privileged EXEC mode.

What to Do Next

If you are done modifying parameters for phones in Cisco Unified CME, generate a new configuration file and restart the phones. See "SIP: Generating Configuration Profiles for SIP Phones" on page 363.

SIP: Applying Voice Translation Rules before Cisco Unified CME 4.1

To apply an already-configured voice translation rule to modify the number dialed by extensions on a SIP phone, perform the following steps.

Prerequisites

- Cisco CME 3.4 or a later version.
- Voice translation rule to be applied must be already configured. For configuration information, see the "Defining Voice Translation Rules in Cisco CME 3.2 and Later Versions" section on page 393.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag*
4. **translate-outgoing** {**called** | **calling**} *rule-tag*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**`enable`** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**`configure terminal`** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**`voice register pool`** *`pool-tag`* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones.

Step 4	**translate-outgoing** {**called** | **calling**} *rule-tag* **Example:** `Router(config-register-pool)# translate-outgoing called 1`	Specifies an already configured voice translation rule to be applied to SIP phone being configured.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits configuration mode and enters privileged EXEC mode.

What to Do Next

If you are done modifying parameters for phones in Cisco Unified CME, generate a new configuration file and restart the phones. See "SIP: Generating Configuration Profiles for SIP Phones" on page 363.

Verifying Voice Translation Rules and Profiles

To verify voice translation profiles, and rules, perform the following steps.

SUMMARY STEPS

1. **show voice translation-profile**
2. **show voice translation-rule**
3. **test voice translation-rule**

DETAILED STEPS

Step 1 **show voice translation-profile** [*name*]

This command displays the configuration of one or all translation profiles.

```
Router# show voice translation-profile profile-8415

Translation Profile: profile-8415
        Rule for Calling number: 4
        Rule for Called number: 1
        Rule for Redirect number: 5
        Rule for Redirect-target number: 2
```

Step 2 **show voice translation-rule** [*number*]

This command displays the configuration of one or all translation rules.

```
Router# show voice translation-rule 6

Translation-rule tag: 6
   Rule 1:
   Match pattern: 65088801..
   Replace pattern: 6508880101
   Match type: none   Replace type: none
   Match plan: none   Replace plan: none
```

Step 3 **test voice translation-rule** *number*

This command enables you to test your translation rules.

```
Router(config)# voice translation-rule 5
Router(cfg-translation-rule)# rule 1 /201/ /102/
Router(cfg-translation-rule)# exit
Router(config)# exit
Router# test voice translation-rule 5 2015550101

Matched with rule 5
Original number:2015550101  Translated number:1025550101
Original number type: none    Translated number type: none
Original number plan: none    Translated number plan: none
```

Activating a Secondary Dial Tone

To activate a secondary dial tone after a phone user dials the specified number string, perform the following steps.

Prerequisite

- Cisco CME 3.0 or a later version.
- PSTN access prefix must be configured for outbound dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **secondary-dialtone** *digit-string*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.

	Command or Action	Purpose
Step 4	**secondary-dialtone** *digit-string* **Example:** `Router(config-telephony)# secondary-dialtone 9`	Activates a secondary dial tone when *digit-string* is dialed. • *digit-string*—String of up to 32 digits that, when dialed, activates a secondary dial tone. Typically, the *digit-string* is a predefined PSTN access prefix.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Defining Translation Rules for Callback-Number

To define a translation rule for callback numbers on a SIP phone, perform the following steps:

Prerequisites

- To define up to 100 translation rules per translation rule table—Cisco Unified CME 8.6 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice translation-rule** *number*
4. **rule** *precedence* **|** *match-pattern* **|** *replace-pattern* **|**
5. **exit**
6. **voice translation profile** *name*
7. **translate {callback-number | called | calling | redirect-called | redirect-target} translation-rule**-*number*
8. **exit**
9. **voice register pool** *phone-tag*
10. **number** *tag* **dn** *dn-tag*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice translation-rule** *number* **Example:** `Router(config)# voice translation-rule 10`	Defines a translation rule for voice calls and enters voice translation-rule configuration mode. • *number*—Number that identifies the translation rule. Range: 1 to 2147483647.
Step 4	**rule** *precedence* | *match-pattern* | *replace-pattern*| **Example:** `Router(cfg-translation-rule)# rule 1 /^9/ //`	Defines a translation rule. • *precedence*—Priority of the translation rule. Range: 1 to 100. **Note** Range limited to 15 maximum rules in CME 8.5 and earlier versions. • *match-pattern*—Stream Editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern. • *replace-pattern*—SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern.
Step 5	**exit** **Example:** `Router(cfg-translation-rule)# exit`	Exits voice translation-rule configuration mode.
Step 6	**voice translation-profile** *name* **Example:** `Router(config)# voice translation-profile eastern`	Defines a translation profile for voice calls. • *name*—Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.

	Command or Action	Purpose
Step 7	**translate {callback-number | called | calling | redirect-called | redirect-target} translation-rule**-*number* **Example:** `Router(cfg-translation-profile)# translate callback-number 10`	Associates a translation rule with a voice translation profile. • callback-number—Associates the translation rule with the callback-number. • called—Associates the translation rule with called numbers. • calling—Associates the translation rule with calling numbers. • redirect-called—Associates the translation rule with redirected called numbers. • redirect-target—Associates the translation rule with transfer-to numbers and call-forwarding final destination numbers. This keyword is supported by SIP phones in Cisco Unified CME 4.1 and later versions. • translation-rule-*number*—Reference number of the translation rule configured in Step 3. Range: 1 to 2147483647
Step 8	**exit** **Example:** `Router(cfg-translation-profile))# exit`	Exits voice translation-profile configuration mode.
Step 9	**voice register pool** *phone-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.
Step 10	**number** *tag* **dn** *dn-tag* **Example:** `Router(config-register-pool)# number 1 dn 17`	Associates a directory number with the SIP phone being configured. • **dn** *dn-tag*—Identifies the directory number for this SIP phone as defined by the voice register dn command.
Step 11	**end** **Example:** `Router(config-translation-profile)# end`	Returns to privileged EXEC mode.

What to Do Next

- To apply voice translation profiles to SIP phones connected to Cisco Unified CME 4.1 or a later version, see the "SIP: Applying Voice Translation Rules in Cisco Unified CME 4.1 and Later" section.

Examples

The following examples shows translation rules defined for callback-number:

```
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 media flow-around
 allow-connections sip to sip
!
!
voice translation-rule 10
!
!
voice translation-profile eastcoast
!
voice translation-profile eastern
 translate callback-number 10
!
```

Configuration Examples for Dialing Plan Features

This section contains the following example:

Secondary Dial Tone: Example

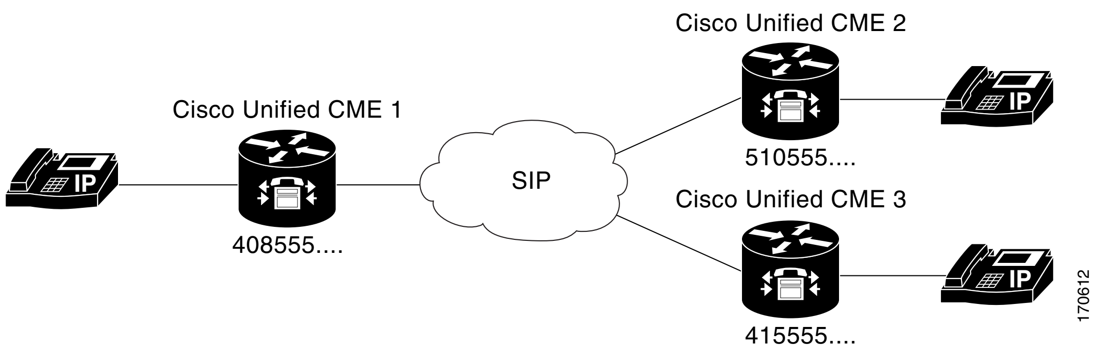
```
telephony-service
 fxo hook-flash
 load 7910 P00403020214
 load 7960-7940 P00305000600
 load 7914 S00103020002
 load 7905 CP7905040000SCCP040701A
 load 7912 CP7912040000SCCP040701A
 max-ephones 100
 max-dn 500
 ip source-address 10.153.233.41 port 2000
 max-redirect 20
 no service directed-pickup
 timeouts ringing 10
 system message XYZ Company
 voicemail 7189
 max-conferences 8 gain -6
 moh music-on-hold.au
 web admin system name admin1 password admin1
 dn-webedit
 time-webedit
 !
 !
 !
 secondary-dialtone 9
```

Voice Translation Rules: Example

In the following configuration examples, if a user on Cisco Unified CME 1 dials 94155550100, the call matches on dial peer 9415 and uses translation profile *profile-9415*. The called number is translated from 94155550100 to 4155550100, as specified by the **translate called** command using translation rule 1.

If a user on Cisco Unified CME 1 calls a phone on Cisco Unified CME 2 by dialing 5105550120, and the call forward number is 94155550100, Cisco Unified CME 1 attempts to forward the call to 94155550100. A 302 message is then sent to Cisco Unified CME 1 with the "Contact:" field translated to 4155550100. When the 302 reaches Cisco Unified CME 1, it matches the To: field in the 302 message (5105550120) with dial peer 510. It does incoming translation from 4155550100 to 84155550100, and an INVITE with 84155550100 is sent, which matches dial-peer 8415.

Figure 15 Translation Rules in SIP Call Transfer



Cisco Unified CME 1 with 408555.... dialplan-pattern	Cisco Unified CME 2 with 510555.... dialplan-pattern

<pre>dial-peer voice 9415 voip
 translation-profile outgoing profile-9415
 destination-pattern 9415555....
 session protocol sipv2
 session target ipv4:10.4.187.177
 codec g711ulaw
 	
voice translation-profile profile-9415
 translate called 1
 translate redirect-target 1
 	
voice translation-rule 1
 rule 1 /^9415/ /415/</pre>
	<pre>dial-peer voice 8415 voip
	 translation-profile outgoing profile-8415
	 destination-pattern 8415555....
	 session protocol sipv2
	 session target ipv4:10.4.187.177
	 codec g711ulaw
	
	dial-peer voice 510 voip
	 translation-profile incoming profile-510
	 destination-pattern 510555....
	 session protocol sipv2
	 session target ipv4:10.4.187.188
	 codec g711ulaw
	
	voice translation-profile profile-8415
	 translate called 1
	 translate redirect-target 2
	
	voice translation-profile profile-510
	 translate called 3
	
	voice translation-rule 1
	 rule 1 /^9415/ /415/
	
	voice translation-rule 2
	 rule 2 /^415/ /9415/
	
	voice translation-rule 3
	 rule 1 /^8415/ /415/</pre>

Additional References

The following sections provide references related to Cisco Unified CME features.

Related Documents

Related Topic	Document Title
Cisco Unified CME configuration	• *Cisco Unified CME Command Reference* • Cisco Unified CME Documentation Roadmap
Cisco IOS commands	• *Cisco IOS Voice Command Reference* • Cisco IOS Software Releases 12.4T Command References
Cisco IOS configuration	• *Cisco IOS Voice Configuration Library* • Cisco IOS Software Releases 12.4T Configuration Guides
Phone documentation for Cisco Unified CME	• User Documentation for Cisco Unified IP Phones

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Feature Information for Dialing Plan Features

Table 31 lists the features in this module and enhancements to the features by version.

To determine the correct Cisco IOS release to support a specific Cisco Unified CME version, see the *Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix* at http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/requirements/guide/33matrix.htm.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 31 lists the Cisco Unified CME version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature.

Table 31 Feature Information for Dialing Plan Features

Feature Name	Cisco Unified CME Versions	Feature Information
Dial-Plan Pattern	4.0	Added support for dial-plan pattern expansion for call forward and call transfer when the forward or transfer-to target is an individual abbreviated SIP extension or an extension that appear on a SIP phone.
	2.1	Strips leading digit pattern from extension number when expanding an extension to an E.164 telephone number. The length of the extension pattern must equal the value configured for the extension-length argument.
	1.0	Adds a prefix to extensions to transform them into E.164 numbers.
E.164 Enhancements	8.5	Added support for E.164 enhancements.
Secondary Dial Tone	3.0	Support for secondary dial tone after dialing specified number string.
Voice Translation Rules	8.6	Added support for an increased number of translation rules per translaiton table. Old value is 15 maximum, new value is 100 maximum.
	4.1	Added support for voice translation profiles for incoming call legs to a directory number on a SIP phone.
	3.4	Added support for voice translation rules to modify the number dialed by extensions on a SIP phone.
	3.2	Adds, removes, or transforms digits for calls going to or originating from specified ephone-dns.