Understand IOS and IOS XE Call Routing

Contents

Introduction

This document describes an explanation of Cisco IOS® and Cisco IOS XE Call Routing.

Prerequisites

Requirements

While there are no formal prerequisites needed to read this document, it is written with the expectation that the reader already has some knowledge of underlying voice signaling protocols that are used to establish and connect phone calls. These protocols are referenced many times throughout.

Signaling Protocols: Session Initiation Protocol (SIP), H323 (h225 / h245), Media Gateway Control Protocol (MGCP), Skinny Client Control Protocol (SCCP), ISDN Q931, E1 R2.

Media Protocols: Real Time Protocol (RTP), voice codecs, video codecs.

Analog Technologies: Ear and Mouth (E&M), Foreign Exchange Subscriber (FXS) and Foreign Exchange Office (FXO).

Components Used

The information in this document is based on these software and hardware versions:

- Cisco IOS and Cisco IOS XE Gateways
- 2800 / 3800 / 2900 / 3900 / 4300 / 4400 / CSR1000v / CAT8000v / ASR100X / C8200 / C8300 / ISR1100

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

Background Information

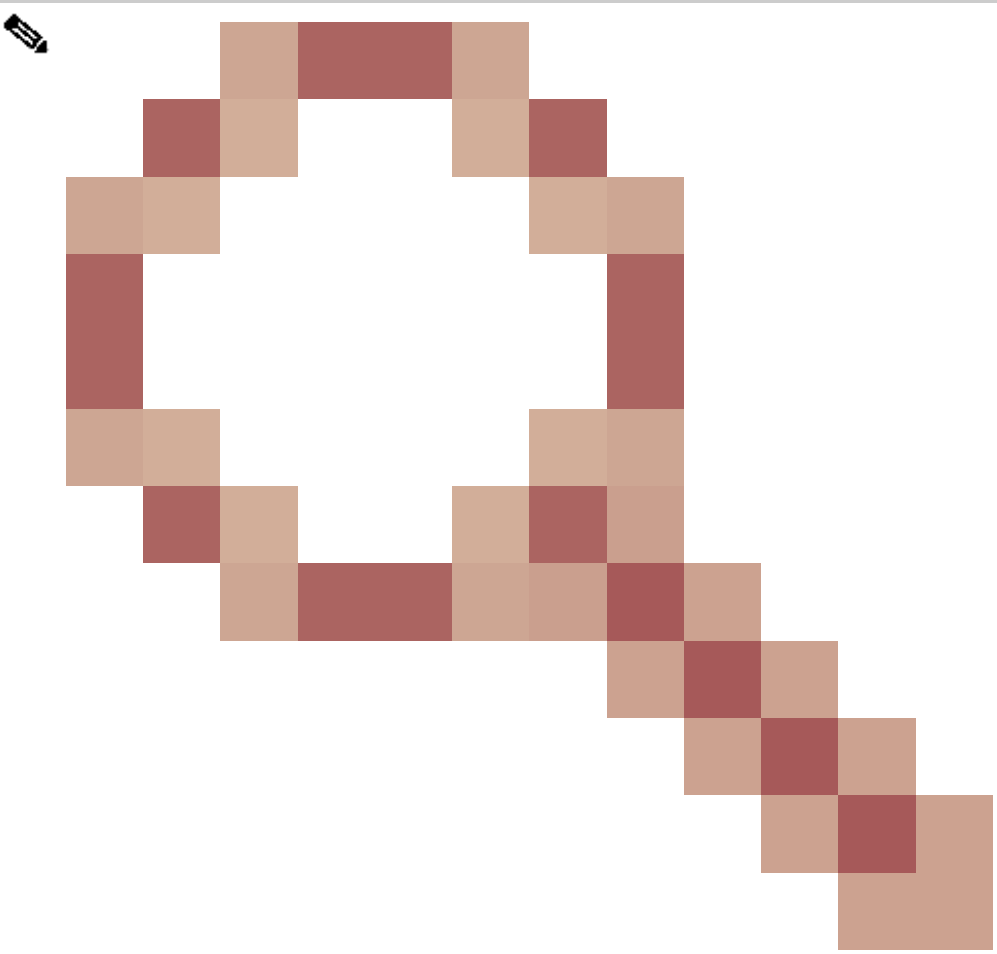
This document addresses the mechanisms behind inbound and outbound dial-peer matching with Plain Old Telephone Service (POTS) and Voice over IP (VoIP) Network call legs.

In addition to dial-peer information, this document covers important topics that pertain to call routing. These include digit manipulation, a quick overview of Session Initiation Protocol (SIP) message manipulation, a few methods for restricting calling capabilities, a quick media and signaling binding overview, and lastly a bit of troubleshooting.

This document utilizes configuration examples as well as debug and show command outputs as reference points. The many features in this document are clearly marked with the version the feature was introduced to both Cisco IOS and Cisco IOS XE. This information can also be referenced quickly in the Command and Feature Roadmap section. If there is a very notable defect, it is linked within the text so that readers are aware.

Common Definitions

Attribute	Description
Digit String	Also referred to as number string, phone number, number, or E164 number.Consists entirely of digits 0 through 9 with an optional leading plus symbol (+). **Example**: 8675309 123456789 +1972525222 +442084445555 +85225353333
Dialed Number Identification Service (DNIS)	This is the Called Number or the Destination Number for a call.
Automatic Number Identification (ANI)	This is the Calling Number or the Originating Calling Number for a call. This can also be referred to as the Calling Line Identifier (CLID) which can also be titled the Caller ID.
Uniform Resource Identifier (URI)	A URI is either a sip: or tel: string most commonly used with VoIP Protocols SIP and H323. **URL Examples:** sip:user@example.com sip:user@sub.example.com sip:user@192.0.2.1 sip:user@2001:4860:4860::8888 tel:8675309 sip:example.com
Carrier-id	**CID Examples:** cid:orange@example.com cid:orange@sub.example.com cid:orange@192.0.2.1 cid:orange@2001:4860:4860::8888 **Note**: Cisco bug ID CSCua14749

Attribute	Description
	 Carrier-ID does not work on Cisco IOS XE platforms.
Route-String	A Cisco Proprietary Header for ILS Route-Strings used with SIP. **Example**: X-cisco-dest-route-string: <sip:configured-value>
ENUM	ENUM is a protocol that uses Domain Name Service (DNS) to translate E164 phone numbers into URIs. This is not covered in this document.
PSTN	Public Switched Telephone Network
ITSP	Internet Telephony Service Provider
SBC	Session Border Controller. This is the device that stands as the demarcation point between your LAN and an ITSP/PSTN Network

Command and Feature Roadmap

Feature	Cisco IOS Version	Cisco IOS XE Version
Number Expansion (num-exp) Dial-peers (POTS and VOIP) answer-address destination-pattern incoming called-number session target (IPv4 and DNS) Max Connections (max-conn) direct-inward-dial forward-digits (POTS) prefix (POTS) timeouts inter-digit (voice-port)	11.3(1)T	All
dial-peer terminator	12.0	All
huntstop	12.0(5)T	All
ISDN Maps	12.0(6)T	All
Dial-peer Hunting Schemes	12.0(7)XK	All
Voice Translation Rule and Profile translate-outgoing numbering-type digit-strip (POTS)	12.0.(7)XR1	All

session target (sip-server)	12.1(1)T	All
POTS Trunk Group	12.1(3)T	All
DNIS-Map (Outbound)	12.2(2)XB	All
trunk-group-label	12.2(11)T	All
Dial-peer (Data)	12.2(13)T	All
Voice Class URI (Outbound)	12.3(4)T	All
outbound-proxy	12.4(15)T	All
session target (IPv6)	12.4(22)T	All
SIP Profiles (Outbound)	15.0(1)M	All
Voice Class URI (Inbound) voice source-group	15.1(2)T	3.8S
SIP Copylist session target (Registrar)	15.1(3)T	3.6S
call-route (url)	15.2(1)T	3.3S
max-bandwidth	15.2(2)T	3.7S
E164-Pattern-Maps (Outbound)	15.2(4)M	3.7S
Voice Class Route-String call-route (dest-route-string)	15.3(3)M	3.10S
Dial-Peer Groups (VOIP)	15.4(1)T	3.11S

E164-Pattern-Maps (Inbound) Destination Server Group requri-passing session target (sip-uri)		
Dial-peer Provision Policy SIP-Profiles (Inbound)	15.4(2)T	3.12S
Dial-Peer Group (POTS)	15.5(1)T	3.14S
Voice Class Tenants	15.6(2)T	16.3.1
VRF Filtering for dial-peers	15.6(3)M	16.3.1
e164-translation	n/a	16.8.1
SIP DSAPP	n/a	16.12.1
Huntstop for server-groups	n/a	17.4.1
sip listen port for tenant and Tenant Filtering for dial-peers	n/a	17.8.1
DNS SRV Based Option Keepalive	n/a	17.9.1

Cisco IOS / Cisco IOS XE Call Routing Fundamentals

Cisco IOS and Cisco IOS XE gateways utilize a concept of a dial-peer to control call routing and capabilities negotiation for each leg of a call. A call leg is the bidirectional communication between two call agents. A call agent is a device that initiates, processes, or forwards telephony calls. This can be and is not limited to Telephony Provider equipment, a Cisco Gateway, an IP Phone, a Cisco Unified Communication Manager (CUCM), Cisco Unity connection (CUC), and so on. There are far too many Call Agents to list.

Scenario: A call arrives at a Cisco gateway from another call agent and is the inbound call leg (in-leg). The gateway processes the call and based on its processing sends the call to the next call agent. This is the outbound call leg (out-leg).

Image 1 shows a call from the PSTN to CUCM routing through a Cisco Voice Gateway and the respective inbound and outbound call leg information.

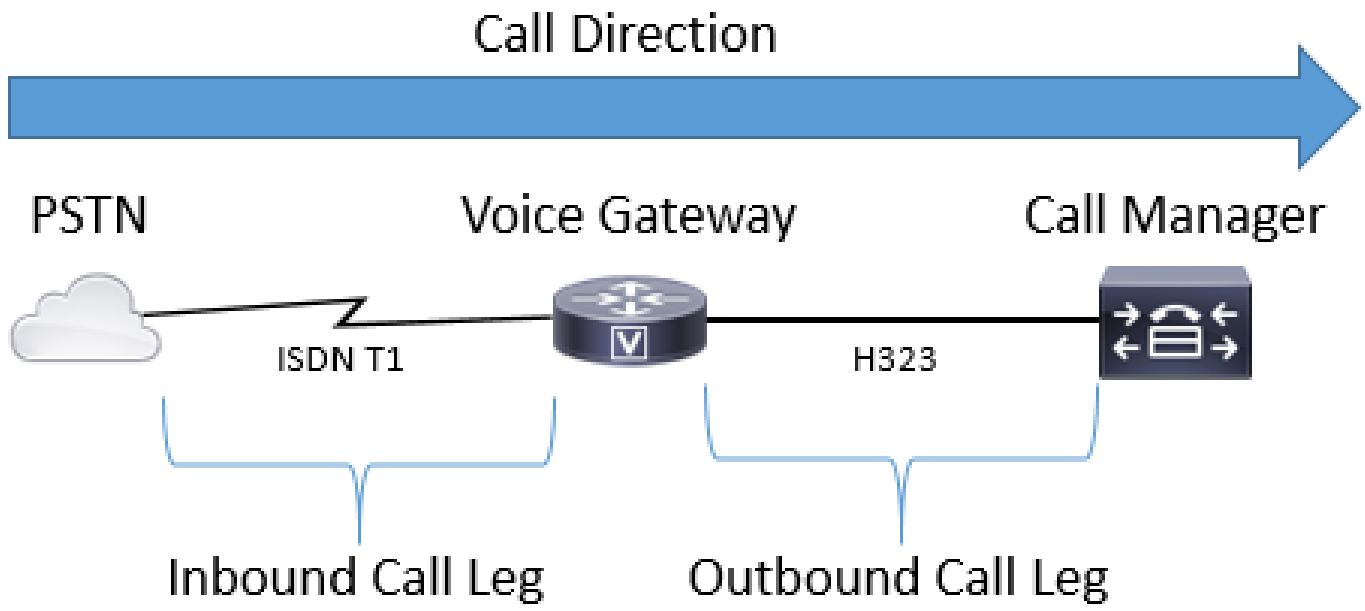


Image 1 - Inbound and Outbound Call Legs Illustrated

A successful call through a Cisco Gateway ALWAYS (see note) matches an inbound or outbound dial-peer to route properly. Inbound and outbound dial-peers are similar to the call-legs mentioned earlier. In **I**mage 1, the call arrives from the PSTN at the Cisco Gateway and needs to match an inbound dial-peer. Then the gateway utilizes an outbound dial-peer to route the call to the next call agent. It is important to remember that these terms are defined from the perspective of the Cisco Gateway.

By matching a dial-peer for each side of the call, an administrator has the power to control many aspects of each specific call leg. Examples of these include voice codecs, DTMF preferences, digit manipulation, where the call is to be routed, and many other settings. Dial-peers can be configured with both inbound and outbound match statements so matching the same dial-peer for both the in-leg and out-leg is possible if a valid inbound and outbound matching configuration is applied to that specific dial-peer.

Note: The exception to this rule is with MGCP and SCCP voice-ports. These signaling protocols do not follow normal dial-peer matching mechanism during call routing. See the SCCP and MGCP section for further details.

Image 2 illustrates the same inbound and outbound call legs as Image 1 but with the respective dial-peers for a call from the PSTN to CUCM routing through an Cisco Voice Gateway.

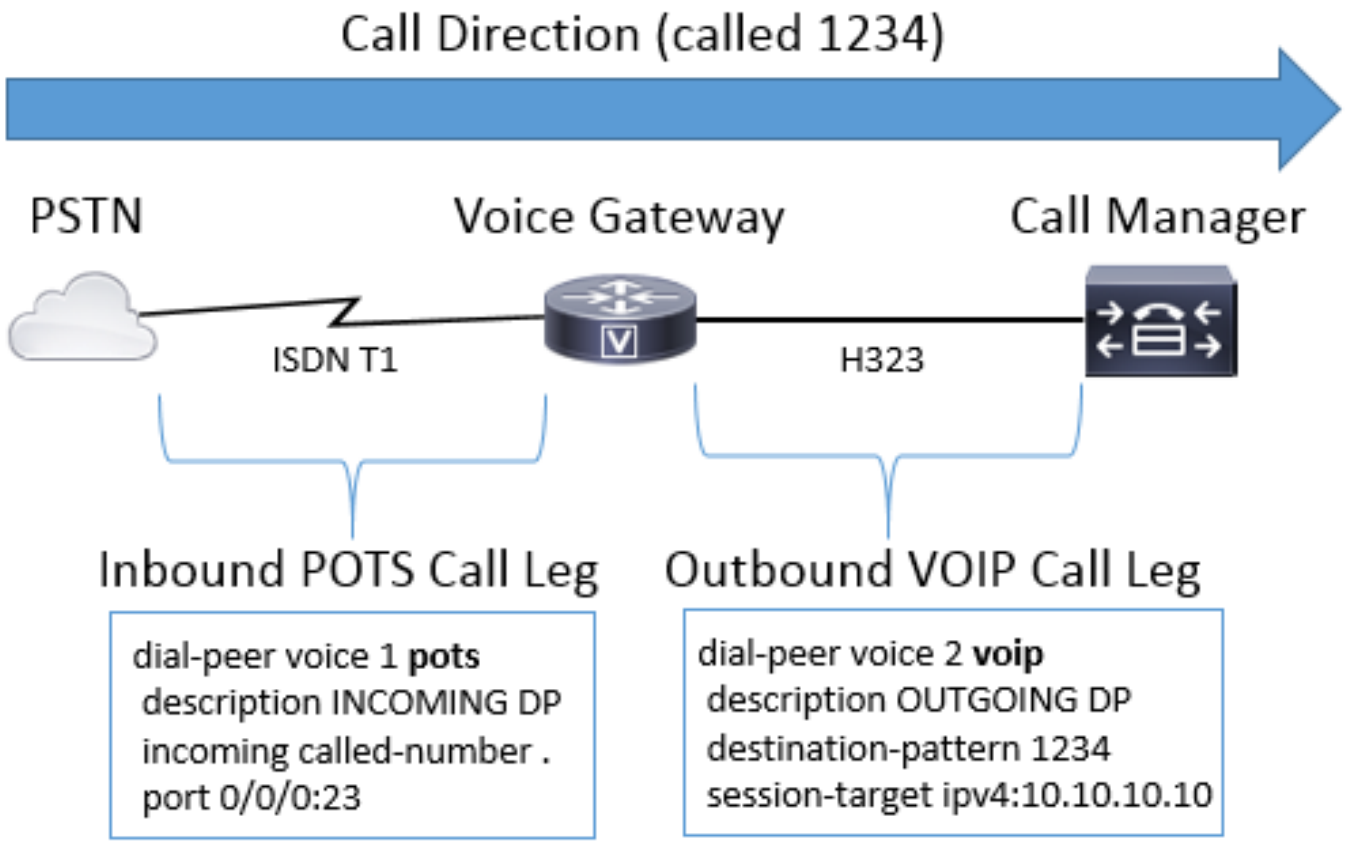


Image 2 - Inbound and Outbound dial-peers Illustrated

Cisco Voice Gateways can interwork many different types of voice calls and protocols including IP to IP, POTS to POTS and IP to POTS or vice versa.

Image 3 illustrates a VoIP to VoIP call through Cisco Unified Border Element (CUBE).

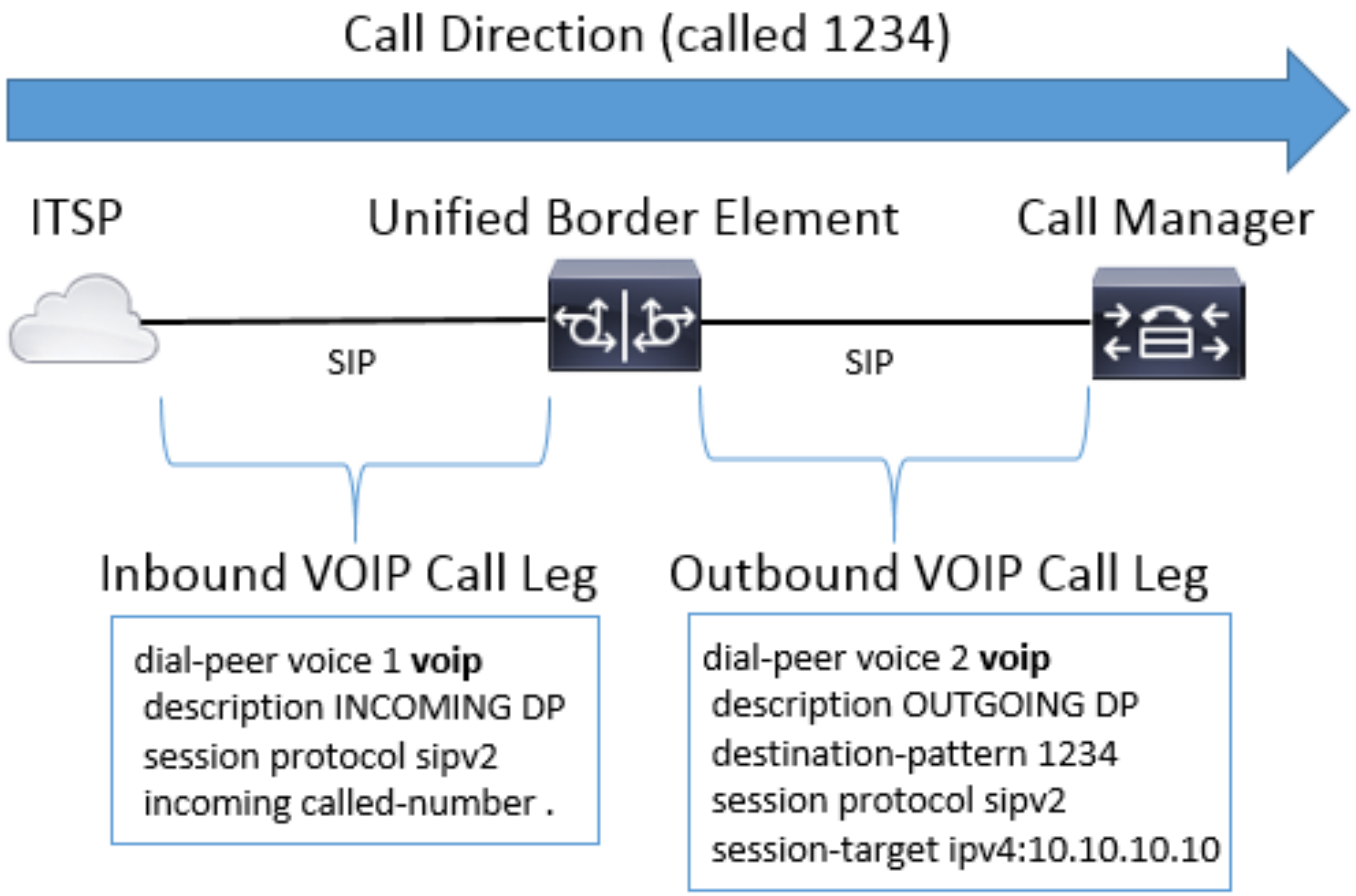


Image 3 - Inbound and Outbound dial-peers for a Voip to VoIP call

Image 4 displays a POTS to POTS call through a Cisco Gateway.

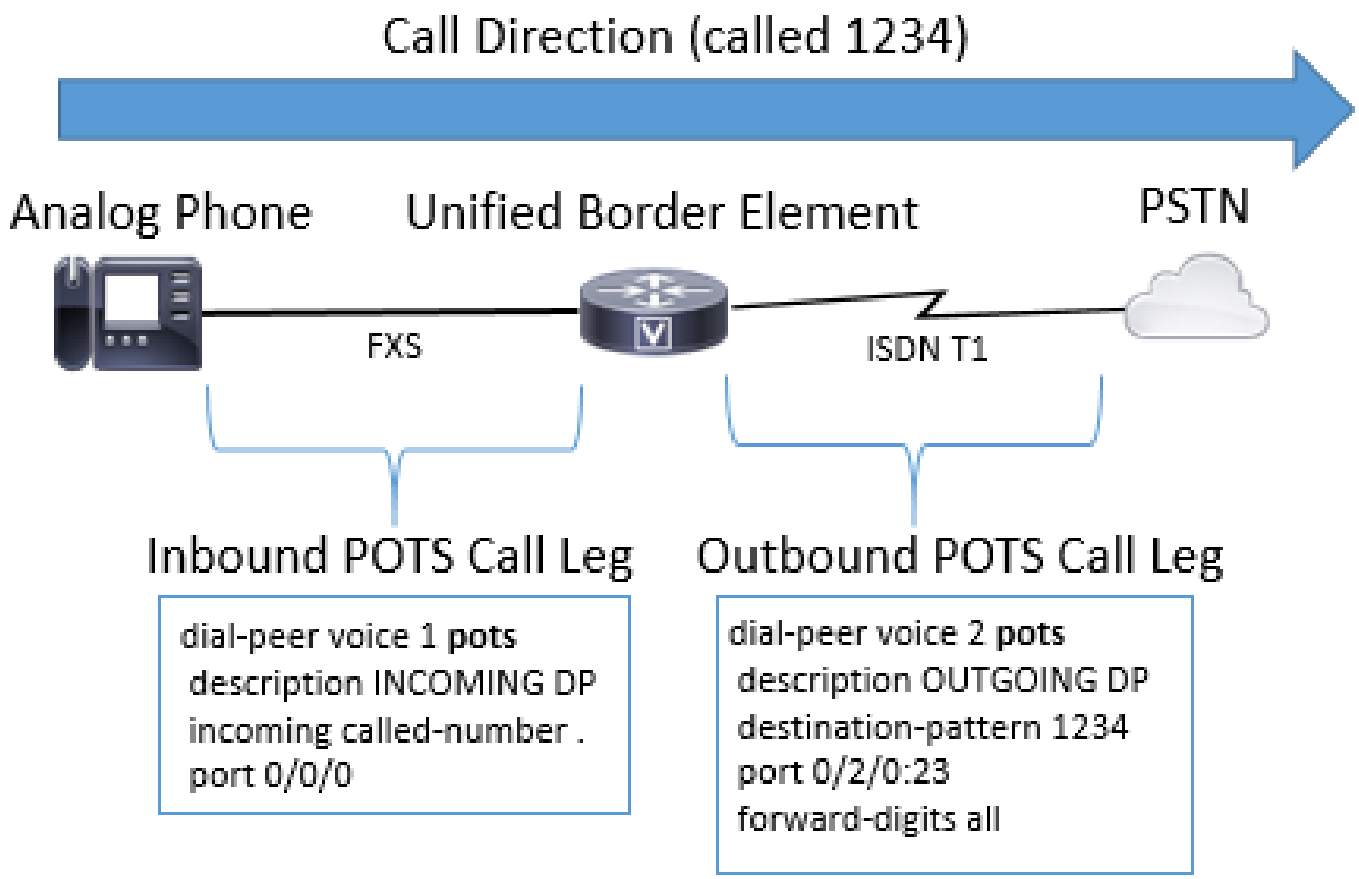


Image 4 - Inbound and Outbound dial-peers for a POTS to POTS call

Voice Dial-Peer Types

POTS	Plain Old Telephony Service dial-peers are matched for analog connections such as Analog FXS, FXO, ISDN T1 / E1s, E1 R2, and Ear and Mouth (E&M) connections. These send or receive a call to / from a physical voice-port on the gateway.
VOIP	Voice Over IP dial-peers are used to mainly control H323 and SIP connections to and from the gateway. These dial-peers send and receive signaling from both IPv4 and IPv6 addresses as well as Fully Qualified Domain Names (FQDN) using Domain Name System (DNS). --- VoIP dial-peers can also be used for Voice over Frame Relay (VoFR), Voice over ATM (VoATM), Voice over High-Level Data Link Control (VoHDLC) and Registration, Admission, and Status (RAS) signaling and session targets for these dial-peers can also include settlements and ENUM values. Note: Some of these types of configurations are older technologies not seen in newer networks and with Cisco IOS XE some are no longer supported. As a result they are not be covered in this document.

MMOIP	Multimedia Mail Over IP dial-peers are utilized to send emails to exchange servers. These are mostly utilized for t37 on-ramp / off-ramp faxing. These dial-peer types are not within the scope of this document.

Note: The maximum number of dial-peers that can be configured on a Cisco gateway depends on the available memory (DRAM). Each dial-peer consumes approximately 6KB of memory, so ensure that the gateway has at least 20% of the total memory reserved for other CPU processes. A large number of configured dial-peers can add to the delay to route a call. This can be significant as the Cisco voice application looks through dial-peers from the top down, similar to an Access Control List (ACL). This is usually not a problem on newer Cisco Gateways.

Sample Error:

```
May 26 12:59:46.406: %DIALPEER_DB-3-ADDPEER_MEM_THRESHOLD: Addition of dial-peers limited by available
```

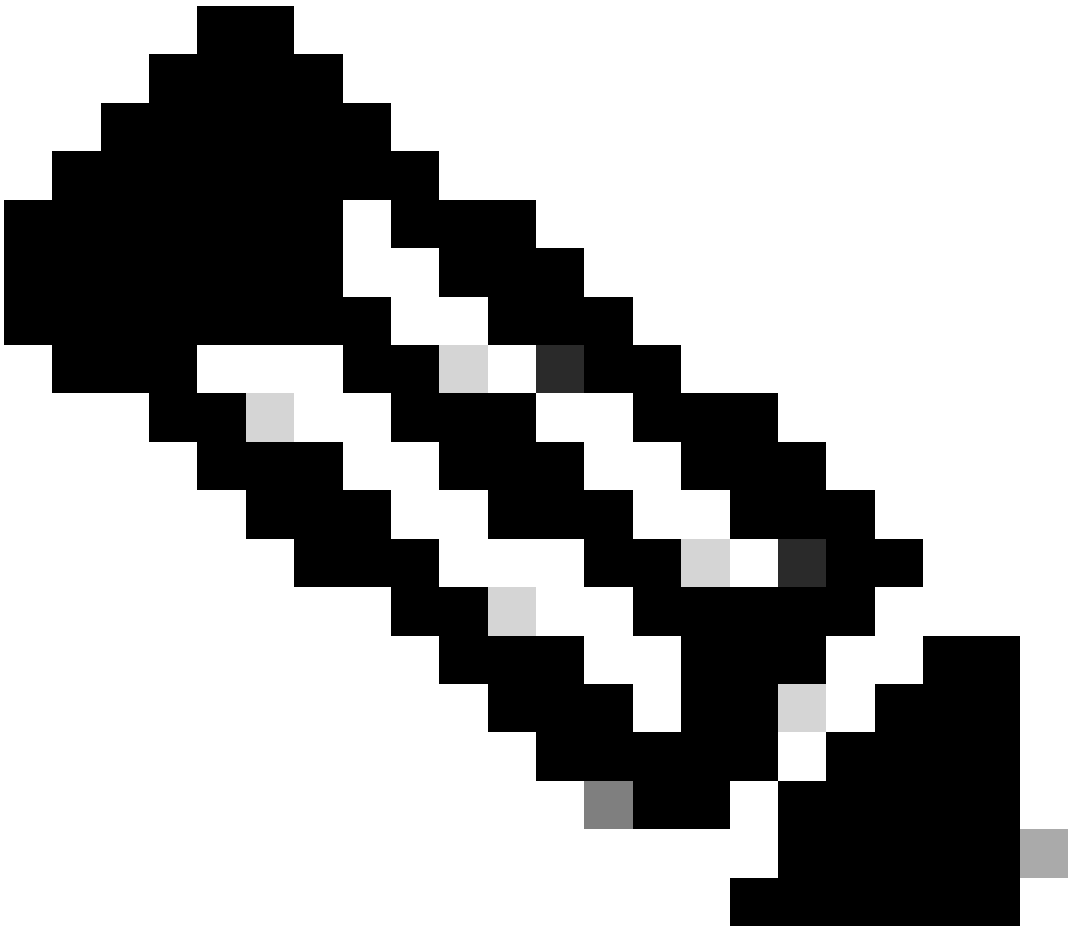
Inbound Dial-Peer Matching

When a Cisco Gateway receives a call setup request, the gateway begins searching for an applicable incoming dial-peer for this call. This is not a digit-by-digit analysis; rather, the full message is utilized to determine which inbound dial-peer is selected. The order of items in the message checked is largely dependent on the protocol for the call as indicated by the preference lists defined in Table 1, Table 2, and Table 3. A dial-peer only needs to satisfy one of the conditions for matching. It is not necessary for all the attributes to be configured in the dial-peer or that every attribute match the call setup information. All dial-peers are searched based on the first match criteria. The gateway moves on to the next criteria only if no match is found.

Table 1. Inbound SIP Dial-Peer Selection Preference

Preference	**Match Criteria**	**Dial-peer Commands**
1	URI	incoming uri via <uri-tag>
2	URI	incoming uri request <uri-tag>
3	URI	incoming uri to <uri-tag>
4	URI	incoming uri from <uri-tag>
5	Called Number	incoming called-number <number-string>

		incoming called e164-pattern-map <pattern-map-number>
6	Calling Number	incoming calling e164-pattern-map <pattern-map-number> answer-address <number-string>
7	Destination-pattern (ANI)	destination-pattern <number-string>
8	Carrier-ID	carrier-id source <string>



Note: Eligible inbound dial-peers can be filtered by VRF, or Tenant. if the applicable feature is configured. For more information, see Virtual Routing and Forwarding (VRF) and Dial-Peer Hunting and Voice Class Tenants sections.

Table 2. Inbound H323 Dial-Peer Selection Preference

Preference	Match Criteria	Dial-peer Commands
1	URI	incoming uri called <uri-tag> incoming uri calling <uri-tag>
2	Called Number	incoming called-number <number-string> incoming called e164-pattern-map <pattern-map-number>
3	Calling Number	incoming calling e164-pattern-map <pattern-map-number> answer-address <number-string>
4	Destination-pattern (ANI)	destination-pattern <number-string>
5	Carrier-ID	carrier-id source <string>

Table 3. Inbound Enbloc POTS Dial-Peer Selection Preference

Preference	Match Criteria	Dial-peer Commands
1	Called Number	incoming called-number <number-string>
2	Calling Number	answer-address <number-string>
3	Destination-pattern (ANI)	destination-pattern <number-string>
4	Voice-port	port <voice-port-number>

When No Matches Exist / Default Dial-Peer 0 peer_tag=0, pid:0

When there are no eligible matches for an inbound dial-peer for either POTS or VoIP calls, the gateway allocates dial-peer 0. This is not ideal as dial-peer 0 has limited capabilities and can cause issues with calls. The outlier to this is SCCP and MGCP protocols which do not use dial-peers for routing calls. See the MGCP and SCCP section for further details.

dial-peer 0 capabilities

- No dtmf-relay mechanisms.
- Advertised all voice codecs for VoIP calls.

- Fax-rate voice.
- Voice Activity Detection (VAD) is enabled.
- No RSVP Support.
- No IVR Application Support for POTS calls.
- Direct-inward-dial is enabled.
- Does not support VRF.

Outbound Dial-Peer Matching

Outbound dial-peers are utilized to route POTS or VoIP calls from the gateway to the next call agent. Like inbound dial-peer matching, there is a list of items the gateway can use to match dial-peers based on the preference order for the specific protocol. However, unlike inbound dial-peers, if there is no eligible outbound dial-peer to route the call, then the call fails. Like inbound dial-peer matching, all dial-peers are searched based on the first match criteria. The gateway moves on to the next criteria only if no match is found.

Table 4. Outbound SIP Dial-Peer Selection Preference

Preference	**Match Criteria**	**Dial-peer Commands**
1	Dial-Peer Group Dial-Peer	destination dpg <dpg-tag> *(DPG configured on inbound dial-peer)*
2	Dial-Peer Provision Policy URI	destination uri-from <uri-tag> destination uri-to <uri-tag> destination uri-via <uri-tag> destination uri-diversion <uri-tag> destination uri-referred-by <uri-tag> *(DPP configured on inbound dial-peer)*
3	ILS Route String	destination route-string <route-string-tag>
4	URI and Carrier-ID	destination uri <uri-tag> AND carrier-id target <string>
5	Called Number and Carrier-ID	destination-pattern <number-string> AND carrier-id target <string>
6	URI	destination uri <uri-tag>
7	Called Number	destination-pattern <DNIS-number> destination e164-pattern-map <pattern-map-number> dnis-map <dnis-map-number>

8	Calling Number	destination calling e164-pattern-map <pattern-map-number>

Note: When both incoming and outgoing dial peers are configured with **calling e164-pattern-map**, it is essential to apply a **dial-peer provision policy** to the incoming dial peer to avoid call failures.

Table 5. Outbound H323 Dial-peer Selection Preference

Preference	**Match Criteria**	**Dial-peer Commands**
1	Dial-Peer Group Dial-Peer	destination dpg <dpg-tag> *(configured on inbound dial-peer)*
2	URI and Carrier-ID	destination uri <uri-tag> AND carrier-id target <string>
3	Called Number and Carrier-ID	destination-pattern <number-string> AND carrier-id target <string>
4	URI	destination uri <uri-tag>
5	Called Number	destination-pattern <number-string> destination e164-pattern-map <pattern-map-number> dnis-map <dnis-map-number>
6	Calling Number	destination calling e164-pattern-map <pattern-map-number>

Table 6. Outbound POTS Dial-peer Selection Preference

Preference	**Match Criteria**	**Dial-peer Comamnds***
1	Dial-Peer Group Dial-Peer	destination dpg <dpg-tag>(configured on inbound dial-peer)
2	URI and Carrier-ID	destination uri <uri-tag> AND carrier-id target <string>
3	Called Number and Carrier-ID	destination-pattern <number-string> AND carrier-id target <string>
4	URI	destination uri <uri-tag>

5	Called Number	destination-pattern <DNIS-number>dnis-map <map-number>

Note: The Number String Dial-Peer Hunting and URI Dial-Peer Hunting section go into how the gateway evaluates a list of potential commands for each match criteria row before moving to the next match criteria. For example, it evaluates all potential destination-pattern matches and destination e164-pattern-map matching commands before it examines the calling number commands.

Number String Dial-Peer Hunting

Number String Preference:

Much like URIs have a specific order of operations for evaluating matches, there is also a set of rules used when evaluating a numeric digit-string. The default dial-peer hunt scheme for a Cisco gateway is set to 0. This means the gateway searches for a pattern with the longest match (most specific). If there are two dial-peers with the same match length, the gateway looks at the explicitly defined dial-peer preference. Lastly, if both of those are the same, it chooses one in a random order.

There are other dial-peer hunt schemes available for configuration; however, most deployment keep the default of 0.

Tip: If dial-peers are being matched outside the default order, an administrator can examine the running configuration for a non-default dial-peer hunt scheme.

```
Gateway(config)# dial-peer hunt ?
  <0-7>  Dial-peer hunting choices, listed in hunting order within each choice:
  0 - Longest match in phone number, explicit preference, random selection.
  1 - Longest match in phone number, explicit preference, least recent use.
  2 - Explicit preference, longest match in phone number, random selection.
  3 - Explicit preference, longest match in phone number, least recent use.
  4 - Least recent use, longest match in phone number, explicit preference.
  5 - Least recent use, explicit preference, longest match in phone number.
  6 - Random selection.
  7 - Least recent use.
```

The longest match number string dial-peer algorithm finds the dial-peer with the most numbers in a sequence that exactly match a sequence of numbers in a number string. This concept is clarified in the subsequent scenario.

Scenario: Eligible dial-peers have been configured with these possible matches, and the gateway is evaluating a digit-string of 2001. Dial-peer 1 can match any number 2000 through 2999 while dial-peer 2 can match 2000 through 2009. Dial-Peer 2 would be matched for this call as it is the longest match (most specific) for the digit string 2001 when the default dial-peer hunting mechanisms is employed (dial-peer hunt 0). In other words, the sequence of numbers 200 is the largest sequence that exactly matches a sequence of number in the number string 2001.

!

```
dial-peer voice 1 voip
 destination-pattern 2...
!
dial-peer voice 2 voip
 destination-pattern 200.
!
```

Preference is defined as the administrator defined weight for each dial-peer. Administrators can configure a preference so the call always uses a specific dial-peer before others. By default, all dial-peers are preference 0. A dial-peer with preference 0 is matched before another dial-peer with preference 1 through 10. Most administrators setup multiple dial-peers to send a call to a specific CUCM subscriber with a backup subscriber or another call agent being configured using another dial-peer with a lower preference (which is configured with a higher number).

Scenario**:** Two dial-peers are configured with the same match length for the digit string of 2001. The administrator defines an explicit preference. The gateway evaluates both dial-peers the same since their match length is the same. However, the administrator sets dial-peer 1 with a higher preference so that dial-peer is chosen as the first dial-peer used in routing the call. Dial-Peer 2 would remain as a secondary option can a failure occur on the first dial-peer.

```
!
dial-peer voice 1 voip
 destination-pattern 2...
 preference 1
!
dial-peer voice 2 voip
 destination-pattern 2...
 preference 2
!
```

A Cisco gateway only attempts to route a call via one eligible outbound dial-peer at a time. If a failure condition is observed on the first selected dial-peer, then the gateway attempts to route the call out the next eligible dial-peer. This continues until the call either succeeds or fails because there are no more eligible dial-peers left to try. A common symptom of dial-peer hunting and failure is a noticeable delay in ringback while making calls. Debugs are usually needed to verify exactly why the call is failing on a given dial-peer. The command **huntstop** can be employed on a dial-peer if an administrator does not want a gateway to look for another dial-peer when a failure condition is observed.

Scenario**:** Two dial-peers are configured with the same match length for the digit string of 2001. The administrator has defined an explicit preference and does not want to match dial-peer 2 for this particular call. Since there are two dial-peers with the same match-length, the preference is used to determine the dial-peer. Dial-Peer 1 has the lowest configured preference number, so it is used to route the call. If a failure condition occurs on the outbound call leg using dial-peer 1, then the gateway immediately stops dial-peer hunting since the **huntstop** command is configured. In this scenario, dial-peer 2 is never utilized for outbound routing.

```
!
dial-peer voice 1 voip
 destination-pattern 2...
 preference 1
 huntstop
```

```
!
dial-peer voice 2 voip
 destination-pattern 2...
 preference 2
!
```

Note: **huntstop** and **preference** commands can also be used in conjunction with URI matching statements as they are general dial-peer configuration commands. Furthermore, voice class server-group configurations can utilize **huntstop** commands in 17.4.1a. Refer to the section Destination Server-Groups for more details on this.

URI Dial-Peer Hunting

The gateway looks at each match criteria and exhausts it before it moves to the next match criteria. An example of this would be on an inbound SIP call. Based on Table 1. Inbound SIP Dial-Peer Selection Preference, the first thing the Cisco gateway checks is the URI and evaluates all potential URI commands to find one that fits. If there is no match, or none are configured, then the gateway moves to the next matching item and performs an evaluation on that criteria. This process repeats until the call either routes based on a match or the gateway runs out of match criteria to check.

When an inbound or outbound dial-peer is configured with a URI command, the gateway examines the URI that was received in multiple headers for a potential match. The match preference is based on the most specific match and the exact preference goes Full URI match, Host Portion, User Portion, or telephone URI. Knowing the order of operations for URI matching can greatly aid in dial-peer matching with SIP and CUBE deployments.

This preference order can be manipulated using the command **voice class uri sip preference** to specify the user-id as the first option instead of host.

URI Preference:

1. The Host portion of the URI. Examples: (@<ip_address> or @example.com)
2. The User portion of the URI. Examples: (sip:8675309 or sip:user)
3. The tele-uri. Example: (tel:18005532447)
4. Exact match for the full URI. Examples: (user@host.domain.name, user@<ip_address>, 8675309@example.com, 8675309@example.com)

Supporting Document: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Scenario: An administrator has configured this dial-peers and sends a call to the gateway. The From header in the received Invite is From: <sip:testuser@192.0.2.1>. The gateway can potentially match two different dial-peers based on this header. Dial-Peer 1 based on the user portion and dial-peer 2 based on the host portion. However, since a host match is a preference over a user match, dial-peer 2 is used for the inbound dial-peer in the call.

```
!
voice class uri URI1 sip
 user-id testuser
!
voice class uri URI2 sip
 host ipv4:192.0.2.1
!
```

```
dial-peer voice 1 voip
 sess protocol sipv2
 incoming uri FROM URI1
!
dial-peer voice 2 voip
 sess protocol sipv2
 incoming uri FROM URI2
!
```

Voice Class URI

URI matching for inbound and outbound dial-peers allows an administrator the ability and flexibility to perform matches on more than a phone number string for VoIP protocols supporting URIs in their messaging. Prior to Cisco IOS 15.4(1)T and Cisco IOS-XE 3.11S, a request URI had to contain an alphanumeric user@host else a Cisco gateway would reject the call with a 4xx message. Now a URI can contain just the host portion, and the gateway routes the call based on just the host provided. For example, sip:cisco.com.

Additionally, prior to Cisco IOS 15.4(1)T and Cisco IOS-XE 3.11S voice-class URI user-ids could only be numeric e.164 values (sip:1234@example.com). This was changed so administrators can configure alphanumeric user-ids on CUBE (sip:user@example.com).

The host or user portion of a voice class uri can contain regular expressions (regex) patterns which greatly expands the possible values that can be matched.

```
Gateway(config-voice-uri-class)# user-id .)
% unmatched ()user-id pattern can be of format ^([][0-9A-Za-z\|\/()*+^$&?#--.])*$

Gateway(config-voice-uri-class)# host .)
% unmatched ()host pattern can be of format ^([][0-9A-Za-z\|@\/()*+^$&?#--.])*$

Gateway(config-voice-uri-class)# pattern .)
% unmatched ()pattern pattern can be of format ^([][0-9A-Za-z\|@;:=%!~\/()*+^$&?#--.])*$
```

Example: Voice Class URIs

```
!
voice class uri HOST sip
 host example.com
 host dns:cisco.example.com
 host ipv4:10.50.244.2
 host ipv6:[2001:4860:4860::8888]
!
voice class uri USER sip
 user-id username
!
voice class uri PATTERN sip
 pattern 8675309
!
voice class uri HostRegex sip
```

```
 host (.*)example.com
!
voice class uri ipRegex sip
 host 172\.18\.110\.20[567]
!
voice class uri PatternRegex sip
 pattern 555(.*)
!
voice class uri ipRegex sip
 pattern (172\.18\.110\.10[134]|10\.10\.10\.10)
 ! One Line that matches 172.18.110.101, 172.18.110.103, 172.18.110.104 OR 10.10.10.10
!
voice class uri UserRegex sip
 user-id test(.*)
!
```

Only 10 hosts, 1 pattern, or 1 user-id can be configured per voice class uri, as this example demonstrates. If more items need to be matched, it is recommended to use Regex.

```
Gateway(config)# voice class uri TEST sip
Gateway(config-voice-uri-class)#host ipv4:10.1.1.1
Gateway(config-voice-uri-class)#host ipv4:10.2.2.2
Gateway(config-voice-uri-class)#host ipv4:10.3.3.3
Gateway(config-voice-uri-class)#host ipv4:10.4.4.4
Gateway(config-voice-uri-class)#host ipv4:10.5.5.5
Gateway(config-voice-uri-class)#host ipv4:10.6.6.6
Gateway(config-voice-uri-class)#host ipv4:10.7.7.7
Gateway(config-voice-uri-class)#host ipv4:10.8.8.8
Gateway(config-voice-uri-class)#host ipv4:10.9.9.9
Gateway(config-voice-uri-class)#host ipv4:10.10.10.10
Gateway(config-voice-uri-class)#host ipv4:10.11.11.11
Error:Maximum of 10 hosts can only be configured.

Gateway(config)# voice class uri TEST2 sip
Gateway(config-voice-uri-class)#host dns:1.com
Gateway(config-voice-uri-class)#host dns:2.com
Gateway(config-voice-uri-class)#host dns:3.com
Gateway(config-voice-uri-class)#host dns:4.com
Gateway(config-voice-uri-class)#host dns:5.com
Gateway(config-voice-uri-class)#host dns:6.com
Gateway(config-voice-uri-class)#host dns:7.com
Gateway(config-voice-uri-class)#host dns:8.com
Gateway(config-voice-uri-class)#host dns:9.com
Gateway(config-voice-uri-class)#host dns:10.com
Gateway(config-voice-uri-class)#host dns:11.com
Error:Maximum of 10 hosts can only be configured.

Gateway(config)# voice class uri TEST3 sip
Gateway(config-voice-uri-class)#user-id 8675309
Gateway(config-voice-uri-class)#user-id 123456789
Gateway(config-voice-uri-class)#do sh run | s TEST3
voice class uri TEST3 sip
user-id 123456789

Gateway(config)# voice class uri TEST4 sip
Gateway(config-voice-uri-class)#pattern 8675309
Gateway(config-voice-uri-class)#pattern 123456789
```

```
Gateway(config-voice-uri-class)#do sh run | s TEST4
voice class uri TEST4 sip
pattern 123456789
```

Inbound URI Dial-Peer Matching

This feature was added in Cisco IOS 15.1(2)T and Cisco IOS-XE 3.8S and utilizes a voice class uri configured and applied to an inbound dial-peer. Incoming URI has been adopted by many people over the traditional incoming called-number statement for SIP calls as it the first match criteria checked when selecting inbound dial-peers. The command also allows administrators the ability to better match calls that come from a particular call agent or user.

Full Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Common Use Cases

1. An inbound dial-peer matching on the host portion of the URI to answer OPTIONS ping requests from CUCM.
2. An inbound dial-peer matching on the host portion of the URI to control inbound calls from an Internet Telephony Service Provider (ITSP)
3. An inbound dial-peer matching on the user-id portion of the URi for call treatment from certain users or numbers.

Configuration Example

This example output matches dial-peer 777 for any SIP request sourcing from the two HOST IPs defined in the voice class URI. The header watched is defined as the From header on the dial-peer; however, an administrator can define many others including VIA, TO, and REQUEST (Request URI). If the CUCM sends an OPTIONS ping to the CUBE now matches dial-peer 777 and source my 200 OK reply to the OPTIONS from the specified interface. If CUCM sends an Invite to the CUBE matches dial-peer 777 as the inbound dial-peer.

```
!
voice class uri CUCM sip
 host ipv4:10.50.244.2
 host ipv4:10.50.244.20
!
dial-peer voice 777 voip
 description INCOMING URI
 session protocol sipv2
 incoming uri from CUCM
 voice-class sip bind control source-interface Loopback777
 voice-class sip bind media source-interface Loopback777
!
```

Outbound URI Dial-Peer Matching

Cisco IOS gateways can match an outbound dial-peer using a URI by applying a voice class uri to an outbound dial-peer and adding call-route url to global configuration. When this is present, the CUBE can try to route calls based on the Request URI. This feature was added in Cisco IOS 12.3(4)T and is present in all

Cisco IOS XE versions. It can be noted that by default the outgoing SIP Request-URI and To header URI have the session target of the outbound-dial-peer. This can be disabled by using the command **requri-passing** which allows the gateway to pass the in-leg URI host portion to the out-leg instead of replacing the URI host portion with the session-target. The command **requri-passing** was added in 15.4(1)T and Cisco IOS XE 3.11S.

Configuration Example

```
voice service voip
 sip
  call-route url
  requri-passing
!
voice class uri CUCM sip
 host dns:.*.com
!
dial-peer voice 777 voip
 description OUTGOING URI
 session protocol sipv2
 destination uri CUCM
 session target sip-uri
!
```

Source: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

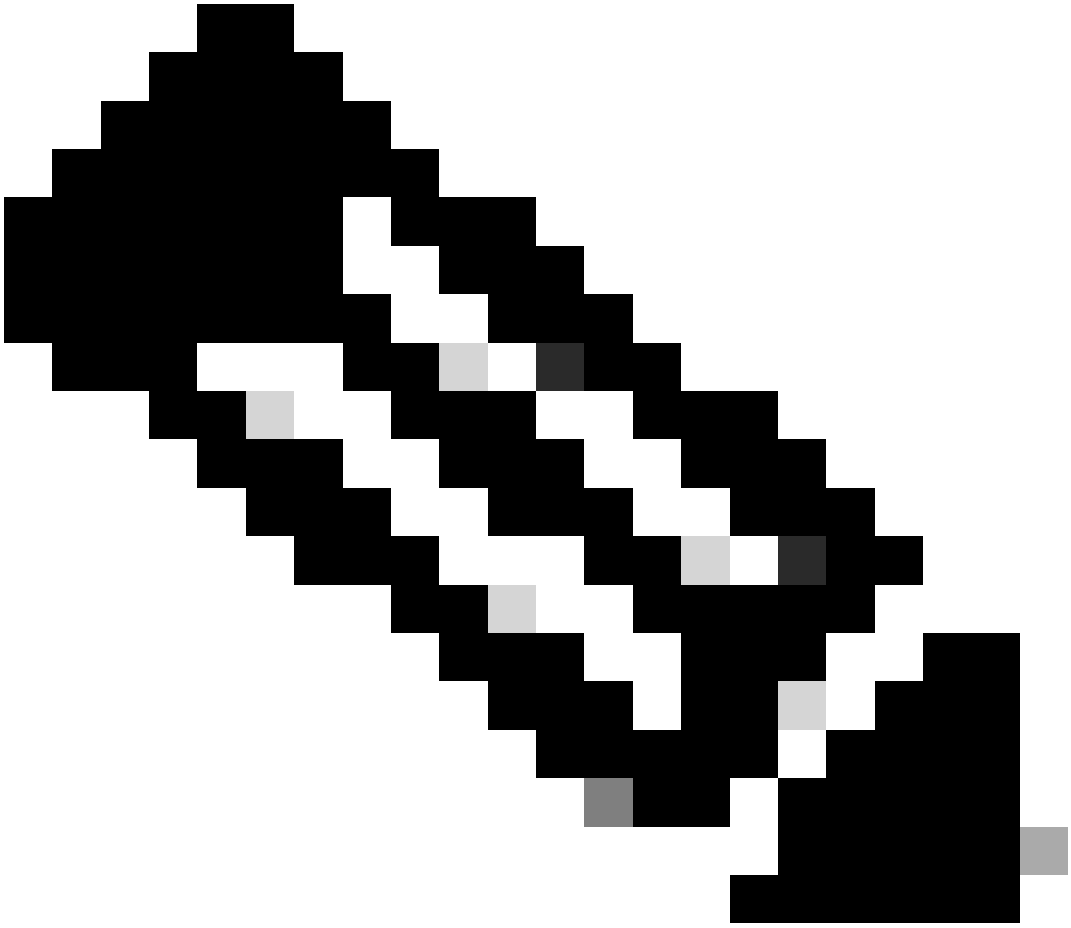
In addition to voice class URI, administrators can use a dial-peer provision-policy (DPP) to match an in-leg URI for an outbound dial-peer match. This feature was added in Cisco IOS 15.4(2)T and Cisco IOS XE 3.12S. A dial-peer provision-policy requires defining a primary match attribute with a secondary match attribute being optional. The provision-policy is applied to an inbound dial-peer, and when that dial-peer is selected for use on an inbound call-leg, the policy is invoked. The result is an outbound dial-peer selection based on the attribute from the dial-peer provision-policy.

The outbound match can be a single header or multiple headers which all must be true in order to match the dial-peer.

In the example, there is a voice class uri for the From and To headers. For an OR match, a dial-peer provision-policy is configured which contains two preferences. The From header is the first preference, and the To header is the backup preference. Dial-Peer 1234 is built to apply the provision policy for inbound matching. Then built dial-peer 11111 and 22222 which apply the destination uri-from and destination uri-to commands respectively. These commands point back to their voice class URI. For the call, you can receive the Invite, match dial-peer 1234 and check the provision-policy. The device can then try to route on the From header first which as an applicable match on dial-peer 11111. If this fails, you can also try to route on the to header with 22222.

The example also details how to achieve an And match with dial-peer provision-policies. Assuming the same Invite is received, you can define two headers under one preference and apply this to the inbound dial-peer.

Now when the invite is received, it can check for eligible outbound dial-peers which satisfy both matching criteria defined in the provision-policy. So in this example, your outbound dial-peer needs to be defined with both the TO and FROM header in order to be matched. If either are not a valid match, this dial-peer 12345 is not used.



Note: Although we are routing the call on the From header the Invite that leaves the gateway still has the original Request URI. We simply use the Dial-peer provision-policy to match an outbound dial-peer not change the request URI.

Configuration Example

<#root>

```
### Received INVITE

Received:
INVITE sip:8675309@172.18.110.58:5060 SIP/2.0
From: sipp <sip:sipp@172.18.110.65>;tag=1
To: sut <sip:cube@172.18.110.58:5060>



### Common Configurations


!
```

```
voice class uri FROM sip
 user-id sipp
!
voice class uri TO sip
 user-id cube
!
```

OR Match

```
!
voice class dial-peer provision-policy 1
 description match from header. If false, try to header
 preference 1 from
 preference 2 to
!
dial-peer voice 1234 voip
 session protocol sipv2
 destination provision-policy 1
 incoming called-number .
!
dial-peer voice 11111 voip
 destination uri-from FROM
 session protocol sipv2
 session target ipv4:172.18.110.48
!
dial-peer voice 22222 voip
 destination uri-to TO
 session protocol sipv2
 session target ipv4:172.18.110.48
!
```

AND Match

```
!
voice class dial-peer provision-policy 2
 description match from AND to headers
 preference 1 from to
!
dial-peer voice 1234 voip
 session protocol sipv2
 destination provision-policy 2
 incoming called-number .
!
dial-peer voice 12345 voip
 destination uri-from FROM
 destination uri-to TO
 session protocol sipv2
 session target ipv4:172.18.110.48
!
```

Source: [Cisco Unified Border Element Configuration Guide Through Cisco IOS XE 17.5]

session target sip-uri

Prior to Cisco IOS 15.4(1)T and Cisco IOS XE 3.11S if the host portion of a URI was different, but the user was the same, this would require two separate outbound dial-peers.

After this release, an administrator can configure one dial-peer to service multiple hosts for the same user. For example, testuser@example.com and testuser@example.net under the same dial-peer. Using session target sip-uri triggers DNS resolution of the domain of incoming Invite Req-URI and dynamically determine the session target IP.

Example Configuration:

The gateway recieves two SIP Invites with these headers Invite sip:testuser@example.com:5060 SIP/2.0 Invite sip:testuser@example.net:5060 SIP/2.0 The gateway matches the incoming SIP request of testuser@example.com and testuser@example.net on dial-peer 1 because of the incoming URI command and the user-id definition both match testuser. The command voice-class sip call-route url is present means you evaluate outbound dial-peers based on the request URI of this inbound Invite. You match dial-peer 2 because of the same reasons you matched dial-peer 1, the user-id of testuser. The session target of this dial-peer is the original sip-uri as defined by session target sip-uri, which was a FQDN. After a DNS resolution has taken place, and change example.com and example.net into an IP for layer 3 routing you send a message out of the gateway.

```
!
ip host example.com 192.0.2.10
ip host example.net 192.0.2.10
!
voice class uri TEST-IN sip
 user-id testuser
!
dial-peer voice 1 voip
 description INCOMING dial-peer
 incoming uri request TEST
 session protocol sipv2
 voice-class sip call-route url
!
dial-peer voice 2 voip
 description OUTBOUND dial-peer
 destination uri TEST
 session protocol sipv2
 session target sip-uri
!
```

Verification:

```
show voice class uri <uri-name>
show voice class dial-peer provision-policy <number>
debug voip uri
```

Dial-Peer Wildcards

An administrator can utilize dial-peer wildcards when defining inbound and outbound matching mechanisms that involve a number string. These include destination-pattern, incoming called-number, e164-pattern-maps, and answer-address as well as the prefix command. Dial-peer wildcards are regular expressions

(regex) available for configuration which allow greater flexibility over the matching of dial-peers.

Wildcard Table

Character	Definition	Examples
*****	On a dial-peer this is a literal value of * (star) on the keypad.	12345*
#	On a dial-peer this is a literal value of # (pound) on the keypad.	8675309#
,	Inserts a 1 second pause between digits.A comma can also be used within brackets [ ] to break up a continuous range.	9,,,,,55591[1-3,5-9]8675309
.	Regex character for matching any value 0-9, A-F and *, #, + Up to 15 dot characters can be defined per dial-peer although the CLI lets an administrator configure as many as they see fit. If more than 15 dots are require please use T.	2.... 91[2-9]..[2-9].....
%	Regex for preceding digit occurring zero or more times.	
+	When used at the beginning of a string it means a literal + used in E164 numbers. When used anywhere else in the string it is a regex value for the preceding digit occurring one or more times.	+19191112222
?	Regex for the preceding digit occurring zero or one time.	(206)?5015111 (0)?(1)?(1)?21933...
^	Regex character to indicate the start of the string when used outside of brackets When used inside brackets it is treated as an exclude or a DO NO MATCH Statement This is no longer required in later versions as the gateway automatically assume a ^ when processing a regex string without a ^.	^8675309 91[^135]555
$	Regex character to indicate the end of a string.	8675309$
\	Escape character to mean a literal value	

[ ]	Brackets define a range of characters for a single position. Commas must be used to break up continuous strings.	[1-5]0000 [2,5-8]0000
( )	Parenthesis define a group of characters in a set.	9(258)7777
T	A variable length match of up to 32 digits. The router waits on the inter-digit timeout to occur before routing the call. The default value for the interdigit timeout is 10 seconds and is modifiable via timeouts inter-digit on a voice-port. Administrators can terminate the inter-digit timeout using # on the keypad. This is modifiable via the command dial-peer terminator configured globally. T also references the T302 timer.	9011T
-	Used in brackets to define the range.	[5-9]1234

Output from Gateway that displays the possible regular expression inputs.

```
Gateway(config-dial-peer)# destination-pattern asdfqw4r3~2
Incorrect format for E.164 Number
        regular expression must be of the form  ^[][^0-9,A-F#*.?+%()-]*T?(\$)?$
```

Dial-Peer States

Dial-peers can be in one of two Operational states.

1. Up
2. Down

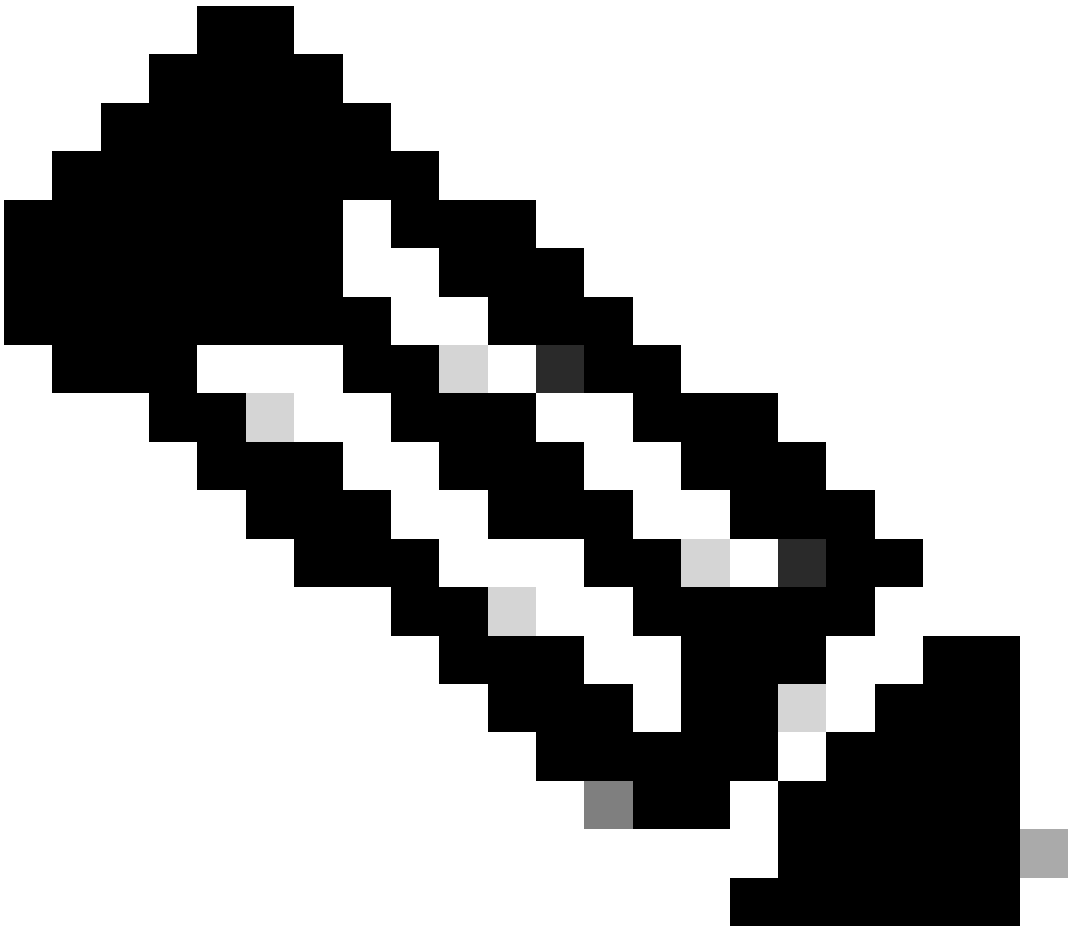
For a dial-peer to be in a valid operational state, and eligible for use with call routing, it needs to be in the UP state. For outbound VOIP dial-peers, this means there can be a valid outbound matching mechanisms as well as a valid session target to route the call towards. For outbound POTS dial-peers, a valid outbound matching mechanism as well as a valid voice-port can be configured. With inbound dial-peers only, a valid inbound matching mechanisms must be configured.

The busyout state is seen when a dial-peer is configured with a keepalive mechanisms and the remote target has failed the parameters of that keepalive mechanism. The gateway then moves the dial-peer into a busyout state so it is no longer used for call routing decisions, and when the keepalive mechanism is fulfilled again, the gateway puts the dial-peer back into an up state. If a dial-peer is selected as an outbound dial-peer, and this dial-peer is in a busyout state, the gateway fails the call with a cause code 188.

Along with operational states there are Administrative states.

1. Up
2. Down

An administrator can disable a dial-peer without removing it from the configuration by entering the **shutdown** command on the dial-peer. To re-enable the dial-per enter **no shutdown**.



> **Note**: A dial-peer with a voice-port that is down, shutdown, or not operational, remains in operational state of Up however the Out State shows as Down.

Verification

```
Gateway# show dial-peer voice summary
dial-peer hunt 0
             AD                                      PRE PASS                 OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN       FER THRU SESS-TARGET     STAT PORT    KEEPALIVE
1      voip  up   up                                 0  syst
777    voip  up   up             9...               0  syst ipv4:10.50.244.2
```

```
555     voip  up   down          555               0  syst
888     pots  up   up            888               0                       up   0/2/0
999     pots  up   up            999               0                       down 0/2/0
123     voip  up   up            123               0  syst ipv4:10.10.10.10            busyout
```

Virtual Routing and Forwarding (VRF) and Dial-Peer Hunting

Inbound Dial-Peer Matching with VRF

Starting with Cisco IOS 15.6(3)M and Cisco IOS-XE 16.3.1, Cisco gateways can match inbound dial-peers using VRF IDs. To take advantage of this, an administrator must bind the inbound dial-peer to an interface which in turn binds the dial-peer to the VRF ID on the specified interface. After the bind is complete, inbound calls are filtered by the Cisco Gateway to only only include eligible inbound dial-peers that match the VRF ID of the interface the packet was received on. From here the inbound dial-peer is matched based on regular dial-peer matching order of operations.

Prior to these Cisco IOS / Cisci IOS-XE releases, the Cisco Gateway would make an inbound selection based on regular inbound dial-peer matching without any filtering. This means a VRF1 call could be matched by a VRF2 dial-peer. Additionally, since only one VRF was supported by H323 and SIP prior to these releases other issues arise when attempting to use multi-VRF features. The use of a single VRF for voice applications was known as VRF-Aware configuration.

Full VRF-Aware Documentation: VRF-Aware H.323 and SIP for Voice Gateways

Full Multi-VRF Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Outbound Dial-Peer Matching with VRF

Cisco Gateways have the ability to bridge calls across VRFs without the need for route leaks to be configured. This means an inbound call on VRF1 can be routed outbound on a dial-peer for VRF2 if the normal outbound dial-peer matching selection is satisfied. Dial-peer groups can be employed to force the Cisco gateway to keep the call within the same VRF.

VRF and Dial-Peer Group Configuration Example

This configuration example has VRF1 and VRF2 with two overlapping IP Ranges and two overlapping phone number ranges.

Utilize VRF binding to ensure the correct inbound dial-peer is matched, and Dial-peer Groups to ensure the correct VRF bound outbound dial-peer is matched. If a SIP packet for a call to 8675309 arrives on gig0/0/1.2, then the gateway filters out all available inbound dial-peers based on the VRF2 ID. This means you cannot match dial-peer 10. Now when you check the digit-string, you can match dial-peer 20. Dial-peer 20 has a dial-peer group which tells the gateway the only outbound dial-peer that can be matched is also dial-peer 20. This dial-peer group allows you to avoid matching dial-peer 10 and crossing a call coming from VRF1 into VRF2. From there the call can proceed as normal.

```
!
interface GigabitEthernet0/0/1.1
 description VRF1
```

```
 encapsulation dot1Q 10
 ip vrf forwarding VRF1
 ip address 10.10.10.10 255.255.255.0
!
interface GigabitEthernet0/0/1.2
 description VRF2
 encapsulation dot1Q 20
 ip vrf forwarding VRF2
 ip address 10.10.10.10 255.255.255.0
!
voice service voip
 no ip address trusted authenticate
 media-address voice-vrf VRF1
 media-address voice-vrf VRF2
 allow-connections sip to sip
 sip
!
voice class dpg 10
 description INBOUND VRF1 to OUTBOUND VRF1
 dial-peer 10 preference 1
!
voice class dpg 20
 description INBOUND VRF2 to OUTBOUND VRF2
 dial-peer 20 preference 1
!
dial-peer voice 10 voip
 description VRF1
 destination-pattern 8675309
 session protocol sipv2
 session target ipv4:10.10.10.20
 destination dpg 10
 incoming called-number 8675309
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
!
dial-peer voice 20 voip
 description VRF2
 destination-pattern 8675309
 session protocol sipv2
 session target ipv4:10.10.10.20
 destination dpg 20
 incoming called-number 8675309
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
!
```

Verification

```
Gateway# show vrf brief | i VRF
  VRF1                             1:1                  ipv4        Gi0/0/1.1
  VRF2                             2:2                  ipv4        Gi0/0/1.2
```

```
Gateway# show dial-peer voice summary
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN      FER THRU SESS-TARGET    STAT PORT    KEEPALIVE    VR
10     voip  up   up             8675309            0  syst ipv4:10.10.10.20                        VR
20     voip  up   up             8675309            0  syst ipv4:10.10.10.20                        VR
```

```
Gateway# show voice class dpg 10
Voice class dpg: 10      AdminStatus: Up
 Description: INBOUND to OUTBOUND VRF1
 Total dial-peer entries: 1

 Peer Tag       Pref
 --------       ----
 10             1
-------------------------------------
```

```
Gateway# show voice class dpg 20
Voice class dpg: 20      AdminStatus: Up
 Description: INBOUND to OUTBOUND VRF2
 Total dial-peer entries: 1

 Peer Tag       Pref
 --------       ----
 20             1
-------------------------------------
```

Advanced Call Routing Techniques

Over the years as business needs grow, the company expands and requires more DIDs and enterprise administrators can find that the basic dial-peers do not meet scale well. There can be on-off situations that need to be addressed, or perhaps there are just too many dial-peers in general. Having thousands of dial-peers does not make administration and troubleshooting easy. Having a dial-peer for each specific CUCM server or call agent starts to compound the problem of too many dial-peers because now an administrator needs to configure a dial-peer for for each digit-string. If there is have more than one SIP provider connecting to a gateway, or a few different people using the same CUBE, this makes isolating a specific tenant very tough.

Cisco has taken this feedback and created a set of items that can address these issues and more. Dial-peer Groups, Voice Class tenants, destination server-groups, e164-pattern-maps and POTS trunk groups allow for an administrator to solve all of the problems listed and many more not listed.

Dial-Peer Groups

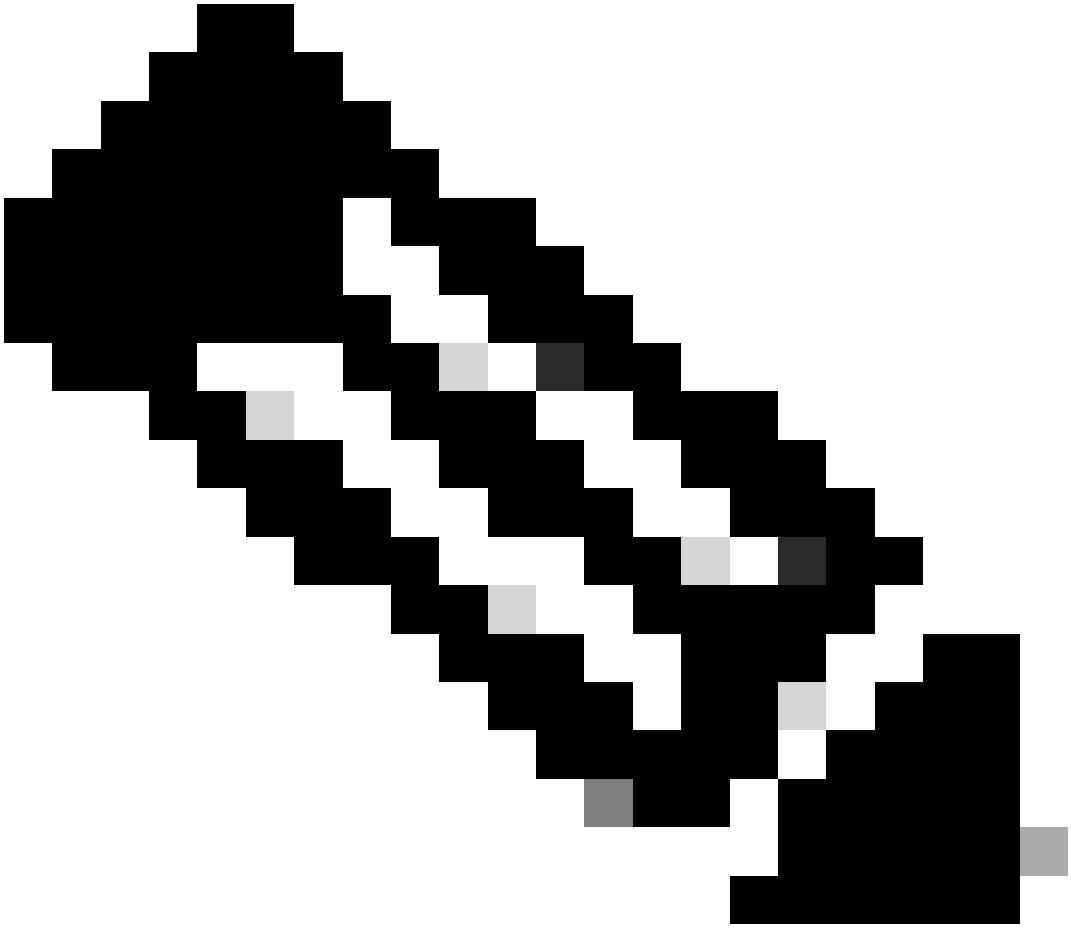
Dial-peer groups were added in Cisco IOS 15.4(1)T and Cisco IOS-XE 3.11S and POTS dial-peers were added as an option in Cisco IOS 15.5(1)T and Cisco IOS-XE 3.14S. A dial-peer group allows administrators to specify an exact dial-peer for outbound routing based on the inbound dial-peer matched. Once an inbound dial-peer with a dial-peer group configured is matched, the call uses the dial-peer defined in the dial-peer group even if the destination-pattern does not match. The only prerequisite is the outbound dial-peer must be Up so an outbound matching method must be configured, however, this is not actually used to route the call.

The best way to describe dial-peer groups is to liken them to the concept of static routes in a routing table. These are static inbound to outbound routing decisions that take away some of guesswork for the gateway because they are telling it exactly how to route the call.

Full Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Configuration Example

With this example, the called number is 8675309. This matches dial-peer 1234 based on the incoming called-number statement. This dial-peer is configured with a dial-peer group that states the call can now route out dial-peers 2, then 3, and finally 1 if dial-peer 2 fails. This the gateway so now try to route the call out dial-peer 2 as it has been explicitly told via the dial-peer group that is what it can do.



> **Note**: The destination-pattern on dial-peer 1, 2, and 3, is not the called number of 8675309. This is fine and the call still routes without an issue.

Remember, as discussed in the dial-peer states section, you need something/anything configured as an outbound matching statement. In this case, destination pattern is only to bring the dial-peer into an Up operational state, and the digit-string of that command is never evaluated. It is recommended to configure a pattern like destination-pattern AAAA as this is a valid destination-pattern. Since this is technically a valid dial-peer, other calls could match it. Thus the AAAA digit-string means that you can never use it for anything other than a specific scenario involving a dial-peer group as the likelihood of a call coming in for AAAA is very, very low.

```
!
dial-peer voice 1 voip
 description Server 1
 destination-pattern ^1234$
 session target ipv4:192.0.2.1
!
dial-peer voice 2 voip
 description Server 2
 destination-pattern ^5678$
 session target ipv4:192.0.2.2
!
dial-peer voice 3 voip
 description Server 3
 destination-pattern AAAA
 session target ipv4:192.0.2.3
!
voice class dpg 1
 description Dial-peer Group for specific called number 8675309
 dial-peer 2 preference 1
 dial-peer 3 preference 2
 dial-peer 1 preference 3
!
dial-peer voice 1234 voip
 description INCOMING dial-peer with DPG
 incoming called-number ^8675309$
 destination dpg 1
!
```

Verification

```
Gateway# show voice class dpg 1
Voice class dpg: 1       AdminStatus: Up
 Description: Dial-peer Group for specific called number 1234
 Total dial-peer entries: 3

 Peer Tag       Pref
 --------       ----
 2              1
 3              2
 1              3
-------------------------------------
```

E164-Pattern-Maps

This feature give administrators the ability to reduce the number of total dial-peers by combining many possible number matches (destination-patterns, incoming called-number, and so on) into a single pattern map. Outbound dial-peer e164-pattern-map support was added in Cisco IOS 15.2(4)M and Cisco IOS-XE 3.7S while inbound dial-peer e164-pattern-map support was added in Cisco IOS 15.4(1)T and Cisco IOS-XE 3.11S.

An e164-pattern-map can be configured via the the CLI or pre-configured and saves as a .cfg file. The .cfg file is then added to the flash of the gateway and then referenced when configuring the rest of the command. The .cfg file can utilize 5000 entries.

The entries in both configuration methods can utilize all normal dial-peer wildcards for further aggregation!

Note: When both incoming and outgoing dial peers are configured with **calling e164-pattern-map**, it is essential to apply a **dial-peer provision policy** to the incoming dial peer to avoid call failures.

Full Documentation: [Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards]

CLI Configuration Example - Calling Numbers

```
!
voice class e164-pattern-map 1
 description E164 Pattern Map for calling numbers
 e164 919574100.
 e164 919574300.
 e164 8675309
!
dial-peer voice 1 voip
 description INBOUND Dial-peer based on CALLING #
 incoming calling e164-pattern-map 1
!
dial-peer voice 11 voip
 description OUTBOUND Dial-peer based on CALLING #
 destination calling e164-pattern-map 1
!
```

CLI Configuration Example - Called Number

```
!
voice class e164-pattern-map 2
 description E164 Pattern Map for called 800 numbers
 e164 91800T
 e164 91855T
 e164 91888T
!
dial-peer voice 2 voip
 description INBOUND Dial-peer based on CALLED #
 incoming called e164-pattern-map 2
!
dial-peer voice 22 voip
 description OUTBOUND Dial-peer based on CALLED #
 destination e164-pattern-map 2
!
```

Flash Configuration Example

```
!
```

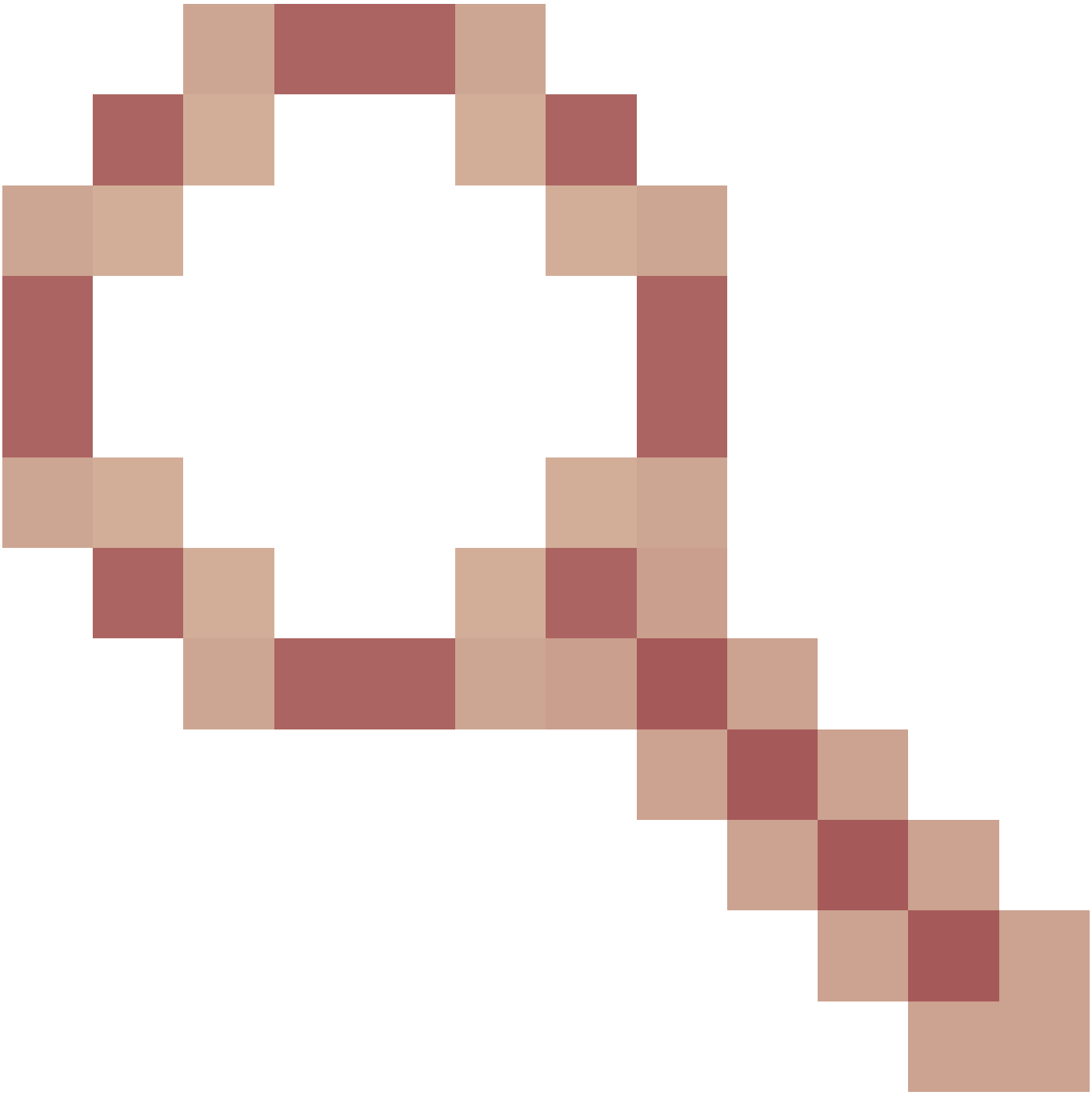
```
voice class e164-pattern-map <tag>
 description FILEPATH for E164 Pattern Map
 url flash:<filepath>/e164-pattern-list.cfg
!
dial-peer voice ### voip
 description E164 Pattern Map Dial-peer
 incoming calling e164-pattern-map <tag>
!

voice class e164-pattern-map load <tag>
```

Verification

```
Gateway# show voice class e164-pattern-map 1

e164-pattern-map 1
----------------------------------------
  Description: CUCM phones
  It has 3 entries
  It is not populated from a file.
  Map is valid.

E164 pattern
--------------------
8675309
1...
[2-5]...$
```

Notable Defects

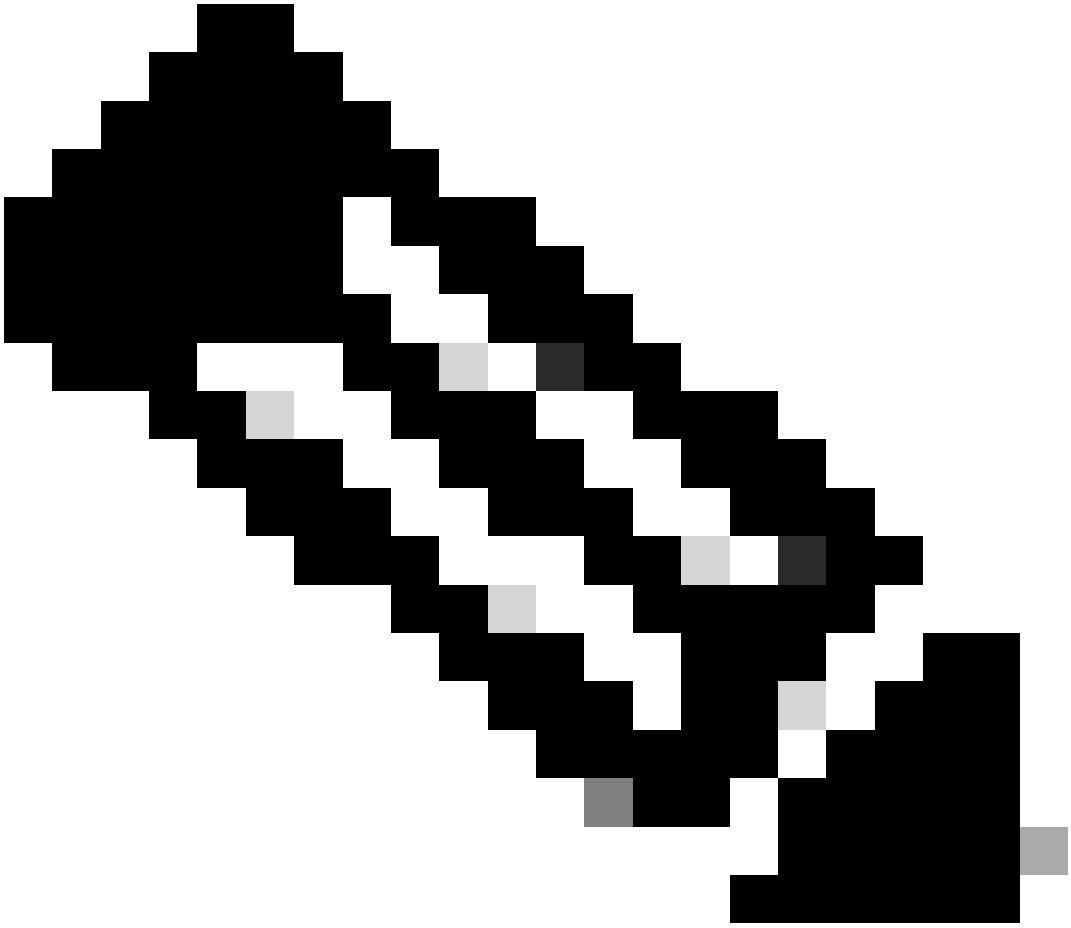
Cisco bug ID CSCva64393



e164-pattern-map does not parse the last line of the config file.

Destination Server-Groups

Server-groups give administrators the ability to configure multiple destinations (session-targets) on the same VOIP dial-peer. By default, the sort order is the preference defined in the server-group entries. Round-robin hunting can be employed when you use the command **hunt-scheme round-robin**. Server-Groups were added in Cisco IOS 15.4(1)T and Cisco IOS XE 3.11S. In Cisco IOS XE 17.4.1a configurable huntstop error codes was added to voice class-server group configurations. That is, you can configure a single error code, say 404 Not Found, and SIP error would normally trigger the device to try the next option in the server-group. With the config **huntstop 1 resp-code 404** in place within the the server-group; hunting can stop. These can also be configured for a range like: **huntstop 1 resp-code 401 to 599.**



Note: The maximum number of entries is 5 per server-group.

Full Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Configuration Example - Normal

```
!
voice class server-group 1
 hunt-scheme round-robin
 ipv4 10.50.244.2 port 5060 preference 1
 ipv4 10.50.244.62
 ipv6 2010:AB8:0:2::1 port 2323 preference 3
 ipv6 2010:AB8:0:2::2 port 2222
!
dial-peer voice 1 voip
 session protocol sipv2
 destination-pattern 8675309
 session server-group 1
!
```

Verification

```
Gateway# show voice class server-group 1
Voice class server-group: 1
 AdminStatus: Up OperStatus: Up
 Hunt-Scheme: round-robin Last returned server:
 Description:
 Total server entries: 4

 Pref Type IP Address IP Port
 ---- ---- ---------- -------
 1 ipv4 10.50.244.2     5060
 0 ipv4 10.50.244.62
 3 ipv6 2010:AB8:0:2::1 2323
 0 ipv6 2010:AB8:0:2::2 2222
[..truncated..]
```

Destination Server Group and OPTIONS Keepalive

Be aware that Server Groups do not follow normal Out-of-dialog OPTIONS Keepalive mechanisms. They utilize a feature called an option-keepalive profile. This allows the gateway to monitor each call agent defined in the specific server-group.

Option-keepalive Example with Server Group

```
!
voice class server-group 1
 hunt-scheme round-robin
 ipv4 10.50.244.2
 ipv4 10.50.244.62
!
dial-peer voice 1 voip
 session protocol sipv2
 session server-group 1
 voice-class sip options-keepalive profile 1
!
```

Verification

<#root>

```
Gateway#

show voice class sip-options-keepalive 1

Voice class sip-options-keepalive: 1             AdminStat: Up
 Description:
 Transport: system               Sip Profiles: 0
 Interval(seconds) Up: 5                 Down: 5
 Retry: 5
```

```
  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  1             1                               Active          87

  Server Group: 1               OOD Stat: Active
   OOD SessID   OOD Stat
   ----------   --------
   1            Active
   2            Active

 OOD SessID: 1                  OOD Stat: Active
  Target: ipv4:10.50.244.2
  Transport: system             Sip Profiles: 0

 OOD SessID: 2                  OOD Stat: Active
  Target: ipv4:10.50.244.62
  Transport: system             Sip Profiles: 0
```

Outbound Proxy

The SIP Outbound Proxy configuration can be added to voice service voip, voice class tenant, or dial-peer configurations to specify the destination for a Layer 3 SIP Packet.

That is, the session target on a dial-peer can be used to create the SIP Packet, but the outbound proxy can be where the packet is sent at Layer 3.

```
!
voice service voip
 sip
  outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
!
voice class tenant 100
 outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
!
dial-peer voice 100 voip
 session target ipv4:192.168.1.1
 voice-class sip outbound-proxy dns:la01.sipconnect-us10.cisco-bcld.com
!
```

It can be noted that the default configuration for a dial-peer is voice-class sip outbound-proxy systemwhich can cause a dial-peer to use global voice service voip > sip configuration.

This behavior can be disabled and force a dial-peer to fall back and use the session target as the layer 3 destination per dial-peer with this configuration:

```
dial-peer voice 777 voip
 no voice-class sip outbound-proxy
```

POTS Trunk Groups

Trunk Groups are a collection of physical voice-ports with similar signaling capabilities. This is a feature which can be employed to reduce the total number of POTS dial-peers that need to be configured. Trunk groups were introduced into Cisco IOS in 12.1(3)T and are present in all versions of Cisco IOS XE.

Full Documentation: Gateway Trunk and Carrier Based Routing Enhancements

Configuration Example

```
!
trunk group PSTN
 description PSTN voice-ports
!
trunk group FXO
 description FXO voice-ports
!
voice-port 0/2/0
 trunk-group PSTN 1
!
voice-port 0/2/1
 trunk-group PSTN 2
!
voice-port 0/2/2
 trunk-group FXO 1
!
voice-port 0/2/3
 trunk-group FXO 2
!
dial-peer voice 1234 pots
 trunkgroup PSTN 1
 trunkgroup FXO 2
!
```

Voice Class Tenants

Cisco IOS 15.6(2)T and Cisco IOS XE 16.3.1 introduced voice class tenants which allows each tenant to have their own individual configurations. A tenant can be an Telephony Provider, Cisco Unified Communication Manager (CUCM), or any other 3rd Party Call Agent an administrator would like have specific global settings for. First an administrator creates a voice class tenant and defines the parameters. The voice class tenant is then applied to the specific dial-peer or choice. This new configuration gives administrators another level of control over calls beyond dial-peers and global configuration.

With 17.8.1a, Voice Class Tenant configurations can be configured with a sip-listen command (coupled with the appropriate SIP control binding command) to define the non-secure or secure port that tenant. This means tenant 1 could listen for unsecure SIP on UDP 5060 + VRF Red while tenant 2 listens for SIP on TCP TLS 5070 + VRF Blue. After matching the tenant based on listen-port + bind + optional vrf inbound dial-peers are filtered to those that have the tenant applied.

Full Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Normal Order of Command Preference without Tenants

1. Dial-peer command
2. Global command (voice service voip and sip-ua)

Order of Command Preference with Tenants

1. Dial-peer command
2. Tenant command
3. Global command (voice service voip and sip-ua)

Multi-Tenant Configuration Example

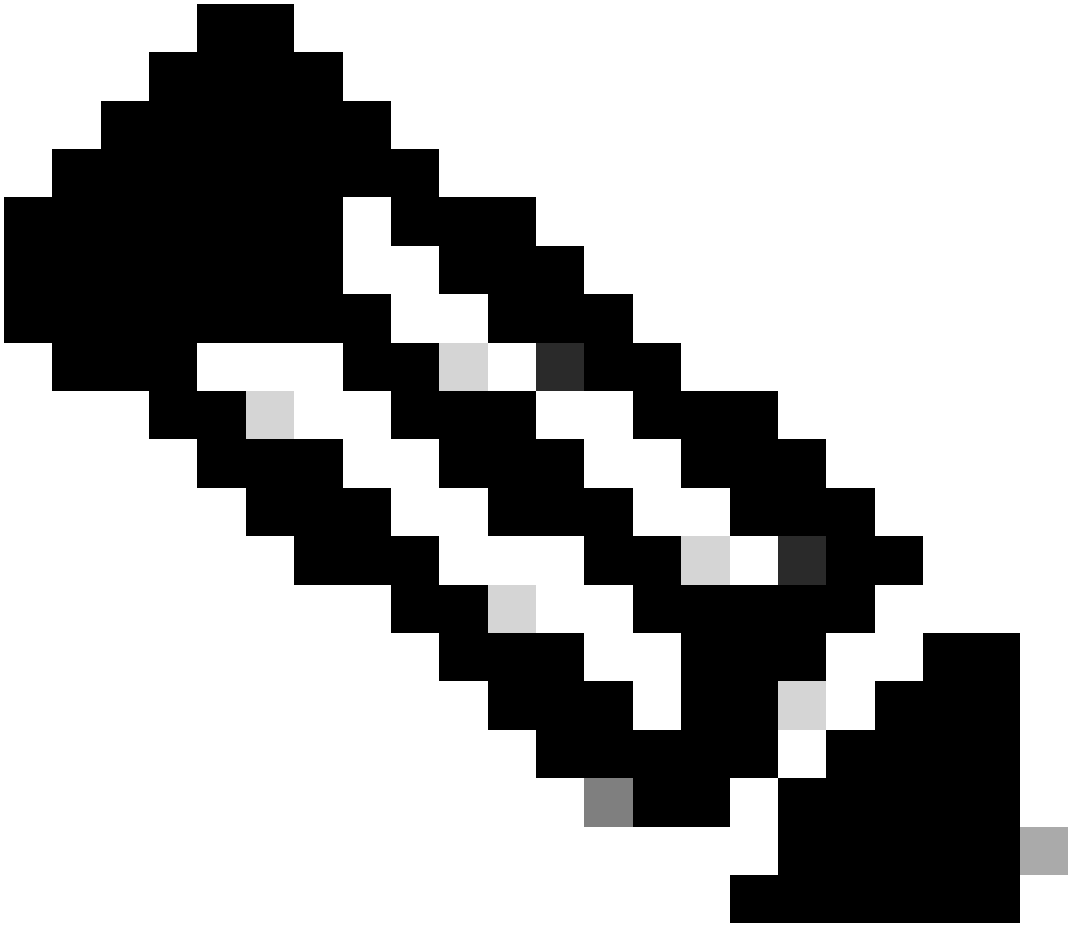
You have two tenants 777 and 999. You have configured them with slightly different configurations and applied them to the dial-peers. This means that calls using the different dial-peers have the dial-peer based configurations as well as the tenant specific configurations. The options listed are only a snippet of the power of voice class tenants. Refer to the documentation to see what can be configured on a tenant. It is recommend to employ strict matching mechanisms like voice class uri or tagging numbers with certain number strings to separate tenant dial-peer matching, or even configuring VRFs so that Tenant A never overlaps with Tenant B and accidentally matches a dial-peer they cannot.

```
!
voice class tenant 999
 asymmetric payload full
 bind control source-interface GigabitEthernet0/0/0.228
 bind media source-interface GigabitEthernet0/0/0.228
 g729 annexb-all
!
voice class tenant 777
 sip-server ipv4:192.168.1.2
 bind control source-interface Loopback0
 bind media source-interface Loopback0
 pass-thru content sdp
!
dial-peer voice 999 voip
 destination-pattern 8675309
 session protocol sipv2
 incoming called-number 8675309
 voice-class sip tenant 999
!
dial-peer voice 777 voip
 destination-pattern 8675309
 session protocol sipv2
 session target sip-server
 voice-class sip tenant 777
!
```

Verification

Currently, there are no individual commands to see voice class tenant configurations. This command can be sufficient for filtering the running config to just the tenant information.

```
show run | sec tenant
```



Note: Cisco bug ID CSCvf28730 is where show sip-ua register status does not reflect the status of SIP trunk registration on a voice class tenant.

ILS URI Calls CUBE (Voice Class Route-String)

Route Strings are used with CUCM Intercluster Lookup Service (ILS) and can be configured to allow Cisco Gateways to route VoIP calls via the route-string included in the SIP Invite received from a CUCM 9.5+ running the ILS service. This feature was added in Cisco IOS 15.3(3)M and Cisco IOS XE 3.10S. Most ILS connections are CUCM to CUCM and administrators do not bother involving a CUBE for intercluster trunking. However, if you need to perform the function with CUBE in the middle, the options are there. CUCM needs to have the setting **Send ILS Learned Destination Route String** enabled on the SIP Profile applied to the SIP Trunk in order to send the x-cisco-dest-route-string header to CUBE

Full Documentation: Enterprise Application Interoperability for H.323-to-SIP and SIP-to-SIP Configuration Guide, Cisco IOS Release 15M&T

Configuration Example CUCM - SIP - CUBE - SIP - CUCM

```
!
voice service voip
 sip
  call-route dest-route-string
!
voice class route-string rt1
 pattern london.uk.eu
!
voice class sip route-string rt2
 pattern *.eu
!
voice class sip-hdr-passthrulist hdr1
 passthru-hdr x-cisco-dest-route-string
!
dial-peer voice 1 voip
 description INBOUND dial-peer
 session protocol sipv2
 voice-class sip pass-thru headers hdr1
 incoming called-number .
!
dial-peer voice 2 voip
 description OUTBOUND dial-peer
 destination route-string rt2
 session protocol sipv2
 session target ipv4:172.16.104.178
!
```

Verification

```
show voice class route-string
```

Legacy Call Routing Techniques

The items covered in this section are considered legacy techniques. While the ability to configure these are still present within a Cisco Gateway it is not recommended to make use of these commands in modern configurations. This document only covers them because they could be encountered while working with legacy configurations or when performing upgrades.

DNIS-Map

DNIS-maps could be considered the precursor for what would now be an E164-pattern-map. DNIS maps were added to Cisco IOS in 12.2(2)XB and have always existed in Cisco IOS XE.

If there are DNIS-maps configured, it would be worth converting them to the more robust e164-pattern-map feature.

Command Syntax: Cisco IOS Voice Command Reference - D through I

Configuration Example

```
!
voice dnis-map 34
 dnis 8675309
!
dial-peer voice 88 voip
 dnis-map 34
!
```

Trunk Group Labels

Trunk Group Labels were added in Cisco IOS 12.2(11)T and exist in all Cisco IOS XE Versions. The purpose of a trunk-group-label is similar to a Carrier-ID in the sense that it can be used to augment the matching of dial-peers. This is available for configuration within POTS trunk groups, VOIP and POTS dial-peers as well as voice source groups. The use of Trunk Group Labels are rarely seen in modern Cisco Gateway configurations.

Command Syntax: Cisco IOS Voice Command Reference - T through Z

Configuration Example

```
!
dial-peer voice 112 pots
 trunk-group-label source north3
 trunk-group-label target east17
!
```

Numbering Type

With ISDN Q.931 integrations there exists the ability to match a dial-peer based on the calling or called number as well as the specific ITU number type from the Q.931 SETUP messaging. This is configurable via the **numbering-type** command on a VOIP or POTS dial-peer. Numbering-type cannot be used alone and must be used in conjunction with destination-pattern, answer-address, or incoming called-number. This means both the condition of the inbound / outbound matching statement and the number type must match to be a success for the dial-peer to be considered for inbound and outbound call routing.

Numbering-match can be thought of as a dial-peer filtering mechanism rather than a matching mechanism. This is because a dial-peer with and without a numbering type command applied are considered the same default preference weight if no administrator preference is applied. This is unlike carrier-id which, when applied to a dial-peer along side other matching mechanism, adds preference to that dial-peer over others if both conditions are true.

Numbering Type matching was added in Cisco IOS 12.0(7)XR1 and is present in all Cisco IOS XE releases. With the decline of traditional POTS ISDN lines being deployed in collaboration networks the use of numbering-type is rarely seen in modern deployments.

Command Syntax: Cisco IOS Voice Command Reference - K through R

Configuration Example

This dial-peer can match 4085150000 through 4085159999 only if the ISDN number type is National.

```
!
dial-peer voice 408 voip
 numbering-type national
 destination-pattern 408515....
 session target ipv4:10.1.1.2
!
```

Possible Number Types:

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

Dial-Peer Data

Data Dial-peers were introduced in Cisco IOS 12.2(13)T and the use of such dial-peers was for incoming data modem calls on a Cisco Gateway. This dial-peer is only for use in the inbound direction and rarely seen in modern deployments.

Command Syntax: Cisco IOS Voice Command Reference - D through I

Configuration Example

```
!
dial-peer data 100 pots
 incoming called-number 100
!
```

Voice Source-Group

This feature was added in 15.1(2)T but is not implemented in many modern deployments. Other Security methods for Cisco IOS/CUBE are usually deployed.

The CUBE Application Security overview can be seen in this whitepaper starting at section 4.2.

Cisco Unified Border Element (CUBE) Management and Manageability Specification

Command Syntax: Voice Source-Group Feature

Dial-Peer Permissions

This configuration allows an administrator to restrict a dial-peer to either allow inbound connections only (term / terminate) or egress connections (orig / originate). This would be like explicitly configuring an inbound dial-peer to only be used for inbound calls and an outbound dial-peer for outbound calls. The default for any dial-peer is to allow both inbound and outbound connections. This CLI is not often deployed in modern deployments.

```
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# permission ?
 both allow both orig/term on this dialpeer
 none no orig/term allowed on this dialpeer
 orig allow only orig on this dialpeer
 term allow only term on this dialpeer
```

URI and Digit Manipulation

At some point in a collaboration deployment an administrator can need to manipulate digits or a URI / SIP Header. Cisco Gateways have numerous methods for digit manipulation which allows an administrator complete control over how and when a digit can be manipulated. However, this is not always easy and some people become overwhelmed with the different options or the administrator does not know an option exists.

Digit Manipulation via POTS Dial-Peers

POTS dial-peers have a few unique digit manipulations techniques unique to them that VOIP dial-peers do not have.

The first is the stripping of explicitly defined left-justified digits in a destination-pattern. This can be disabled by using the command **no digit-strip** on the POTS dial-peer.

Example:

In this example, 9011T is defined as the string for the destination-pattern.

With this in place, you can receive a call for 90113227045555. This matches the dial-peer for outbound call routing, and the explicitly defined digits of 9011 are stripped off before the call is routed out the voice-port.

```
!
dial-peer voice 1 pots
 destination-pattern 9011T
 port 0/0/0:23
!
```

This example shows a configuration with no digit-strip in place.

If the same number is called, the 9011 is sent .

```
!
dial-peer voice 1 pots
 destination-pattern 9011T
 port 0/0/0:23
 no digit-strip
!
```

The second is the ability to specify how many digits you would like to forward on the POTS dial-peer.

Take this example where you receive a call for 918005532447 from CUCM. In this situation, you want to remove the 9, but send the rest of the number starting with the 1.

If you configure the **forward-digits** command on the POTS dial-peer, you can specify exactly how many digits you send.

```
!
dial-peer voice 1 pots
 destination-pattern 918005532447
 forward-digits 11
 port 0/2/0
!
```

Lastly, POTS dial-peers can make use of the prefix command to add digits to a call before routing out the voice-port. This example strips off the explicitly defined 91 and prefix 007 to the number before sending the call out the voice-port.

```
!
dial-peer voice 1 pots
 destination-pattern 91T
 prefix 007
 port 0/1/0:15
!
```

Digit Manipulation via Voice Translation Rules and Profiles

Voice translation-rules are regular expressions (regex) used to transform digits. Translation-rules and

Profiles were added to Cisco IOS in 12.0(7)XR1. A translation-rule is applied to voice translation-profiles which are then applied to a dial-peer or voice-port. Translation-rules contain a match input and a modification output. Along with the match input on the number there is a match and modify input for the ISDN plan and type. The combination of match number string, plan, and type is considered a match. This means all match inputs defined must be true for the translation to take place.

Translation-rules have the ability to change Called, Calling, redirect-called, redirect-target, and callback-number in ISDN, SIP, and H323 signaling protocols. Translation-rules match based on a top-down search, so order of the rules is of utmost importance. If a match is found in a higher rule, the gateway immediately stops searching and processes the translation. Translation rules cannot change non-numeric sip headers such as testuser@10.10.10.10. For this manipulation, utilize a SIP profile.

Transition-rules can be used to block calls on Cisco Gateways.

Translation-Profile Selection Preference

1. Incoming voice translation-profile on the voice-port
2. Incoming voice translation-profile on Trunk Group applied to Serial Interface
3. Incoming voice translation-profile on the inbound dial-peer
4. Incoming voice translation-profile defined via voice service pots
5. Incoming voice translation-profile defined via global 'voip-incoming translation-profile'
6. Outbound voice translation-profile defined via voice service pots
7. Outbound voice translation-profile or translate-outgoing on the outbound dial-peer
8. Outbound voice translation-profile on Trunk Group applied to Serial Interface
9. Outbound voice translation-profile on the voice-port

In addition to dial-peer regex and wilcards translation-rules have their own regex characters.

Character	**Definition**
*	When used in translation-rules this is regex for 0 or more of the previous character. To match a literal * use an escape character: *
\	Commonly used to escape sets in translation rule \(\)
&	Ampersand is used to bring over anything matched in the initial match set for the modification set
( )	Items wrapped in parenthesis are considered a set.
^	Defines the explicit start of a string. Unlike dial-peers translation-rules do not define the start of the string. This means defining a string without a ^ can possible match anything in the input string which can lead to unwanted translations in the middle of a number.

Modification Sets

- Sets are specified as \0, \1, \2, and so on.
- \0 matches anything in between the first match-set. This can also be accomplished via an ampersand character: &.
- \1 matches the first set of ( ) in the match-set
- \2 matches the second set of ( ) in the match-set
- So on and so forth.

Translation-rule example with two sets

In this example, you can examine the number 000111000222.

You want to remove the 0s from the number and realize a final number of 111222.

To do this, you configure set 1 and 2 to grab the 111 and 222 respectively while dropping the 0s.

```
!
voice translation-rule 333
 rule 1 /000\(111\)000\(222\)/ /\1\2/
!
voice translation-profile SET-EXAMPLE
 translate called 333
!

Gateway# test voice translation-rule 333 000111000222
Matched with rule 1
Original number: 000111000222   Translated number: 111222
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none
```

Example to strip the 9 out-dial pattern from a called number

```
!
voice translation-rule 9
 rule 1 /^9\(.*\)/ /\1/
!
voice translation-profile STRIP-9
 translate called 9
!
dial-peer voice 9 voip
 translation-profile outgoing STRIP-9
!
voice-port 0/0/0
 translation-profile outgoing STRIP-9
!

Gateway# test voice translation-rule 9 918675309
Matched with rule 1
Original number: 918675309      Translated number: 18675309
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none
```

Truncate Called Number to 4 Digits

```
!
voice translation-rule 4
 rule 1 /.*\(....\)/ /\1/
!
voice translation-profile STRIP-TO-4
 translate called 4
!

Gateway# test voice translation-rule 4 8675309
Matched with rule 1
Original number: 8675309        Translated number: 5309
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none
```

Stripping Plus + From the Called Number

```
!
voice translation-rule 999
 rule 1 /\+\(.*\)/ /\1/
!
voice translation-profile STRIP-PLUS
 translate called 999
!
Gateway# test voice translation-rule 999 +8675309
Matched with rule 1
Original number: +8675309       Translated number: 8675309
Original number type: none      Translated number type: none
Original number plan: none      Translated number plan: none
```

Translation Rules can also be applied directly to a dial-peer without first being applied to a translation-profile.

```
!
voice translation-rule 1
 rule 1 /1234/ /8678309/
!
voice translation-rule 2
 rule 2 /^4...$/ /1408515\0/
!
dial-peer voice 1 voip
 translate-outgoing called 1
!
dial-peer voice 2 voip
 translate-outgoing calling 2
!
```

Translation Profile on Trunk Group

```
!
trunk group <name>
 translation-profile incoming <profile-name>
 translation-profile outgoing <profile-name>
!
```

Debug Voice Translation Rules and Profiles

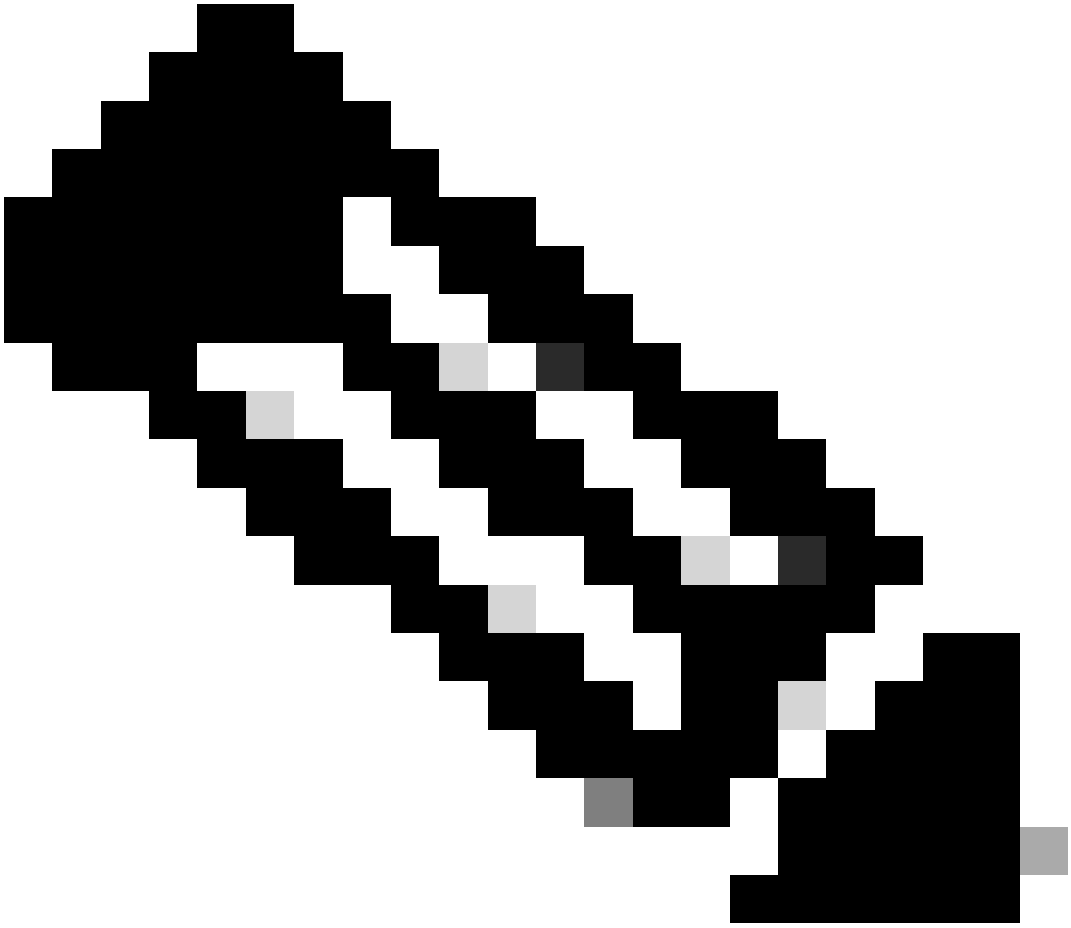
```
debug voip ccapi inout
debug voice translation
debug dialpeer
test voice translation-rule <number> <string> type <type> plan <plan>
```

Voice Class e164-translation

The voice class e164-translation feature is a newer Cisco IOS XE feature which allows an administrator to create a list of match statements and modify statements to be loaded via a configuration file from the flash or a network directory. This is similar to the concept for the e164-pattern-map feature discussed in this document. This allows an administrator to configure up to 100 translations inside a configuration file and apply them in a single translation profile. For more information, reference the Cisco IOS Voice Command Reference

Follow this syntax for the .cfg file:

```
pattern1_to_be_matched<tab>replaced_pattern<space><enter>
pattern1_to_be_matched<tab>replaced_pattern<space><enter>
```



Note: The trailing space is very important, and the import can fail without that extra formatting step.

Sample.cfg

```
+111111     8897
+222222     8312
928747     +123456789
737362     +987654321
```

This file then references as such:

```
voice class e164-translation 164
 url ftp://username:password@10.10.10.10/sample.cfg
```

Now you apply to a translation-profile normally, and then from there apply to dial-peers using normal translation profile syntax.

```
voice translation-profile e164
 translate calling voice-class e164-translation 164
 translate called voice-class e164-translation 164
```

The command **show voice class e164-translation e164-translation-number** can be used to view the contents of the translation profile.

Digit Manipulation via ISDN Maps

ISDN MAPS are an older technique for modifying digits. This was added in Cisco IOS 12.0(6)T and most new configurations do not utilize this feature as they are not as robust as voice translation-rules and profiles. ISDN Maps are defined under the Serial interface.

Configuration Example

```
Serial0/0/0:23
 isdn map address ^911 plan isdn type unknown
 isdn map address ^1.......... plan isdn type national
 isdn map address ^2......... plan isdn type national
 isdn map address ^3......... plan isdn type national
 isdn map address ^4......... plan isdn type national
 isdn map address ^5......... plan isdn type national
 isdn map address ^6......... plan isdn type national
 isdn map address ^7......... plan isdn type national
 isdn map address ^8......... plan isdn type national
 isdn map address ^9......... plan isdn type national
```

Digit Manipulation via Number Expansion (num-exp)

Like ISDN Maps, Number Expansion is an older technique added in Cisco IOS 11.3(1)T and not used much in new networks. This feature was added before voice translation-rules and profiles existed. Number Expansion is a global change of digits applied to all dial-peers on a Cisco Gateway. The modification is applied to the called number after the dial-peer has been matched, and right before the call is sent to the next call-agent.

Configuration Example

```
num-exp 4... 18005554...
num-exp 1234 8675309
```

Inbound / Outbound SIP Profiles

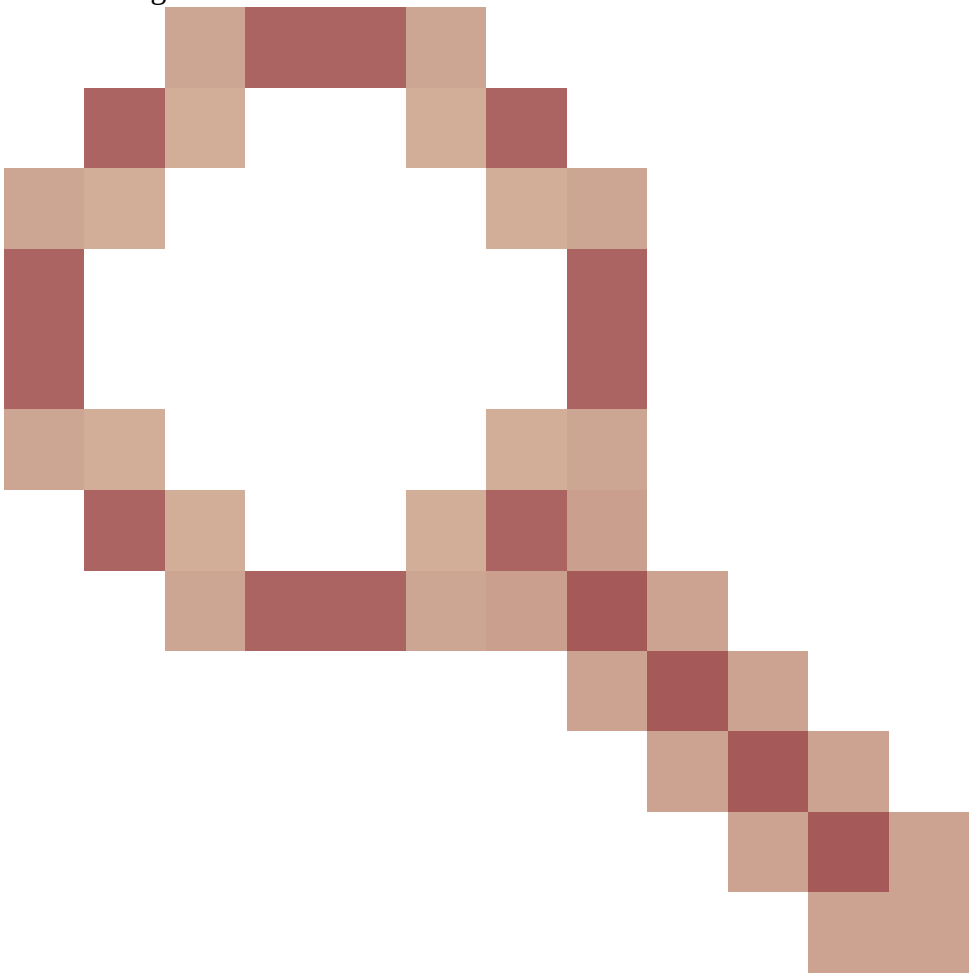
SIP profiles are robust Regular Expression (regex) Match statements which allow an administrator to change any aspect of a SIP message which includes SDP and SIP headers. These can be enabled globally, per dial-peer or per tenant. SIP Profiles are available for inbound modifications starting with with Cisco IOS 15.4(2)T and Cisco IOS XE 3.12S . Since SIP profiles are so robust, this document only covers a few specific examples. SIP profiles also add the ability for custom SIP headers to be modified or added in Cisco IOS 15.5(2)T and Cisco IOS-XE 3.13S.

Key Points about inbound versus outbound SIP Profiles

- Inbound SIP Profiles change the message BEFORE the CUBE processes the message for call routing.
- Outbound SIP Profiles change the message AFTER the CUBE has processed the call routing and before the message is sent to the next hop.

Other notes about sip-profile Configuration:

- SIP Profiles cannot manipulate m=image SDP attributes. The enhancement for this was filed under



  Cisco bug ID CSCsr20474
  Additionally, SIP profiles cannot remove or add values to SDP. Only you can modify these values. However, it is possible to modify an SDP value into a null value by specifying the entire value then setting the output to a set of empty quotes without a space.
- There are no checks performed to see if the current command being entered already exists or a version of that command already exists when entering commands within voice class sip-profile. If an administrator pastes a line 7 times into a sip-profile, it displays 7 times in the running configuration. It is advised to remove the command being modified and then enter the new command when editing sip-profiles to avoid multiple commands being present.

Full Documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

SIP Profile Testing Tool on CSA: SPT Tool

Inbound/Outbound SIP Profile Example Syntax

```
!
voice class sip-profiles <number>
 request <message-type> sip-header <header> modify "match-pattern" "replace-pattern"
 request <message-type> sip-header <header> add "add-pattern"
 request <message-type> sip-header <header> remove

 request <message-type> sdp-header <header> modify "match-pattern" "replace-pattern"
 request <message-type> sdp-header <header> add "add-pattern"
 request <message-type> sdp-header <header> remove

 response <number> sip-header <header> modify "match-pattern" "replace-pattern"
 response <number> sip-header <header> add "add-pattern"
 response <number> sip-header <header> remove

 response <number> sdp-header <header> modify "match-pattern" "replace-pattern"
 response <number> sdp-header <header> add "add-pattern"
 response <number> sdp-header <header> remove
!
```

Inbound/Outbound SIP Profile Example with Numbers

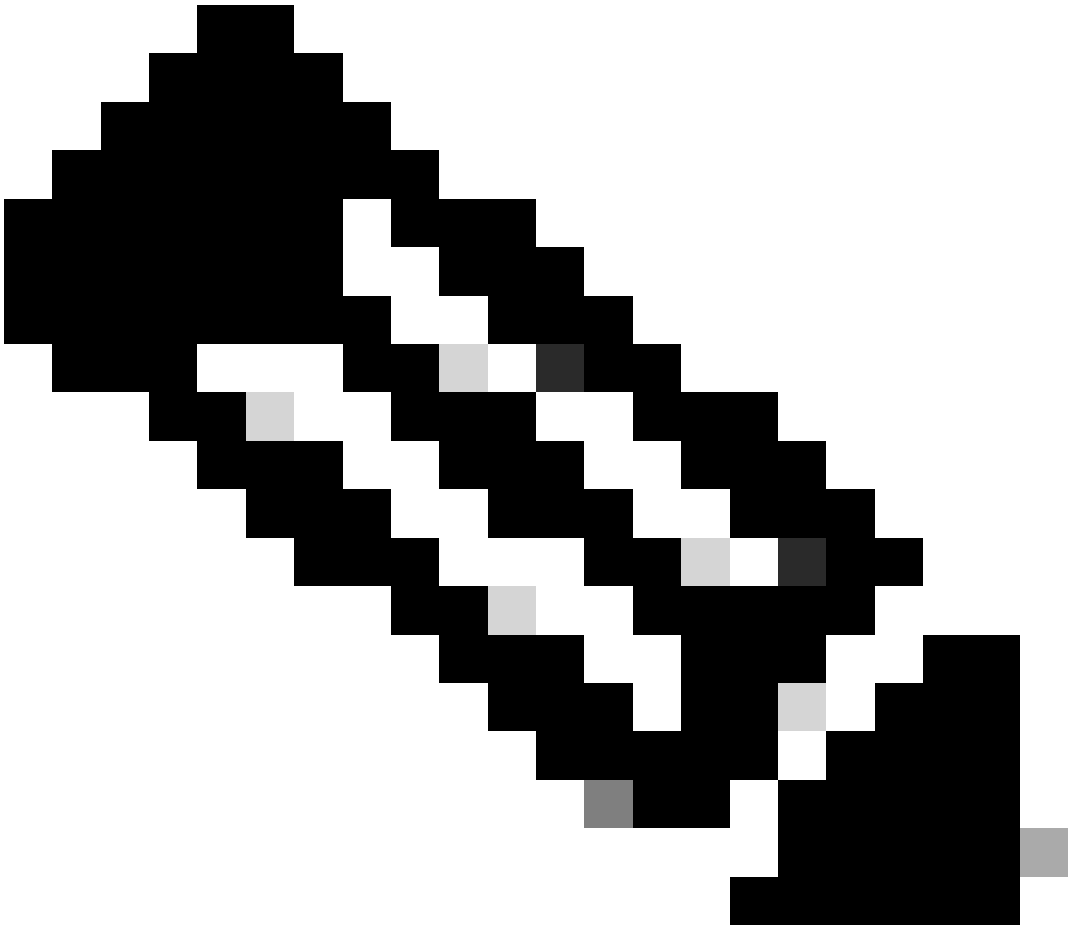
```
voice class sip-profiles 200
 rule 1 response ANY sip-header Remote-Party-ID modify "match-pattern" "replace-pattern"
 rule 2 response ANY sdp-header Audio-Attribute modify "match-pattern" "replace-pattern"
```

Outbound SIP Profile Application Methods

<#root>

```
! Global Application

voice service voip
 sip
  sip-profiles <number>
!

! Tenant Application


voice class tenant <tag>
 sip-profiles <tag>
!

! Dial-peer Application


dial-peer voice <tag> voip
 voice-class sip profiles <number>
!
```

Inbound SIP Profile Application Methods



Note: It is required to enable sip-profile inbound under voice service voip sip whether the global application, tenant, or dial-peer application is used.

<#root>

```
! Global Application

voice service voip
 sip
  sip-profiles inbound
  sip-profiles <number> inbound
!
```

<#root>

```
! Tenant Application


voice service voip
```

```
 sip
  sip-profiles inbound
!
voice class tenant <tag>
 sip-profiles <tag> inbound
!
```

<#root>

```
! Dial-Peer Application


voice service voip
 sip
  sip-profiles inbound
!
dial-peer voice <tag> voip
 voice-class sip profiles <number> inbound
!
```

Example SIP Profile to modify OPTIONS keepalive messages.

```
!
voice class sip-options-keepalive 200
 transport tcp tls
 sip-profiles 299
!
```

Example SIP Profile to modify Host, Domain, or both portions of a URI.

<#root>

```
! Host

!
voice class sip-profiles 1
 request ANY sip-header Contact modify "sip:(.*)@" "sip:8675309@"
!

! Domain

!
voice class sip-profiles 2
 request ANY sip-header Contact modify "10.67.138.241:5060" "cisco.com"
!

! Note: Port is optional

!

! Modify Both User and Host

!
voice class sip-profiles 3
 request ANY sip-header Contact modify "sip:(.*)>" "sip:8675309@cisco.com>"
```

```
!
```

Example SIP Profile to add, modify, or remove Diversion headers.

<#root>

```
! Add

!
voice class sip-profiles 777
 request INVITE sip-header Diversion add "Diversion: <sip:1234@cisco.com>"
!
!

! Modify

!
voice class sip-profiles 888
 request INVITE sip-header Diversion modify "sip:(.*)>" "sip:1234@cisco.com>"
!
!

! Remove

!
voice class sip-profiles 999
 request INVITE sip-header Diversion remove
!
```

Example SIP Profile to modify Caller ID Name portion of SIP headers.

```
!
voice class sip-profiles 123
 request INVITE sip-header From modify "\".*\"" "\"TEST CLID*\""
!
```

Example SIP Profile to change a 183 Session In Progress to a 180 Ringing.

```
!
voice class sip-profiles 789
 response 183 sip-header SIP-StatusLine modify "SIP/2.0 183 Session in Progress" "SIP/2.0 180 Ringing"
!
```

Example SIP Profile for one-way or no-way audio interoperability with a provider.

<#root>

```
!
voice class sip-profiles 200
```

```
 request ANY sdp-header Audio-Attribute modify "a=inactive" "a=sendrecv"
 request ANY sdp-header Audio-Connection-Info modify "0.0.0.0" "10.10.10.10"
!

! where 10.10.10.10 is CUBE's provider facing IP address
```

Example SIP profile to remove UPDATE method for interoperability issues.

```
!
voice class sip-profiles 200
 request ANY sip-header Allow-Header modify ", UPDATE" ""
!
```

Example SIP Profile showing SET use within SIP profile. This is the same concept of Sets described within the voice translation-rule section.

```
!
voice class sip-profiles 1
 request ANY sip-header Contact modify "sip:(.*)@" "sip:\1@"
!
```

Configuration IF logic and newline breaks with a SIP profile.

Newline breaks are supported in SIP profiles, however, there is only one very specific use case for these. Since SIP profiles do not have any If, Then, Else logic, there is now a way to perform modifications to one header based on an input from another header. For example, an administrator only wants to modify a diversion header if the FROM header contains 1234@cisco.com. Utilizing the newline break we can spoof the IF statement within a SIP profile. See the example configuration: You match 1234 at any domain in the From header. Then you bring over the first set and add a new line break \x0D\x0AD. Finally, you add the header you want. See that this method only allows you to ADD a header. There is no way to modify another header. So this only partially meets the requirements an administrator wanted to achieve previously.

```
!
voice class sip-profiles 1
 request INVITE sip-header From modify “(.*sip:1234@.*)” “\1\x0D\x0ADiversion: <sip:5678@example.com>”
!
```

Example of SIP profile with OR logic.

```
!
voice class sip-profiles 123
 request ANY sdp-header Audio-Attribute modify "(a=sendonly|a=recvonly|a=inactive)" "a=sendrecv"
 response ANY sdp-header Audio-Attribute modify "(a=sendonly|a=recvonly|a=inactive)" "a=sendrecv"
```

```
!
```

Example of Layer 7 SIP Inspection via SIP-Profile.

<#root>

```
### Usage

10.21.15.10 replace with private IP of CUBE
a.b.c.d replace with public IP
-----------------------------------------------------


### Inbound from ITSP Layer 7 Fixup

!
voice class sip-profiles 888
 request INVITE sip-header SIP-Req-URI modify "@.*;" "@10.21.15.100;"
!
voice service voip
 sip
  sip-profiles inbound
!


### Outbound Layer 7 Fixup

!
voice class sip-profiles 777
 request ANY sip-header Contact modify "<sip:(.*)@10.21.15.100:5060>" "<sip:\1 a.b.c.d:5060>"
 response ANY sip-header Contact modify "<sip:(.*)@10.21.15.100:5060>" "<sip:\1 a.b.c.d:5060>"
 request ANY sip-header Via modify "SIP(.*) 10.21.15.100(.*)" "SIP\1 a.b.c.d\2"
 request ANY sdp-header Session-Owner modify "(.*IP4 ).*" "\1a.b.c.d"
 request ANY sdp-header Connection-Info modify "IN IP4 10.21.15.100" "IN IP4 a.b.c.d"
 request ANY sdp-header Audio-Connection-Info modify "IN IP4 10.21.15.100" "IN IP4 a.b.c.d"
 response ANY sdp-header Session-Owner modify "(.*IP4 ).*" "\1a.b.c.d"
 response ANY sdp-header Audio-Connection-Info modify "IN IP4 10.21.15.100" "IN IP4 a.b.c.d"
 response ANY sdp-header Connection-Info modify "IN IP4 10.21.15.100" "IN IP4 a.b.c.d"
 request ANY sip-header Remote-Party-ID modify "<sip:(.*)@10.21.15.100>" "<sip:\1 a.b.c.d>"
 response ANY sip-header Remote-Party-ID modify "<sip:(.*)@10.21.15.100>" "<sip:\1 a.b.c.d>"
!


### Apply to dial-peers for the side of the CUBE facing the ITSP


!
dial-peer voice 1 voip
 voice-class sip profiles 777
 voice-class sip profile 888 inbound
!
dial-peer voice 2 voip
 voice-class sip profiles 777
 voice-class sip profile 888 inbound
!
```

SIP Copylist

SIP Copylists are an extension of SIP Profiles which allows the gateway to COPY a header from the in-leg of a call and then PASTE to another spot in the egress SIP message on the out-leg. SIP Copylist support was added in Cisco IOS 15.1(3)T and Cisco IOS XE 3.6S. This is a very powerful way of creating dynamic headers based on other headers from the in-leg of the call.

The most common use-case is dynamically copying a FROM header to a different header like diversion or p-asserted-id so that the value of the user portion is the from user. This is mostly done for authentication as well as caller ID purposes.

Full Documentation: [Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards]

SIP Copylist Example

<#root>

```
!
! Create Copylist to copy the FROM header on the inbound message specified later.
!
voice class sip-copylist <number>
 sip-header From
!
! Apply this to the inbound dial-peer of the call.
!
dial-peer voice <tag> voip
 voice-class sip copy-list <number>
!
! Create SIP Profile to take From (peer-header) stored as variable "u01" and apply to a header of choice
! This example modifies the user portion of the Contact by copying the user portion of the From header t
!
voice class sip-profiles <number>
 request INVITE peer-header sip From copy "<sip:(.*)@" u01
 request INVITE sip-header Contact modify "<sip:(.*)>" "<sip:\u01@10.50.244.2>"
!
! Apply the SIP profile to an outbound dial-peer
!
dial-peer voice <tag> voip
 voice-class sip profiles <number>
!
```

Debugging SIP Profiles and Copylist

```
debug voip ccapi inout
debug ccsip mess
debug ccsip info
debug ccsip feature sip-profile
```

Debug Output From the Example SIP Copylist

```
### Ingress from CUCM
Received:
INVITE sip:1001@10.50.228.61:5060 SIP/2.0
Via: SIP/2.0/TCP 10.50.244.3:5060;branch=z9hG4bKaad21bc3ae7e
From: "5001" <sip:5001@10.50.244.3>;tag=100442~cdffff43-5020-4e79-a10b-99d406971010-36470319
Contact: <sip:5001@10.50.244.3:5060;transport=tcp>

### Copylist Details
00440: Mar  8 18:59:49.796: //-1/xxxxxxxxxxx/SIP/Info/info/64/sip_profiles_application_peer_copy_patte
000441: Mar  8 18:59:49.797: //187/D6138E000000/SIP/Info/info/64/sip_profiles_application_peer_copy_pat
000442: Mar  8 18:59:49.797: //-1/xxxxxxxxxxx/SIP/Info/info/64/sip_profiles_prefix_slash_in_copy_var_v
000443: Mar  8 18:59:49.797: //187/D6138E000000/SIP/Info/info/64/sip_profiles_application_peer_copy_pat
000444: Mar  8 18:59:49.797: //-1/xxxxxxxxxxx/SIP/Info/info/64/sip_profiles_application_modify_remove_
000445: Mar  8 18:59:49.797: //187/D6138E000000/SIP/Info/info/64/sip_profiles_check_and_get_variables_i
000446: Mar  8 18:59:49.797: //187/D6138E000000/SIP/Info/info/64/sip_profiles_check_and_get_variables_i
000448: Mar  8 18:59:49.797: //187/D6138E000000/SIP/Info/info/64/sip_profiles_check_and_get_variables_i
000449: Mar  8 18:59:49.797: //-1/xxxxxxxxxxx/SIP/Info/info/64/sip_profiles_app_modify_header: Passing
000450: Mar  8 18:59:49.798: //-1/xxxxxxxxxxx/SIP/Info/info/64/sip_profiles_application_modify_remove_
000451: Mar  8 18:59:49.798: //187/D6138E000000/SIP/Msg/ccsipDisplayMsg:

### Egress from CUBE
Sent:
INVITE sip:1001@14.50.228.63:5060 SIP/2.0
Via: SIP/2.0/UDP 10.50.228.61:5060;branch=z9hG4bK3C7CD
Remote-Party-ID: "5001" <sip:5001@10.50.228.61>;party=calling;screen=yes;privacy=off
From: "5001" <sip:5001@10.50.228.61>;tag=34C458-D6
Contact: <sip:5001@168.117.64.94>
```

Special Notes

Protocol Signaling and Media Binding

All Signaling protocols allow administrators the ability to bind the signaling to a specific interface. By default, a gateway without a static defined binding, then the gateway sources the signaling for a call from the physical interface the packet traverses. With binding on a dial-peer, the packet features source headers, messaging, and packets from the specified interface, but the actual packet still routes over the physical interface. Dial-peer binding always supersedes voice class tenant and global voice service voip binding with Session Initiation Protocol (SIP).

Many times administrators bind signaling to a loopback. This being a logical interface means no packets traverse this interface. In order to perform packet captures, the capture must be performed on a physical interface. The command **show ip cef <remote-ip>** displays the physical interface a packet utilize to route to the destination / remote IP even if the configuration is bound to a virtual interface.

Media and signaling binding do not always need to be the same IP. If an administrator needs to bind to a specific interface for signaling to / from a CUCM but the audio / media between the phone and the gateway can need to bind to another interface.

Configuration Example

This example shows a dial-peer bound to loopback 1 and it receives a call from CUCM.

Even though the media and signaling (control) are bound to loopback 1 the **show ip cef** command shows

that any packets sent to CUCM leave on the physical interface GigabitEthernet0/0/1.

```
!
dial-peer voice 2 voip
 description "Incoming call from CUCM"
 session protocol sipv2
 incoming called-number .
 voice-class sip bind control source-interface Loopback1
 voice-class sip bind media source-interface Loopback1
!
```

Order of Operations for Layer 7 Application Binding

1. According to the bind statement on the matched inbound/outbound dial-peer.
2. According to the binds under the voice class tenant assigned to the matched inbound/outbound dial-peer.
3. According to global binding statement.
4. According to the physical layer 3 interface the packet is expected to exit on based on the routing table.

SIP Binding Commands

<#root>

```
! Per Dial-peer
```

```
!
dial-peer voice 1 voip
 voice-class sip bind control source-interface <interface>
 voice-class sip bind media source-interface <interface>
!
```

```
! Global Binding
```

```
!
voice service voip
 sip
   bind control source-interface <interface>
   bind media source-interface <interface>
!
```

MGCP Binding Commands

```
!
mgcp bind control source-interface <interface>
mgcp bind media source-interface <interface>
!
```

SCCP Binding Commands

```
!
sccp local <interface>
!
sccp ccm group <number>
 bind interface <interface>
!
```

H323 Binding Commands

<#root>

```
!
inteface <interface>
 !
 ! Media Bind Command:

 h323-gateway voip interface
 !
 ! Signaling Bind Command:

 h323-gateway voip bind srcaddr <a.b.c.d>
!
```

DNS and VoIP Dial-Peers

DNS with VOIP is employed just like any other DNS solution. A common configuration is to utilize session target dns:FQDN.com.

A Cisco Gateway performs a DNS Resolution even when no ip domain lookup is configured globally on the gateway. This means that even though you are disabling DNS the VOIP dial-peers still resolve the DNS entry. However, recently in Cisco IOS XE 3.16S there were some changes to the overall DNS functionality within Cisco IOS XE platforms.

After this change, dial-peers configured with session target dns:FQDN.com now obey the fact that DNS is disabled with no ip domain lookup.
I recommend always ensuring the command "ip domain lookup" is configured when working with DNS to avoid this issue.

For outbound SIP connections, CUBE performs this order of operations for DNS resolution.

1. SRV query lookup
2. A Record Lookup
3. AAAA Record Lookup

For information on how the SRV is created, or how to skip the SRV and perform an A record query on a session-target, refer to the full documentation: Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

For inbound SIP connections where an Cisco IOS gateway needs to resolve a header to respond to a message, the gateway can use this order of operations for DNS Resolution

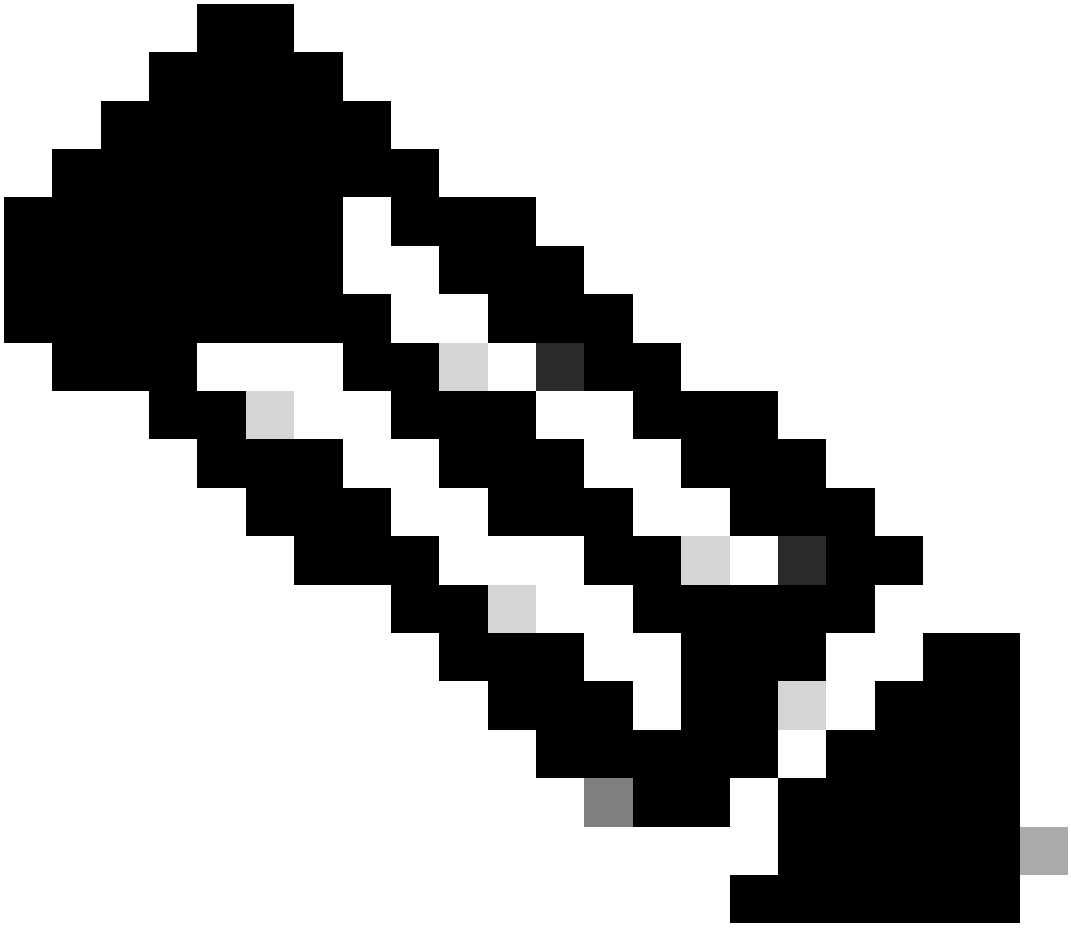
1. A Record Lookup
2. AAAA Record Lookup

In Cisco IOS XE 17.9.1, CUBE can check reachability of DNS session targets by way of options keepalive mechanisms. See full documentation:

Cisco Unified Border Element Configuration Guide - Cisco IOS XE 17.6 Onwards

Cisco IOS DNS Configuration Samples

```
ip host _sip._udp.cucmgroup.lab.local srv 1 50 5060 cucm1.lab.local
ip host _sip._udp.cucmgroup.lab.local srv 1 50 5060 cucm2.lab.local
ip host _sip._udp.cucmgroup.lab.local srv 1 50 5060 cucm3.lab.local

ip host cucm1.lab.local 10.0.0.1
ip host cucm2.lab.local 10.0.0.2
ip host cucm3.lab.local 10.0.0.3

ip domain name lab.local
ip name-server 8.8.8.8
```



Note: DNS SRV support on Cisco IOS XE is supported on 15.6(1)S / 3.17.00.S and higher.

DNS Debugs and Veriification Commands

<#root>

```
show host
clear host all *
!
debug ip dns view
debug ip domain
debug ccsip info
debug ccsip error
```

DNS Testing 3.15S and later

<#root>

Domain Name Verification

```
Gateway# sh run | s lookup
no ip domain lookup
```

Checking the host table for no entry

```
Gateway# show host
Name lookup view: Global
Default domain is cisco.com
Name/address lookup uses static mappings

Codes: UN - unknown, EX - expired, OK - OK, ?? - revalidate
       temp - temporary, perm - permanent
       NA - Not Applicable None - Not defined

Host                      Port  Flags      Age Type   Address(es)
```

Verification of no PING on a FQDN

```
Gateway# ping cucm.cisco.com
Translating "cucm.cisco.com"
% Unrecognized host or address, or protocol not running.
```

Made a test call here

Checking logs to see if it worked

```
Gateway# sh log | s INVITE sip:
INVITE sip:9001@14.50.228.70:5060 SIP/2.0
INVITE sip:5001@cucm.cisco.com:5060 SIP/2.0
```

Host Table now has an entry

```
Gateway# sh host
Name lookup view: Global
Default domain is cisco.com
Name/address lookup uses static mappings

Codes: UN - unknown, EX - expired, OK - OK, ?? - revalidate
       temp - temporary, perm - permanent
       NA - Not Applicable None - Not defined

Host                      Port  Flags      Age Type   Address(es)
cucm.cisco.com            None  (temp, OK)  0   IP    10.50.244.2
```

CCSIP All output showing a proper DNS Query for the FQDN on the dial-peer.

```
001338: Mar  9 15:29:07.437: //-1/xxxxxxxxxxxx/SIP/Info/info/1024/httpish_msg_free: Freed msg=0x7FE9873
001339: Mar  9 15:29:07.437: //-1/xxxxxxxxxxxx/SIP/Info/notify/8192/sip_dns_type_srv_query: TYPE SRV qu
001340: Mar  9 15:29:07.438: //-1/xxxxxxxxxxxx/SIP/Info/info/8192/sip_dns_type_a_aaaa_query: DNS query
001341: Mar  9 15:29:07.441: //-1/xxxxxxxxxxxx/SIP/Info/notify/8192/sip_dns_type_a_query: TYPE A query
001342: Mar  9 15:29:07.441: //-1/xxxxxxxxxxxx/SIP/Info/info/8192/sip_dns_type_a_query: ttl for A recor
001343: Mar  9 15:29:07.441: //-1/xxxxxxxxxxxx/SIP/Info/info/8192/sip_dns_type_a_aaaa_query: IP Address
001344: Mar  9 15:29:07.441: //-1/xxxxxxxxxxxx/SIP/Info/info/8192/sip_dns_type_a_aaaa_query: 10.50.244.
```

DNS Testing 3.16S and higher.

<#root>

Checking the command is present

```
Gateway# sh run | s lookup
no ip domain lookup
```

Verifying the gateway cannot ping a FQDN

```
Gateway# ping cucm.cisco.com
% Unrecognized host or address, or protocol not running.
```

Checking the Host Table for entries

```
Gateway# sh host
Default domain is cisco.com
Name servers are 10.50.244.52
NAME  TTL  CLASS   TYPE      DATA/ADDRESS
----------------------------------------
```

Made a test call here

CCSIP All Outbound showing the failed call

```
000974: *Mar  9 15:53:01.222: //-1/xxxxxxxxxxxx/SIP/Info/info/1024/httpish_msg_free: Freed msg=0x7FF31D
000975: *Mar  9 15:53:01.222: //-1/xxxxxxxxxxxx/SIP/Info/notify/8192/sip_dns_type_srv_query: TYPE SRV q
000976: *Mar  9 15:53:01.224: //-1/xxxxxxxxxxxx/SIP/Info/info/8192/sip_dns_type_a_aaaa_query: DNS query
000977: *Mar  9 15:53:01.225: //-1/xxxxxxxxxxxx/SIP/Error/sip_dns_type_a_query:
 TYPE A query failed for cucm.cisco.com
000978: *Mar  9 15:53:01.225: //-1/xxxxxxxxxxxx/SIP/Error/_send_dns_fail:
 DNS Query for cucm.cisco.com failed
000984: *Mar  9 20:53:01.225: %VOICE_IEC-3-GW: SIP: Internal Error (DNS query fail): IEC=10.1.128.7.47.
```

Maximum Connections and Bandwidth

By default, VOIP and POTS dial-peers allow for unlimited connections (calls) and bandwidth (VOIP dial-peers only). For trunks that have a limit on the number of calls or bandwidth that can be utilized, it can be useful to employ the **max-conn** or **max-bandwidth** commands. **max-conn** was added in Cisco IOS 11.3(1)T and is present in all Cisco IOS XE versions while max-bandwidth was added in 15.2(2)T and Cisco IOS-XE 3.7S.

Configuration Example

Here you tell the gateway to limit dial-peer 1 to 30 calls using the **max-conn 30** command.
Dial-peer 2 is limiting bandwidth for that dial-peer so that we do not go over the allocated limit.

```
!
dial-peer voice 1 voip
 description ITSP SIP Trunk - 30 Max Calls!
 session protocol sipv2
 sess target ipv4:10.10.10.10
 destination-pattern 8675309$
 max-conn 30
!
dial-peer voice 2 voip
 description SIP Trunk with Bandwidth Restrictions!
```

```
 session protocol sipv2
 sess target ipv4:10.10.10.10
 destination-pattern 123456789$
 max-bandwidth 400
!
```

Sample Error when max-conn threshold is crossed.

```
000308: Oct  5 19:01:02.603: %CALL_CONTROL-6-MAX_CONNECTIONS: Maximum number of connections reached for

000309: Oct  5 19:01:02.603: %VOICE_IEC-3-GW: CCAPI: Internal Error (Dial-peer connections exceeded): I
000310: Oct  5 19:01:02.604: %SIP-3-MAXCONNCAC: Call rejected due to CAC based on maximum number of con

000311: Oct  5 19:01:02.604: //17084/86B070800000/SIP/Msg/ccsipDisplayMsg:
Sent:
SIP/2.0 503 Service Unavailable
Via: SIP/2.0/TCP 10.50.244.62:5060;branch=z9hG4bKb78c35aa21b0
From: <sip:9001@10.50.244.62>;tag=72531~2e8ca155-3f0b-4f07-a1b2-b14ef77ceb7f-26250846
To: <sip:1234@10.50.245.70>;tag=3E19564D-1684
Date: Thu, 05 Oct 2017 19:01:02 GMT
Call-ID: 86b07080-9d61816e-b762-3ef4320e@10.50.244.62
CSeq: 101 INVITE
Allow-Events: telephone-event
Warning: 399 10.50.245.70 "Maximum Number of Connections reached for dial-peer 1"
Server: Cisco-SIPGateway/IOS-15.4.3.S4
Content-Length: 0
```

Direct Inward Dial (DID)

One-Stage Dialing

With Direct Inward Dial enabled on POTS dial-peers, the inbound messaging can contain all the digits necessary to route the call. The Cisco Gateway cannot do subsequent digit collection. When the router or gateway searches for an outbound dial-peer, the device uses the entire incoming dial string. This matching is variable-length by default. This match is not done digit-by-digit because by DID definition, all digits have been received. This is the default configuration for POTS dial-peers.

Full Documentation: Understanding Direct-Inward-Dial (DID) on IOS Voice Digital (T1/E1) Interfaces

Configuration Example

```
!
dial-peer voice 1 pots
 incoming called-number 8675309
 voice-port 0/0/0
 direct-inward-dial
!
```

Two-Stage Dialing

If the incoming POTS dial-peer is configured with no direct-inward-dial, the router or gateway enters the digit collection mode (digits are collected inband). Outbound dial-peer matching is done on a digit-by-digit basis. The router or gateway checks for dial-peer matches after the device has received each digit, and then routes the call when a full match is made.

Configuration Example

```
!
dial-peer voice 1 pots
 incoming called-number 8675309
 voice-port 0/0/0
 no direct-inward-dial
!
```

Blocking Calls

Each protocol handles call blocking a bit different. Most protocols can make use of the translation-rule reject pattern which blocks based on a digit string. If an administrator wants to still apply an inbound translation-profile for normal digit manipulation, but not block any numbers within, there is an option of implementing a call-block using the **call-block translation-profile** command.

```
!
voice translation-rule 164
 rule 1 reject /8675309/
!
voice translation-profile CALLBLOCK
 translate calling 164
!
dial-peer voice 1 pots
 desc INCOMING VOICE-PORT with BLOCK
 translation-profile incoming ANOTHER
 call-block translation-profile incoming CALLBLOCK
 call-block disconnect-cause incoming invalid-number
 incoming called-number .
 port 0/0/0:23
!

Gateway#test voice translation-rule 164 8675309
8675309 blocked on rule 1
```

Within E1 R2 there exists the ability for an administrator to block Collect Calls. This is mainly seen and employed in Brazil deployments, but can be configured via any cas-custom group.

The two options are:

1. Block the incoming collect call based on the category II-8 which is received from the telco switch. This is the default mechanism and all Cisco Gateways perform this without any configuration. This method of blocking requires a newer telco switch which supports category based marking. To disable this method use the command **collect-call-enable** on the cas-custom group.
2. Utilize the **double-answer** feature on an r2-digital E1 R2 cas-custom group. The double-answer configuration disables the category based blocking, and replaces it with the double-answer feature. This works by first answering the incoming call and starting a 1 second timer. After 1 second, the gateway sends a release in the form of a CLEAR BWD command. The Telco can then send a CLEAR FWD to the gateway and the call can disconnect. A timer starts after the gateway sends the CLEAR BWD, and if this timer expires, the gateway sends another ANSWER signal thus assuming the call is not a collect call and from here the call proceeds as normal. This timer can be configured using cc-reanswer-to within the the cas-custom group.

Category II-8 Block message (debug vpm signal)

<#root>

```
009228: Nov 21 12:02:00.955 GMT: //-1/BF12BE36BAC8/VTSP:(0/0/0:0):-1:1:2/vtsp_report_cas_digit:
   Begin Digit=8, Mode=CC_TONE_R2_MF_BACKWARD_MODE
009229: Nov 21 12:02:00.955 GMT: htsp_digit_ready_up(0/0/0:0(2)):

Rx digit='8'

009230: Nov 21 12:02:00.955 GMT: R2 Incoming Voice(0/0): DSX (E1 0/0/0:1): STATE: R2_IN_CATEGORY R2 Got
009231: Nov 21 12:02:00.955 GMT: Enter r2_comp_category
009232: Nov 21 12:02:00.955 GMT:

R2 Event : 8

009233: Nov 21 12:02:00.955 GMT:

#######R2_II8 TRUE########

009234: Nov 21 12:02:00.955 GMT:

####### collect_call_enable = 0

009235: Nov 21 12:02:00.955 GMT:

############sending B7 ##########

009236: Nov 21 12:02:00.955 GMT: r2_reg_generate_digits(0/0/0:0(2)):

Tx digit '7'

009237: Nov 21 12:02:01.055 GMT: //-1/BF12BE36BAC8/VTSP:(0/0/0:0):-1:1:2/vtsp_report_cas_digit:
   End Digit=8, Mode=CC_TONE_R2_MF_BACKWARD_MODE
009238: Nov 21 12:02:01.055 GMT: htsp_digit_ready(0/0/0:0(2)): Rx digit='#'
009239: Nov 21 12:02:01.055 GMT: R2 Incoming Voice(0/0): DSX (E1 0/0/0:1): STATE: R2_IN_CATEGORY R2 Got
009240: Nov 21 12:02:01.055 GMT: Enter r2_comp_category
009241: Nov 21 12:02:01.055 GMT: r2_reg_generate_digits(0/0/0:0(2)): Tx digit '#'
009242: Nov 21 12:02:01.359 GMT: htsp_dsp_message: SEND_SIG_STATUS: state=0x8 timestamp=22365 systime=2
009243: Nov 21 12:02:01.359 GMT: htsp_process_event: [0/0/0:0(2), R2_Q421_IC_WAIT_ANSWER, E_DSP_SIG_100
009244: Nov 21 12:02:01.359 GMT: r2_q421_ic_clr_fwd_idle(0/0/0:0(2)) Rx CLEAR FWD
009245: Nov 21 12:02:01.359 GMT: r2_reg_channel_disconnected(0/0/0:0(2))
009246: Nov 21 12:02:01.359 GMT: R2 Incoming Voice(0/0): DSX (E1 0/0/0:1): STATE: R2_IN_CATEGORY R2 Got
009247: Nov 21 12:02:01.359 GMT: Enter r2_comp_category
009248: Nov 21 12:02:01.359 GMT: htsp_timer - 2000 msec
009249: Nov 21 12:02:01.359 GMT: htsp_process_event: [0/0/0:0(2), R2_Q421_IC_CLR_FWD, E_HTSP_RELEASE_REQ
009250: Nov 21 12:02:01.359 GMT: r2_q421_null_release(0/0/0:0(2)) E_HTSP_RELEASE_REQ
009251: Nov 21 12:02:01.359 GMT: r2_reg_process_event: [0/0/0:0(2), R2_REG_COLLECTING, E_R2_REG_DISCONN
009252: Nov 21 12:02:01.359 GMT: r2_reg_disconnect_collect(0/0/0:0(2))
```

```
009253: Nov 21 12:02:01.359 GMT: r2_reg_timer_stop(0/0/0:0(2))
009254: Nov 21 12:02:01.711 GMT: htsp_process_event: [0/0/0:0(1), R2_Q421_IC_CLR_FWD, E_HTSP_EVENT_TIME
009255: Nov 21 12:02:01.711 GMT: htsp_timer_stop
009256: Nov 21 12:02:01.711 GMT: r2_q421_clr_fwd_idle(0/0/0:0(1)) Tx IDLEvnm_dsp_set_sig_state:[R2 Q.42
009257: Nov 21 12:02:01.711 GMT: r2_reg_channel_disconnected(0/0/0:0(1))
009258: Nov 21 12:02:01.711 GMT: //68220 6/0C63B263B9C9/VTSP:(0/0/0:0):0:1:1/vtsp_do_call_history:
   Coder Rate=5
009259: Nov 21 12:02:01.711 GMT: r2_reg_process_event: [0/0/0:0(1), R2_REG_IDLE, E_R2_REG_DISCONNECT(91
```

Double-Answer Configuration Example

```
!
controller e1 0/0/0
 ds0-group 0 timeslots 1-15,17-31 type r2-digital r2-compelled ani
 cas-custom 0
  country brazil
  double-answer
  cc-reanswer-to 3000
!
```

Double-Answer Debugs (debug vpm signal)

<#root>

Answer the call and start a 1 second timer

```
May 23 09:52:59.180 BR: r2_q421_ic_answer(0/0/0:0(18))
```

Tx ANSWER

```
 seizure: delay 0 ms,elapsed 12404 msvnm_dsp_set_sig_state:[R2 Q.421 0/0/0:0(18)] set signal state = 0x
May 23 09:52:59.180 BR: r2_reg_channel_connected(0/0/0:0(18))
May 23 09:52:59.180 BR:
```

htsp_timer - 1000 msec

```
May 23 09:52:59.180 BR: //23899578/92233E71B421/CCAPI/cc_api_voice_mode_event:
   Call Id=23899578
May 23 09:52:59.180 BR: //23899578/92233E71B421/CCAPI/cc_api_voice_mode_event:
   Call Entry(Context=0x1E73AD8)
May 23 09:52:59.180 BR: htsp_process_event: [0/0/0:0(18), R2_Q421_IC_DOUBLE_ANS_ANS, E_HTSP_VOICE_CUT_T
May 23 09:52:59.184 BR: //23899578/92233E71B421/CCAPI/cc_process_notify_bridge_done:
   Conference Id=0x10AD1, Call Id1=23899578, Call Id2=23899579
May 23 09:52:59.184 BR: r2_reg_process_event: [0/0/0:0(18), R2_REG_WAIT_FOR_CONNECT, E_R2_REG_CONNECT(9
May 23 09:52:59.184 BR: r2_reg_connect(0/0/0:0(18))
```

One Second Passes and we clear the call and start a 2 second timer

```
May 23 09:53:00.180 BR: htsp_process_event: [0/0/0:0(18), R2_Q421_IC_DOUBLE_ANS_ANS, E_HTSP_EVENT_TIMER
May 23 09:53:00.180 BR: r2_q421_ic_d_answ_answ_to(0/0/0:0(18)) E_TIMER_EVENT
May 23 09:53:00.180 BR: htsp_timer - 2000 msec
May 23 09:53:00.180 BR: r2_q421_ic_d_answ_answ_to(0/0/0:0(18))
```

Tx CLEAR BWD

```
vnm_dsp_set_sig_state:[R2 Q.421 0/0/0:0(18)] set signal state = 0xC
```

```
May 23 09:53:00.824 BR: htsp_process_event: [0/0/0:0(18), R2_Q421_IC_DOUBLE_ANS_RLS, E_DSP_SIG_1000]
May 23 09:53:00.824 BR: r2_q421_ic_answer_clr_fwd(0/0/0:0(18))
```

Rx CLEAR FWD

```
May 23 09:53:00.824 BR: r2_reg_channel_disconnected(0/0/0:0(18))
May 23 09:53:00.824 BR:
```

htsp_timer - 2000 msec

```
May 23 09:53:00.824 BR: r2_reg_process_event: [0/0/0:0(18), R2_REG_CONNECTED, E_R2_REG_DISCONNECT(91)]
May 23 09:53:00.824 BR: r2_reg_disconnect_idle(0/0/0:0(18))
May 23 09:53:00.824 BR: R2 Incoming Voice(0/0): DSX (E1 0/0/0:17): STATE: R2_IN_IDLE R2 Got Event R2_ST
May 23 09:53:00.824 BR: r2_reg_timer_stop(0/0/0:0(18))
```

2 second passes and the gateway release the call

```
May 23 09:53:02.824 BR: htsp_process_event: [0/0/0:0(18), R2_Q421_IC_CLR_FWD, E_HTSP_EVENT_TIMER]
May 23 09:53:02.824 BR: htsp_timer_stop
May 23 09:53:02.824 BR: r2_reg_channel_disconnected(0/0/0:0(18))
May 23 09:53:02.824 BR: //23899578/92233E71B421/VTSP:(0/0/0:0):17:1:1/vtsp_cc_call_disconnected:
   Cause Value=16
May 23 09:53:02.824 BR: //23899578/92233E71B421/CCAPI/cc_api_call_disconnected:
   Cause Value=16, Interface=0xB41CEBC, Call Id=23899578
```

ISDN overlap-receiving Command

There are implications for inbound dial-peer matching when the **isdn overlap-receiving** command is configured on ISDN interfaces. After every digit is received at the ISDN layer, dial-peers are checked for matches. If a full match is made, the call is routed immediately (to the session app in this case) without waiting for additional digits. The T terminator can be used to suspend this digit-by-digit matching and force the router or gateway to wait until all digits are received. The T refers to the T302 interdigit timer at the ISDN level, configurable under the serial interface associated with the ISDN interface. ISDN also provides other mechanisms to indicate the end of digits, such as setting the Sending Complete Information Element (IE) in Q.931 information messages.

Empty Called Number

The Warning message shown, displays when dial-peer is configured with incoming called-number T.

Sample Output

```
Gateway(config)# dial-peer voice 1 pots
Gateway(config-dial-peer)# incoming called-number T
Warning: Pattern T defines a match with zero or more digits and hence could
match with an empty number. If this is not the desired behaviour please
configure pattern .T instead to match on one or more digits
```

Special Notes about incoming dial-peer match with an empty called number.

- A null called-number is considered less qualified compared to a voice-port and/or in some cases answer-address. Therefore, a match based on a null called number can occur only if there is no match based on either answer-address or port-number.

- In case of overlap dialing, a null called number does not match incoming called-number T because timeout has not occurred.
- A null called-number can match incoming called-number T only in case of ENBLOCK and there is no match either because of answer-address and port-number. The warning displayed when an administrator configures incoming called-number T refers to this specific case.

Class of Restriction

Class of Restriction (COR) is a way to limit calls on a Cisco Gateway. COR is often described as a lock and key mechanism. Locks are assigned to dial-peers with an outgoing COR list. Keys are assigned to dial-peers with an incoming COR list. When COR Lists are applied, the available outbound dial-peers are those that the key can unlock. This filtering occurs before the rest of the outbound dial-peer matching methods are then checked.

Two important rules with Class of Restriction:

1. If there is no outgoing COR list applied the call is always routed.
2. If there is no incoming COR list the call is always routed.

Configuration of Class of Restriction (COR), Logical Partitioning Class of Restriction (LPCOR), and LPCOR with Forced Authorization Codes (FAC) are beyond the scope of this document but these documents can be referenced for further reading.

COR	Configuring Class of Restrictions (COR)
LPCOR w/ CME	CME with LPCOR Configuration Example
LPCOR w/CME and FAC	Cisco Unified Communications Manager Express System Administrator Guide

Cisco Unified Communications Manager Express (CUCME) Dial-peers

CME creates system dial-peers for ephones and voice register pools. These cannot be seen in the running config. To make changes to the CME dial-peers, the changes need to be done on the actual ephone or voice register pool. When viewing show dial-peer voice summary outputs, the dial-peer starting with 2000 are SCCP ephones and dial-peers starting with 4000 are SIP voice register pools. This dial-peer shows up as the inbound dial-peer for calls from CME registered phones and the outbound dial-peer in debugs for call to CME registered phones.

Example Output for show dial-peer voice summary with CME.

```
Gateway# show dial-peer voice sum | s 2000|4000
20001  pots  up   up            1001$            0                              50/0/1
20002  pots  up   up            4001$            0                              50/0/2
20003  pots  up   up            4002$            0                              50/0/3
20004  pots  up   up            7001$            0                              50/0/4
20005  pots  up   up            3009$            0                              50/0/5
20006  pots  up   up            8810....$        0                              50/0/10
20007  pots  up   up            8811....$        0                              50/0/11
40001  voip  up   up            14085151111$     0  syst ipv4:14.50.214.67:50
```

```
40002  voip  up   up            19725252222$       0  syst ipv4:14.50.214.67:50
40003  voip  up   up            85225353333$       0  syst ipv4:14.50.214.67:50
40004  voip  up   up            442084445555$      0  syst ipv4:14.50.214.67:50
40005  voip  up   up            911$               0  syst ipv4:14.50.214.67:50
40006  voip  up   up            18005550100$       0  syst ipv4:14.50.214.67:50
40008  voip  up   up            2001$              0  syst ipv4:14.50.214.51:50
```

Example output for show voice register dial-peers with SIP CME.

```
Gateway# show voice register dial-peers
Dial-peers for Pool 2:

dial-peer voice 40006 voip
 destination-pattern 14085151111$
 session target ipv4:14.50.214.67:5060
 session protocol sipv2
 dtmf-relay rtp-nte
 digit collect kpml
 codec  g711ulaw bytes 160
 no vad
  call-fwd-all         8888
  after-hours-exempt   FALSE

dial-peer voice 40005 voip
 destination-pattern 19725252222$
 session target ipv4:14.50.214.67:5060
 session protocol sipv2
 dtmf-relay rtp-nte
 digit collect kpml
 codec  g711ulaw bytes 160
 no vad
  after-hours-exempt   FALSE
```

MGCP and SCCP with Dial-Peers

MGCP and SCCP follow their own rules for dial-peers. The only concept they utilize is that they must be configured with the desired voice-port for the call. The rest is handled by the STCAPP and MGCPAPP process. When you examine the configuration of these dial-peers, they either have the command **service mgcpapp** or **service stcapp**. These enable the dial-peer for the application of choice, as well as tell the application which dial-peer it can handle.

When debugging these protocols, the output never displays an inbound dial-peer match. This can always show as dial-peer 0. Because it does not exist. The Call Agent handling the application has already chosen which port to send the call to and inbound dial-peer matching is useless since the gateway has no control over that leg of the call. However, an outbound dial-peer match can be observed. This is merely for show as ultimately the call agent handling the process has control over that side of the call as well.

Remember, the dial-peer only tells the application of choice which physical voice-port to control. Since the majority of this is controlled by an external call agent and the gateway it just does what it is told. You are going to be skipping the underlying how to on this section, and provide a few configurations to get started.

Sample MGCP configuration [with CUCM Auto-Configuration*]

```
!
mgcp call-agent 10.10.10.10
mgcp
!
ccm-manager mgcp [codec-all]
ccm-manager config server 10.10.10.10
ccm-manager config
ccm-manger redundant-host 10.10.10.20
!
voice-port 0/0/0
 description The MGCP port to register
 no shut
!
dial-peer voice 1 pots
 description Defining the Port for the MGCP application
 service mgcpapp
 port 0/0/0
!
hostname myrouter
ip domain name cisco.com
ip name server 10.10.10.30
!
ip tftp source-interface gig0/0/0
!
```

Full MGCP Documentation: [Cisco Unified Communications Manager and Interoperability Configuration Guide, Cisco IOS Release 15M&T]

Sample SCCP / STCAPP Configuration [with CUCM Auto-Configuration*]

```
!
stcapp ccm-group 1
stcapp
!
sccp local gig0/0/0
sccp ccm 10.10.10.10 id 1 priority 1 version 7.0+
sccp ccm 10.10.10.20 id 1 priority 2 version 7.0+
sccp
!
sccp ccm group 1
 bind interface gig0/0/0
 associate ccm 1 priority 1
 associate ccm 2 priority 2
!
ccm-manager config server 10.10.10.10
ccm-manager sccp local gig0/0/0
ccm-manager sccp
!
voice-port 0/0/0
 description The SCCP port to register
 no shut
!
dial-peer voice 1 pots
 description Defining the Port for the SCCP application
 service stcapp
 port 0/0/0
!
ip tftp source-interface gig0/0/0
```

```
!
```

If an administrator does not want CUCM to configure the gateway, simply remove the **ccm-manager** commands. The dial-peer configuration is included to drive home the point about how the concept works. With ccm-manager configurations present, CUCM creates these dial-peers based on the port configuration in CUCM so there is no need to actually define the dial-peer. The CUCM created dial-peers usually start with 999 and are then three more digits.

SIP DSAPP with Dial-Peers

SIP DSAPP was added in Cisco IOS XE 16.12.1+ and CUCM 12.5.1SU+

With this feature, analog voice ports such as FXS can be registered and managed by CUCM. Call routing with DSAPP is slightly different than MGCP or SCCP as the dial-peers are still matched normally. That is, the gateway can collect digits from the FXS port and do a dial-peer lookup on the VOIP dial-peers. After a match is found, the INVITE is sent to CUCM enblock for CUCM to perform further digit analysis.

Sample SIP DSAPP Configuration [with CUCM Auto-Configuration*] | Cisco IOS-XE 16.12.1+ and CUCM 12.5.1SU+

```
!
dsapp line
!
voice service voip
 sip
  bind control source-interface GigabitEthernet0/0/0
  bind media source-interface GigabitEthernet0/0/0
  session transport tcp
!
application
 service dsapp
  param dialpeer 777
!
global
 service default dsapp
!
ccm-manager config server 10.10.10.10
ccm-manager sipana auto-config local GigabitEthernet0/0/0
!
dial-peer voice 777 voip
 destination-pattern 9T
 session protocol sipv2
 session target ipv4:10.10.10.10
 session transport tcp
 incoming called-number .
 voice-class sip extension gw-ana
 voice-class sip bind control source-interface GigabitEthernet0/0/0
 dtmf-relay rtp-nte
 codec g711ulaw
!
dial-peer voice 19990100 pots
 service dsapp
 destination-pattern 7776
 voice-class sip extension gw-ana
 port 0/1/0
!
```

```
sip-ua
 registrar ipv4:10.10.10.10 expires 3600 tcp
!
```

Full SIP DSAPP Documentation: Cisco VG450 Voice Gateway Software Configuration Guide.

Call Routing Troubleshoot and Verify

Please see this document for more detailed information.

- Configure Debug Collection for Unified Border Element (CUBE) and Time-Division Multiplexing (TDM) Gateways