Cisco Unified Border Element Configuration Guide

Last Modified: Marcn

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[redacted]

[redacted]

CHAPTER

New and Changed Information

New and Changed Information

This section provides the details of new and changed information in Cisco in

[redacted]

CHAPTER

Supported Platforms

CUBE is supported on

Cisco Router Platforms	Cisco Router Models	Cisco IOS Software Releases
		Cisco IOS XE 3.15 onwards Cisco IOS XE Denali

[redacted]

CHAPTER

Overview of Cisco Unified Border Element

Cisco Unified Border

CUBE extends the functionality provided by conventional session border controllers (SBCs) in terms of protocol interworking, especially

SIP/H.323 Trunking

The Session Initiation Protocol (SIP) is a signaling

How to Configure Basic CUBE Features

Consider a scenario where XYZ corporation uses a VoIP network to provide phone services and uses a PRI connection for telecommunications services, and the PRI trunk is controlled by MGCP. Migration from MGCP PRI to SIP trunk is provided by ITSP telecommunications. CUCM sends the telephone number, as

Enabling the CUBE Application on a Device

SUMMAR

Verifying the CUBE Application on the Device

SUMMARY STEPS

1. **enable**
2. **show cube status**

DETAILED STEPS

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** Example: `Device> enable`	Enables privileged EXEC mode. Enter

Table 1: Feature Information for Virtual CUBE Support

Features Supported with Virtual CUBE

Virtual CUBE supports most of the features available in CUBE. Any feature

Licensing Package Support

Virtual CUBE is enabled with the APPX and AX license packages. The AX license package provides access

You can use the .iso file to

[redacted]

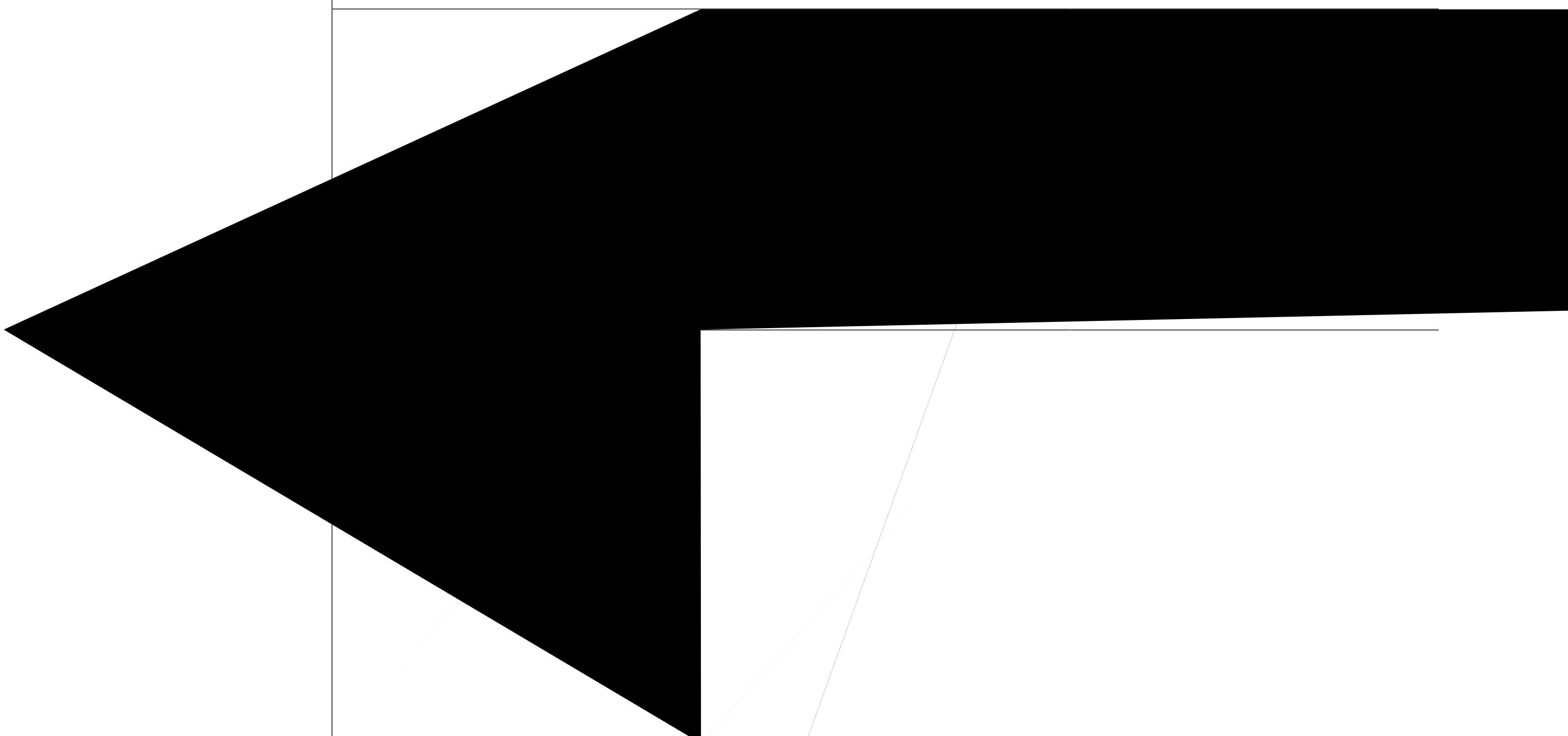
CHAPTER

Dial-Peer Matching

CUBE allows VoIP-to-VoIP

In CUBE, dial peers can also be classified as

A W



answer-address *ANI-string*

The

CHAPTER

Information About DTMF Relay

DTMF Tones

DTMF tones are used during a call to signal to a far-end device; these signals maybe for navigating a menu system, entering data

the call leg. The signal and duration headers specify the digit, in this case 1, and duration, 160 milliseconds in the example, for DTMF tone play.

```
INFO sip:2143302100@172.17.2.33 SIP/2.0
Via: SIP/2.0/UDP 172.80.2.100:5060
From: <sip:9724401003@172.80.2.100>;tag=43
To: <sip:2143302100@172.17.2.33>;tag=9753.0207
Call-ID: 984072_15401962@172.80.2.100
CSeq: 25634 INFO
Supported: 100rel
Supported: timer
Content-Length: 26
Content-Type: application/dtmf-relay
Signal= 1
Duration= 160
```

rtp-nte Real-Time Transport Protocol (RTP) Named Telephone Events (NTE). This is an in-band DTMF relay

If you configure rtp-nte, sip-notify, and sip-kpml, the outgoing INVITE contains a SIP Call-Info header, an Allow-Events header with KPML, and an

Table 7: RTP-RTP Flow-Through

Table 10: RTP-RTP with high-density transcoder Flow Through

Media Source IP Addr:Port:

```
   Number of SIP User Agent Server(UAS) calls: 2
```

Step 3 **show sip-ua history dtmf-relay kpml**

The following sample output displays SIP call history with KMPL DTMF Relay mode.

Example:

```
Device# show sip-ua history dtmf-relay kpml

Total SIP call legs:2, User Agent Client:1, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID                : D0498774-F01311E3-82A0DE9F-78C438FF@10.86.176.119
   State of the call       : STATE_ACTIVE (7)
   Calling Number          : 2017
   Called Number           : 1011
   CC Call ID              : 257
No.         Timestamp         Digit           Duration
=======================================================

Call 2
SIP Call ID                : 22BC36A5-F01411E3-81808A6A-5FE95113@10.86.176.142
   State of the call       : STATE_ACTIVE (7)
   Calling Number          : 2017
   Called Number           : 1011
   CC Call ID              : 256
No.         Timestamp         Digit           Duration
=======================================================

   Number of SIP User Agent Client(UAC) calls: 2

SIP UAS CALL INFO
Call 1
SIP Call ID                : 22BC36A5-F01411E3-81808A6A-5FE95113@10.86.176.142
   State of the call       : STATE_ACTIVE (7)
   Calling Number          : 2017
   Called Number           : 1011
   CC Call ID              : 256
No.         Timestamp         Digit
```

Called Number

[redacted]

CHAPTER

Introduction to Codecs

A codec is a device or software capable of encoding or decoding a digital data stream or signal. Audio codecs can code or M

The illustrations below

numbers in that range, whereas RTcP uses the odd port numbers. While RTP is responsible for carrying the

G.729 Annex-B

How to Configure Codecs

Configuring Audio and Video Codecs at the Dial Peer Level

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer**

Configuring Audio Codecs Using a Codec Voice Class and Preference Lists

Preferences can be used to determine which codecs will be selected over

DETAILED STEPS

show call active voice [compact]
Displays a compact version of call

Table 17: Feature Information for SIP Binding

Bind configuration at global level

Best local IP address to reach the destination

The table below describes the state of the system when the

Table 19: State of the Interface for the bind Command

	Command or Action	Purpose

Example:

Example:

```
Device# show dial-peer voice 101

VoiceOverIpPeer1234
        peer type = voice, system default peer = FALSE, information type = voice,
        description = `',
        tag = 1234, destination-pattern = `',
        voice reg type = 0, corresponding tag = 0,
        allow watch = FALSE
        answer-address = `', preference=0,
        CLID Restriction = None
        CLID Network Number = `'
        CLID Second Number sent
        CLID Override RDNIS = disabled,
        rtp-ssrc mux = system
        source carrier-id = `', target carrier-id = `',
        source trunk-group-label
```

Table 22: Feature Information for Configuring Path of Media

CHAPTER

Table 23: Feature Information for SIP Profiles

businesses may have policies for the information that can enter or exit their networks for policy or security reasons from a service provider SIP trunk.

Figure 18: SIP Profile

In order

The rules configured for an INVITE message are applied only to the first INVITE of

Restrictions for SIP Profiles

Removal or addition of mandatory headers is not supported. You can only modify mandatory headers Mandatory SIP headers include To, From, Via, CSeq, Call-Id, and

Configuring a SIP Profile to Manipulate SIP Request or Response Headers

SUMMARY STEPS

SUMMAR

Example: Configuration to Remove an Attribute

```
response ANY sdp-header mline-index 4 a=test REMOVE
```

Configuring SIP Profile Using Rule Tag

Configure SIP profile rules using the rule tag,

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *profile-id*
4. Enter

Configuring a SIP Profile for Non-standard SIP Header

SUMMARY STEPS

1. **enable**
2.

Upgrading or Downgrading SIP Profile Configurations

You can upgrade or downgrade all the SIP Profile configurations to rule-format or non-rule format automatically.

We recommend that you downgrade the SIP profiles to non-rule format configuration before migrating to a version below Cisco IOS Release 15.5(2)T or Cisco IOS-XE Release 3.15S. If you migrate without downgrading the SIP

DETAILED STEPS

show dial-peer voice *id* | **include profile**

Displays information related to SIP profiles configured on the specified dial peer.

Example:

```
Device# show dial-peer voice 10 | include profile

        Translation profile (Incoming):
        Translation profile (Outgoing):
        translation-profile = `'
        voice class sip profiles = 11
        voice class sip profiles inbound = 10
```

Troubleshooting SIP Profiles

SUMMARY STEPS

1. **debug ccsip all**

DETAILED STEPS

debug ccsip all

This command displays the applied SIP profiles.

Example:

Applied

```
                        New SDP header is added : b=AS: 1600
Oct 12 06:51:53.647: //-1/xxxxxxxxxxxx/SIP/Info/
                        sip_profiles_update_content_length:
                        Content length header before modification :
                        Content-Length:
```

Example: Adding "a=ixmap:0 ping" in M-Line number 4 of the INVITE SDP Request Messages

```
Device(config)# voice class
```

Replace "CiscoSystems-SIP-GW-UserAgent" with "-" in the Originator Header of the SDP in INVITE Request Messages

```
Device(config)# voice
```

Remove Server Header from 100 and 180 SIP Response Messages

```
Device(config)# voice class sip-profiles 20
Device(config-class)# response
```

Example: Upgrading and Downgrading SIP Profiles automatically

Upgrading SIP Profiles to rule-format

The following is a snippet from

The SIP profile will look for a diversion header containing "<sip:5..." , where ... stands for the three-digit extension and then concatenates 9789365 with these three digits.

Original Diversion Header:

```
Diversion:<sip:5100@161.44.77.193>;privacy=off;reason=unconditional;counter=1;screen=no
```

CiscoSystemsSIP-GW-UserAgent has been replaced with -.

The Audio-Bandwidth SDP header has been added with the value b=AS:1600.

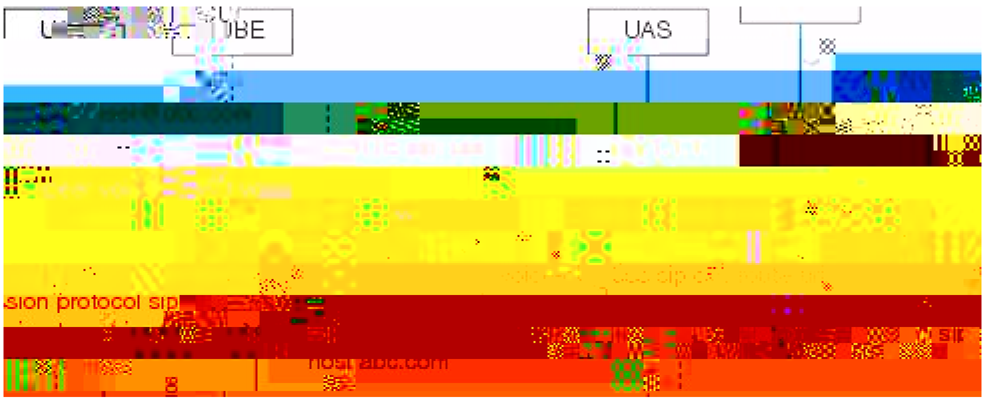
```
INVITE sip:2222000020@9.13.40.250:5060 SIP/2.0
Via: SIP/2.0/UDP 9.13.40.249:5060;branch=z9hG4bK1A203F
From: "sipp " <sip:1111000010@9.13.40.249>;tag=F11AE0-1D8D
To: <sip:2222000020@9.13.40.250>
Date: Mon, 29 Oct 2007 19:02:04 GMT
Call-ID: 4561B116-858811DC-804DEF2E-4CF2D71B@9.13.40.249
Content-Length: 279

v=0
o=- 6906 8069 IN IP4 9.13.40.249
s=SIP Call
c=IN IP4 =                    with                              -.
```

DETAILED STEPS

SIP 2

Case 3: The old behavior of setting the outbound Request-URI to session target is retained when the **requri-passing** command is not enabled.



Case 4: The session target derived from the host part of the URI. The UR

Configuring Pass Though of Request URI and To Header URI (Global Level)

SUMMARY STEPS

Configuring Pass Though of Request URI and To Header URI (Dial Peer Level)

SUMMARY STEPS

DETAILED STEPS

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class uri** *destination-tag* **sip**
4. **host** *hostname-pattern*
5. **exit**
6. **dial-peer voice** *tag* **voip**

   voice class uri v *vl* **v**
7. **session protocol sipv2**
8. **destination uri** *destination-tag*
9. **session target sip-uri**

```
Device(conf-serv-sip)# requri-passing
Device(conf-serv-sip)# end
```

Example: Configuring Pash iguring P0HUYVLSVLS

```
Device(config-voice-uri-class)# host abc.com
Device(config-voice-uri-class)# end
```

Additional References for URI-Based Dialing Enhancements

Related Documents

Table 24: Feature Information for URI-Based Dialing Enhancements

T

You can match a pattern

[redacted]

CHAPTER

Outbound Dial-Peer Group as an Inbound Dial-Peer Destination

This feature can group multiple outbound dial peers into a

Verifying Outbound Dial-Peer Groups as an Inbound Dial-Peer Destination

SUMMARY STEPS

1. **show voice cle**

DETAILED STEPS

Enter the following:

debug voip dialpeer inout

debug voip ccapi inout

Displays the configuration of an outbound dial-peer group.

Example:

```
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/dpMatchCore:
   Dial String=4001, Expanded String=4001, Calling Number=
   Timeout=TRUE, Is Incoming=TRUE, Peer Info Type=DIALPEER_INFO_SPEECH
*Jul 19 10:15:53.310 IST: //-1/xxxxxxxxxxxx/DPM/vepm_match_pattern_map:
   DEPM 1000 use caching dialstring 4001 status 0
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/MatchNextPeer:
```

Incoming dial peer is first matched:

```
Result=Success(0); Incoming Dial-peer=600 Is Matched
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/dpMatchPeertype:exit@6602
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/dpAssociateIncomingPeerCore:
   Result=Success(0) after DP_MATCH_INCOMING_DNIS; Incoming Dial-peer=600
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/dpMatchSafModulePlugin:
   dialstring=NULL, saf_enabled=0, saf_dndb_lookup=0, dp_result=0
*Jul 19 10:15:53.310 IST: //-1/ED647BD1B0F9/DPM/dpAssociateIncomingPeerSPI:exit@7181
*Jul 19 10:15:53.311 IST: //-1/ED647BD1B0F9/DPM/dpMatchPeersCore:
   Calling Number=, Called Number=4001, Peer Info Type=DIALPEER_INFO_SPEECH
```

The dial-peer group associated with a dial peer is selected:

```
*Jul 19 10:15:53.311 IST: //-1/ED647BD1B0F9/DPM/dpMatchPeersCore:
   Outbound Destination DPG Group Request; Destination DPG=1
*Jul 19 10:15:53.311 IST:
```

Configuration Examples for Outbound Dial Peer Group as an Inbound Dial-Peer Destination

```
Device> enable
Device# configure terminal
! Configuring outbound dial peers
```

[redacted]

CHAPTER

Inbound Leg Headers for Outbound Dial-Peer Matching

The Inbound Leg

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on

Information About Inbound Leg Headers for Outbound Dial-Peer Matching

This feature allows you to match headers of an inbound call leg and provision an outbound dial peer for an outbound call leg.

SUMMARY STEPS

1. **enable**

Verifying Inbound Leg Headers for Outbound Dial-Peer Matching

SUMMARY STEPS

1. **show dialplan incall {sip | h323} {calling | called}** *e164-pattern* | **include voice**
2. **show dialplan dialpeer** *inbound-dial-peer-id* **number** *e164-pattern* **[timeout] | include Voice**
3. **show voice class dial-peer provision-policy**

DETAILED STEPS

Step 1 **show dialplan**

Displays a list of outbound dial peers based on a specified inbound

```
 Pref  Policy Rule
 ----  -----------
 1 referred-by via
 2 uri

voice class dial-peer provision-policy: 300  AdminStatus: Up
 Description: match only request-uri

 Pref  Policy Rule
 ----  -----------
 1 uri

Voice class dial-peer provision-policy: 400  AdminStatus: Up
 Description: match only request uri; if no match then match called

 Pref  Policy Rule
 ----  -----------
 1 uri
 2 called
```

```
Device(config)#
```

[redacted]

CHAPTER [redacted]

Information About Server Groups in Outbound Dial Peers

You can now group IPv4 and IPv6 addresses of servers and configure it as in an outbound SIP dial-peer destination. A server group is first created and associated with a SIP outbound dial peer. When a call matchespeer

```
Device(config-dial-peer)# session server-group 171
Device(config-dial-peer)# end

! Displays the configurations made for the outbound dial peer 181 associated with a server
 group
Device# show voice class server-group dialpeer 181

Voice class server-group: 171    AdminStatus: Up
 Hunt-Scheme: preference
 Total Remote Targets: 3

 Pref   Type   IP Address                              IP Port
 ----   ----   ----------                              -------
 1      ipv4   10.1.1.1
 2      ipv4   10.1.1.2
 3      ipv4   10.1.1.3

! Displays the configurations made for the server group.

Device# show voice class server-group 171

Voice class server-group: 171
 AdminStatus: Up                 OperStatus: Up
 Hunt-Scheme: preference         Last returned server: 10.1.1.1
 Description: It has 3 entries
 Total server entries: 3

 Pref   Type   IP Address                              IP Port
 ----   ----   ----------                              -------
 1      ipv4   10.1.1.1
 2      ipv4   10.1.1.2
 3      ipv4   10.1.1.3          oup                         ز171          ٣                onfig)#

-------------------------------------
```

Server Groups in Outbound Dial Peers (Round-Robin-Based Selection)

```
! Configuring the Server Group
Device(config)# voice class server-group 171
```

```
Device# show voice class server-group 171

Voice class server-group: 171
 AdminStatus: Up                OperStatus: Up
 Hunt-Scheme: round-robin       Last returned server: 10.1.1.1
 Description: It has 3 entries
 Total server entries: 3

 Pref   Type   IP Address                                IP Port
 ----   ----   ----------                                -------
```

[redacted]

CHAPTER [redacted]

Domain-Based Routing Support on the Cisco UBE

First Published:

r Dom

With the introduction of the domain-based routing feature, all parameters including the domain name of the request URI will be sent to the application and the

DETAILED STEPS

Example:

```
Device> enable
```

Step 2 **debug ccsip all**

Enables all SIP-related debugging.

Example:

```
Device# debug
```

The following event shows the matched dial peers in the order of priority:

Example:

```
List of Matched Outgoing Dial-peer(s):
     1: Dial-peer Tag=3600
     2: Dial-peer Tag=36
```

Configuration Examples for Domain-Based Routing Support on the Cisco UBE

Example Configuring Domain-Based Routing Support on the Cisco UBE

The following example shows how to enable domain-based routing support on the Cisco

Feature Information for ENUM Enhancement per Kaplan Draft RFC

The following table provides release information about the feature or features described in this module. This table lists only the

```
1.0.9.3.e164-test sip:540 NA      (temp, OK)  0     NAPTR 0 0 U sip+E2U /^.*$/sip:3901@10.1.18.28/
```

PART III

Multi-Tenancy

[redacted]

Feature Information for VRF

The following

Note

One physical interface or sub-interface can be associated with one VRF only. One VRF can be associated with multiple interfaces.

As per the Multi-VRF feature, the dial-peer configuration must include the use of the interface bind functionality. This is mandatory. It allows dial-peers to be mapped to a VRF via the interface bind.

The calls received on a dial-peer are processed based on the interface to which it is associated with. The interface is in turn associated with the VRF. So, the calls are processed based on the VRF table associated with that particular interface.

Restrictions

Supports only SIP-SIP calls.

Cisco Unified Communications Manager Express (Unified CME) and CUBE co-located with VRF is not supported.

Cisco Unified Survivability Remote Si ' Remoemo ɳ

DETAILED STEPS

DETAILED STEPS

The output for command **show voip rtp connections** shows as follows:

```
Device# show voip rtp connections

 VoIP RTP Port Usage Information:
Max Ports Available: 23001, Ports Reserved: 101, Ports in Use: 2
```

Directory Number (DN) Overlap across Multiple-VRFs

Configure

IP Overlap with VRF

Generally, on a router, two interfaces cannot be configured with the same IP address. WiM

show voip rtp connections command shows a video call

As dial-peer 200 is bind to GigabitEthernet0/0/1 , the session targets configured in the server-group 1 should belong to the network which

8565723 ORG T12 g711ulaw

Creating Outbound Dial-peer:

```
Device(config)# dial-peer voice 3333 voip
Device(config)# destination-pattern 2222
Device(config-dial-peer)#
```

If an IP address is already assigned to an interface,

11F3 : 6 243854170ms.2 (*11:48:43.972

If an IP address is already assigned to an interface, then associating a VRF with interface will disable the interface and remove the existing IP address. An error message (sample error message shown below) is displayed on the

The interface used

Creating Outbound Dial-peer:

```
Device(config)# dial-peer voice 3333 voip
Device(config)# destination-pattern 2222
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# session target ipv4:11.0.0.2
```

Configuring Binding

```
Max Ports Available: 19999, Ports Reserved: 101, Ports in Use: 2
                                        Min   Max   Ports     Ports     Ports
Media-Address Range                     Port  Port  Available Reserved  In-use
```

```
No. of remote closures         : 0
No. of conn. failures           : 0
No. of inactive conn. ageouts  : 2
```

CHAPTER 21

Configuring Multi-Tenants on SIP Trunks

The

How to Configure Multi-Tenants on SIP Trunks

Configuring Multi-Tenants on SIP Trunks

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Use the following commands to configure multi-tenants:

    voice class tenant <tag> in the global configuration mode
    Once you configure the **voice class tenant <tag>** command in the global mode, the configuration will move to the **voice class tenant <tag>** submode. You can configure all the sip-specific attributes in this submode.

    voice-class sip

tenant <tag>R

Example: SIP Trunk Registration in Multi-Tenant Configuration

For SIP trunk registration, the **voice class tenant <tag>** command is

[redacted]

[redacted]

CHAPTER

Codec Preference Lists

This

T

T.38 fax, fax-passthru and

This configuration M

Troubleshooting Negotiation of an Audio Codec from a List of Codecs

Use the following commands to debug any errors that you may encounter when you configure the Negotiatio

media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
long

CHAPTER [redacted]

Transcoding

Transcoding is a process of converting one

SSPFARM profile is associated to a new application type CUBE.

Configuring SCCP-based Transcoding (ISR-G2 devices only)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-card** *voice-interface-slot-number*
4. **dspfarm tdm pooling**
5. **dspfarm services dspfarm**
6. **exit**
7. **telephony-service**
8. **sdspfarm**

```
Device(config)# voice-card 1

Device(config-voicecard)# dspfarm
Device(config-voicecard)# dsp services
```

Configuring Transrating for a Codec

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **codec** *codec-name* **bytes** *voice-payload-size* [**fixed-bytes**]
5. **end**

DETAILED STEPS

[redacted]

V

[redacted]

CHAPTER [redacted]

Restrictions

Supports only SIP-SIP calls.

Video suppression is not supported in SDP pass-through mode.

Video suppression feature removes both video and

Configuring Video Suppression

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following

	Command or Action	Purpose
	`Device (config-voi-sip)#`	

PART VI

Media Recording

CHAPTER

Any media service parameter

1 Incoming call from SIP trunk.

2 Outbound call to a Contact Centre

3 Media between endpoints

SIP Recorder Interface

SIP is used as a protocol between CUBE and the MediaSense SIP server. Extensions are made to SIP toto

DETAILED STEPS

Configuring Network-Based Recording (without Media Profile Recorder)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media class** *tag*
4. **recorder parameter**
5. (Optional) **media-type**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

Example:

Example:

```
Device# show voip recmsp session detail call-id 145
RECMSP
```

User-Agent:

.

a=fmtp:126

```
m=video 1596 RTP/AVP 126
.
.
a=fmtp:97 profile-level-id=420015
a=recvonly
m=video 1598 RTP/AVP 126
.
.
a=fmtp:126 profile-level-id=420015
a=recvonly
Sent:
ACK sip:9.45.38.39:7686;transport=UDP SIP/2.0

Via: SIP/2.0/UDP 9.41.36.41:5060;branch=z9hG4bK2CD7

From: <sip:9.41.36.41>;tag=1ECFD128-24DF

To: <sip:575757@9.45.38.39>;tag=16104SIPpTag011

Date: Tue, 19 Mar 2013 11:40:01 GMT

Call-ID: FFFFFFFF91E00FE6-FFFFFFFF8FC011E2-FFFFFFFF824DF469-FFFFFFFFB6661C06@9.41.36.41

Max-Forwards: 70

CSeq: 101 ACK

Allow-Events: telephone-event

Content-Length: 0
```

Output Field	Description
m=audio 1592 RTP/AVP 0	First m-line of recording server after it started listening.
	SeconS

```
m=audio 16392 RTP/AVP 0 19
c=IN IP4 9.41.36.15
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20
a=sendonly
m=audio 16394 RTP/AVP 0 19
c=IN IP4 9.41.36.15
a=rtpmap:0 PCMU/8000
a=rtpmap:19 CN/8000
a=ptime:20
a=sendonly
```

Response from CUBE has inactive video m-lines.

```
*Jun 15 10:37:55.406: //106/3E7E90AE8006/SIP/Info/ccsip_ipip_media_forking_Display_TDContainerData:
 recorder tag = 5
```

For Video:

Media Forking Initialized:

Additional References for Network-Based Recording

Related Documents

[redacted]

CHAPTER

SIPREC (SIP Recording)

The SIPREC (SIP

Recording is not supported if RU ɦ ᘯ

The following figure illustrates a third party recorder deployment with

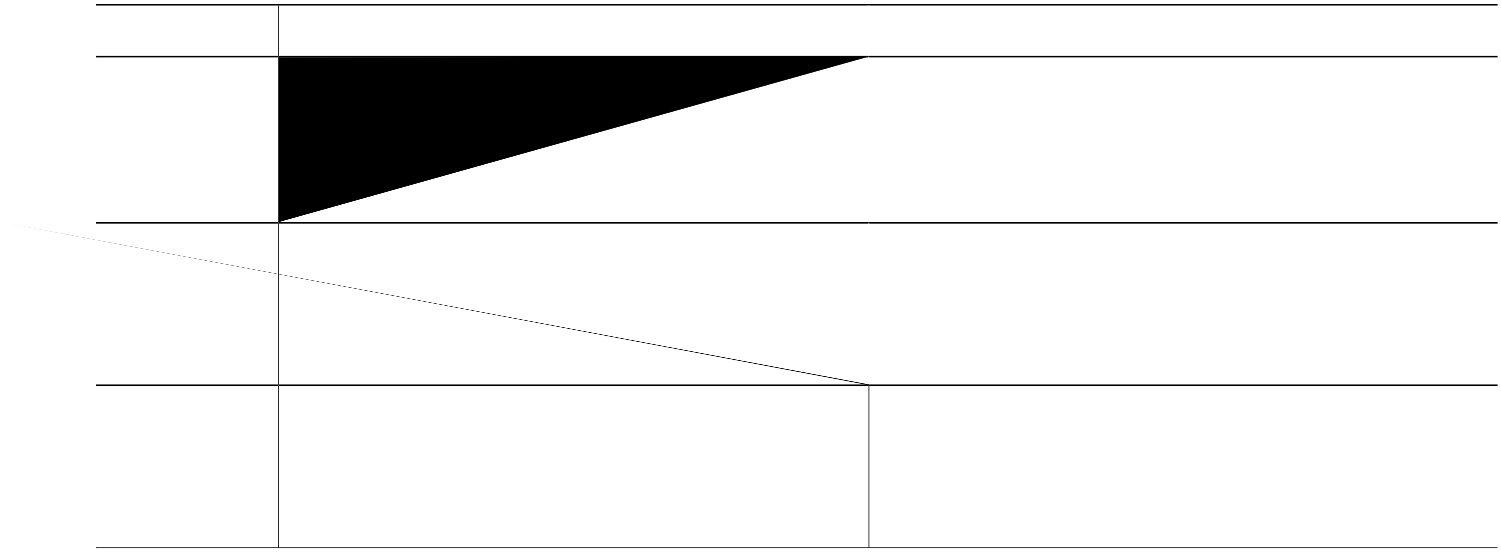
How to Configure SIPREC-Based Recording

Configuring SIPREC-Based Recording (with Media Profile Recorder)

Configuring SIPREC-Based Recording (without Media Profile Recorder)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **media class** *tag*
4. **recorder parametersiprec**
5. (Optional) **media-type audio**



Output Field	Description
	Participant CS Association class describes the association of the second participant to

```
b=TIAS:1000000
a=rtpmap:97 H264/90000
a=fmtp:97
```

```
c=IN IP4 9.42.25.149
b=TIAS:1000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0
a=inM       nl=                                         #                                Mo
```

```
a=fmtp:101 0-16
a=ptime:20
a=inactive
a=label:1
m=audio 16498 RTP/AVP 0 101
c=IN IP4 9.42.25.149
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=inactive
a=label:2
m=video 16500 RTP/AVP 97
c=IN IP4 9.42.25.149
b=TIAS:1000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0
a=inactive
a=label:3
m=video 16502 RTP/AVP 97
c=IN IP4 9.42.25.149
b=TIAS:1000000
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42801E;packetization-mode=0
a=inactive
a=label:4

--uniqueBoundary
```

```
t=0 0
m=audio 16628 RTP/AVP 8 101
c=IN IP4 9.42.25.149
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=sendonly
a=label:1
m=audio 16630 RTP/AVP 8 101
c=IN IP4 9.42.25.149
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
```

```
m=video 16636 RTP/AVP 97
c=IN IP4
```

```
a=maxptime:20
a=sendonly
a=label:1
m=audio 16650 RTP/AVP 116 101
c=IN IP4 9.42.25.149
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:20
a=sendonly
a=label:2
m=video 16652 RTP/AVP 97
c=IN IP4 9.42.25.149
```

```
c=IN IP4 9.42.25.149
t=0 0
m=audio 16648 RTP/AVP 0 101
c=IN IP4 9.42.25.149
a=rtpmap:0 PCMU/8000   000                   ŷ
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=sendonly
a=label:1
m=audio 16650 RTP/AVP 0 101c=INcP4cM
```

After transfer, participant A is disassociated from the call and participant C joins the call. This information

<participant

```
        <nameID aor="sip:7774442214@10.104.54.52">
        </nameID>
    </participant>
    <participantsessionassoc participant_id="t5nW8RM6EeWACd4iOrLrag=="
session_id="t5nW8RM6EeWACd4iOrLrag==">
        <disassociate-time
```

CHAPTER

Table 38: Feature Information for Network-Based Recording of Video Calls Using Cisco Unified Border Element

	Command or Action	Purpose
	Example: `Device(cfg-mediaprofile)# ref-frame-req`	

	Command or Action	Purpose
Step 4	**monitor-ref-frames** Example: `Device(cfg-mediaprofile)# monitor-ref-frames`	Monitors reference frames or intra-frames.
Step 5	**end** Example: `Device(cfg-mediaprofile)# end`	Exits media profile configuration mode.

[redacted]

Table 39: Feature Information for Third-Party GUID Capture for Correlation Between Calls and SIP-based Recording

Feature Name	Releases	Feature Information
		The Third-Party GUID Capture for Correlation Between Calls and SIP-based Recording feature provides support

How to Capture Third-Party GUID for Correlation Between Calls and SIP-based Recording

Verifying Third-Party GUID Capture for Correlation Between Calls and SIP-based Recording

SUMMAR

[redacted]

CHAPTER

Cisco Unified Communications Gateway Services--Extended Media Forking

The Cisco Unified Communications (UC) Services API provides a unified web service interface for the different services in IOS gateway thereby facilitating rapid service development at application servers and managed

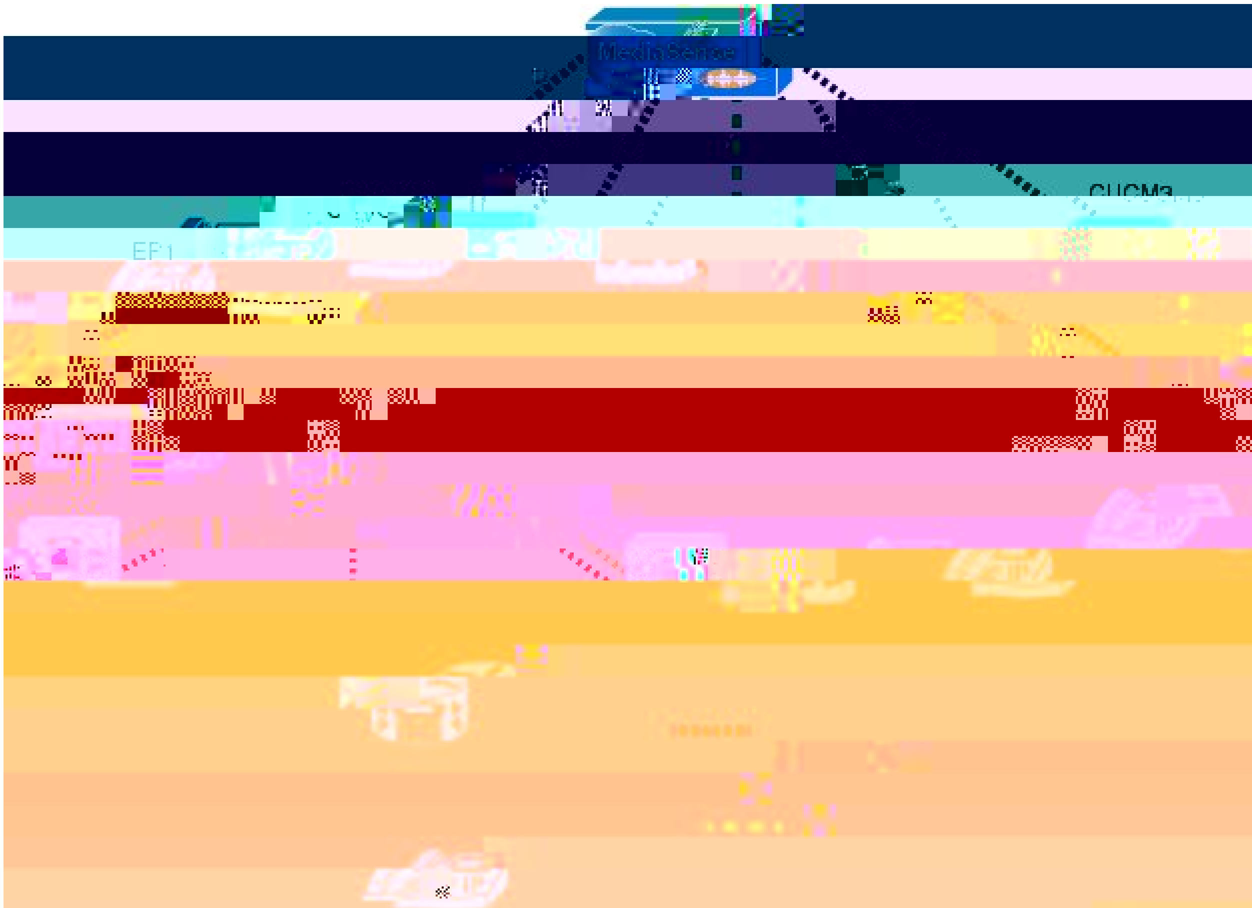
Feature Name	Releases	Feature Information
	Cisco IOS	This feature provides support for Extended Media Forking (XMF) provider to monitor calls and trigger media forking on RTP and SRTP calls.

tar

The following call

An example topology is as shown below where 4 CUCM applications are deployed. CUCM triggers media forking request to Cisco UBE. Recording tone is played to the parties involved in the call based on the recordTone parameter set in the media forking request.

Figure 26: Multiple XMF Applications and Recording Tone



Media forking can be invoked using any of the following APIs:

RequestXmfConnectionMediaForking

RequestXmfCallMediaForking

RequestXmfCallMediaSetAttributes

The recordTone parameter can

COUNTRY_SPAIN

COUNTRY_SWITZERLAND

There is no difference in the recording tone beep when any country value is chosen. Recording tone beep is played at an interval of every 15 seconds. Digital signal

	Command or Action	Purpose
		Sets the maximum number of concurrent connections to the HTTP sever that

Example:

```
Device# show call media-forking

Warning: Output may
```

Configuration Examples for UC Gateway Services

Example: Configuring Cisco Unified Communication IOS Services

The following example

PART VII

SIP Header Manipulation

CHAPTER

Example: Passing a Header Not Supported by CUBE

CUBE does not pass x-cisco-tip . However

[redacted]

CHAPTER

Copying SIP Headers

This feature shows you

	Command or Action	Purpose
Step 2	**configure terminal**	Enters global configuration mode.
	voice class sip-profiles *profile-id* Example: `Device(config)# M`	Creates a SIP profile and enters voice class configuration mode.

Given below is the original SIP

Additionally, if you would like

Table 42: Feature Information for Manipulating SIP Responses

Feature Name	Releases	Feature Information
		This

configured to copy the status

DETAILED STEPS

Modifying Status-Line Header of Outgoing SIP Response with User Defined

DETAILED STEPS

PART VIII

[redacted]

CHAPTER

Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls

The Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls feature provides dynamic payload type interworking for dual tone multifrequency (DTMF) and codec packets for Session Initiation Protocol (SIP) to SIP calls.

Based

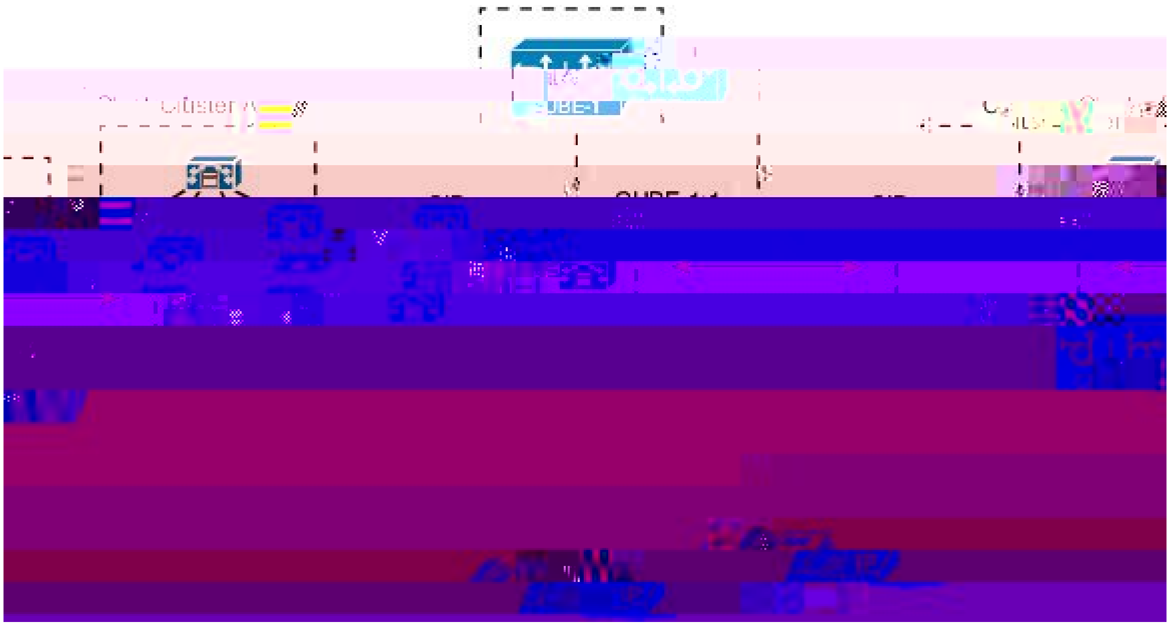
Table 43: Feature Information for Dynamic Payload Interworking for DTMF and Codec Packets Support

Asymmetric payload types are not supported on early-offer (EO) call legs in a delayed-offer

High Availability Checkpointing Support for Asymmetric Payload

High availability for a call involving asymmetric payloads is supported. In case of fail-over from active to stand-by, the asymmetric payload interworking will be continued as new active CUBE passes across the payload type values according to the negotiation and call establishment.

Figure 32: Sample High-Availability Topology



How to Configure Dynamic Payload Type Passthrough for DTMF and Codec Packets for SIP-to-SIP Calls

Configuring Dynamic Payload Type Passthrough at the Global Level

DETAILED STEPS

SUMMARY STEPS

1. **enable**
2.

SUMMARY STEPS

1. **enable**
2. **show call active voice compact**
3. **show call active voice**

DETAILED STEPS

```
 session protocol sipv2
 rtp payload-type cisco-codec-fax-ind 110
 rtp payload-type cisco-codec-video-h264 112
 session target ipv4:9.13.8.23
!
```

CHAPTER

Table 44: Feature Information for Delaye

IP2) in the below image. The RE-INVITE response is consumed

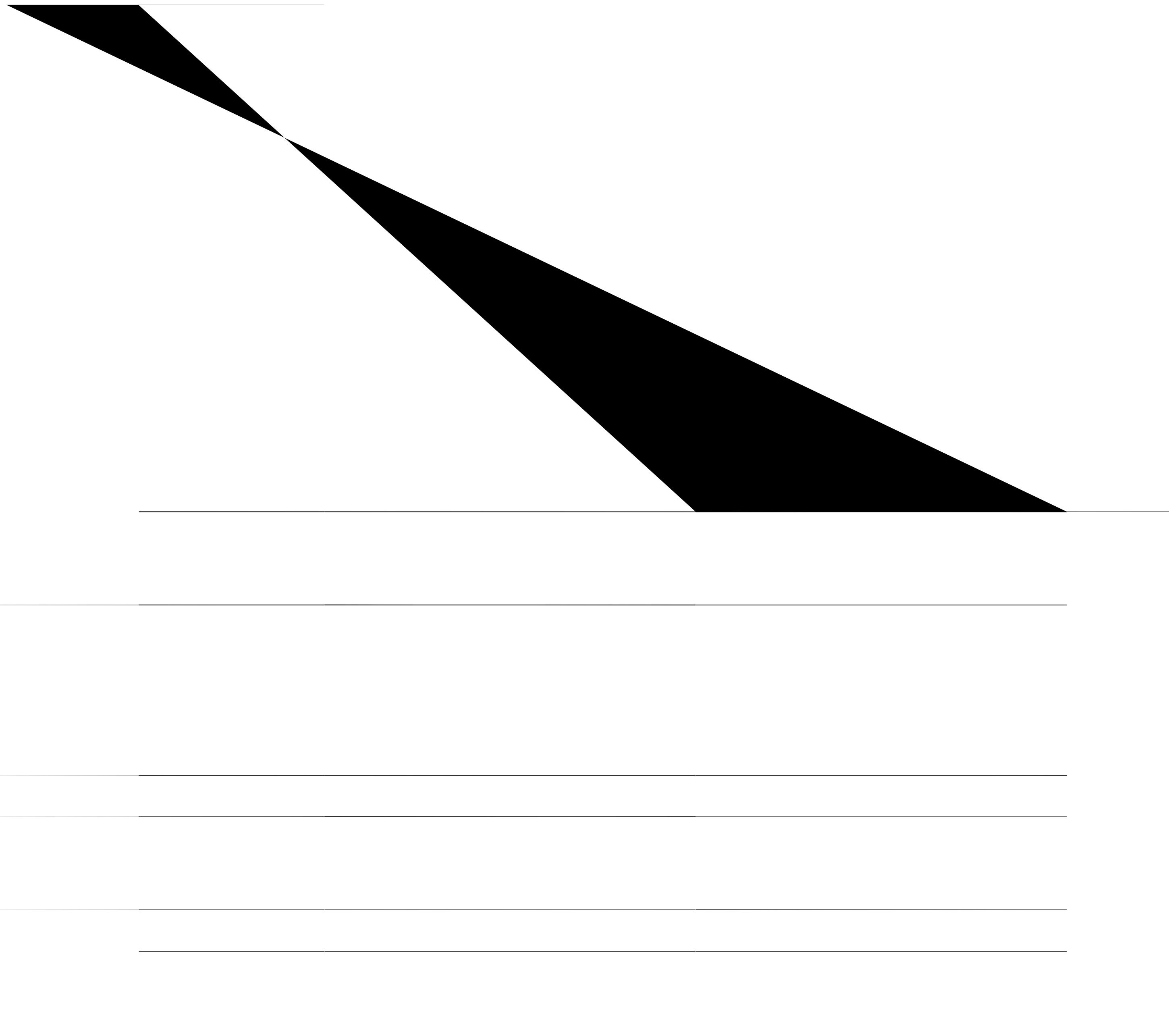
	Command or Action	Purpose
	In global VoIP SIP configuration mode **early-offer forced** Example: In dial-peer configuration mode: `Device (config)` **`dial-peer voice 10 voip`** `Device`	

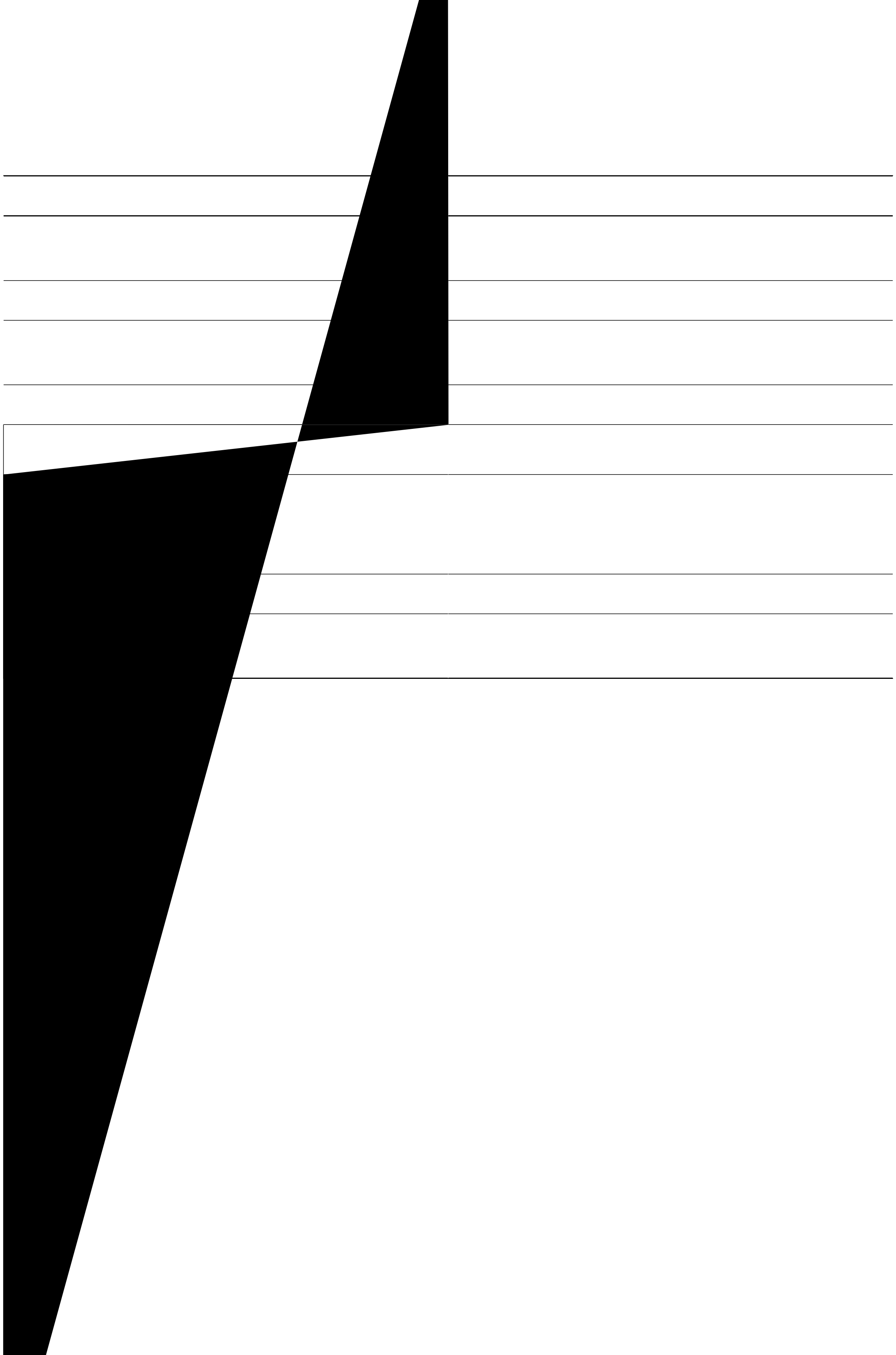
The **early-offer forced renegotiate [always]** command is used to configure this in global VoIP configuration mode (config-voi-serv) and the **voice-class sip early-offer**

Configuring Mid Call Renegotiation Support for Delayed-Offer to Early-Offer Calls

In the figure below, XIP1 is

[redacted]





Configuring H323-to-SIP Interworking

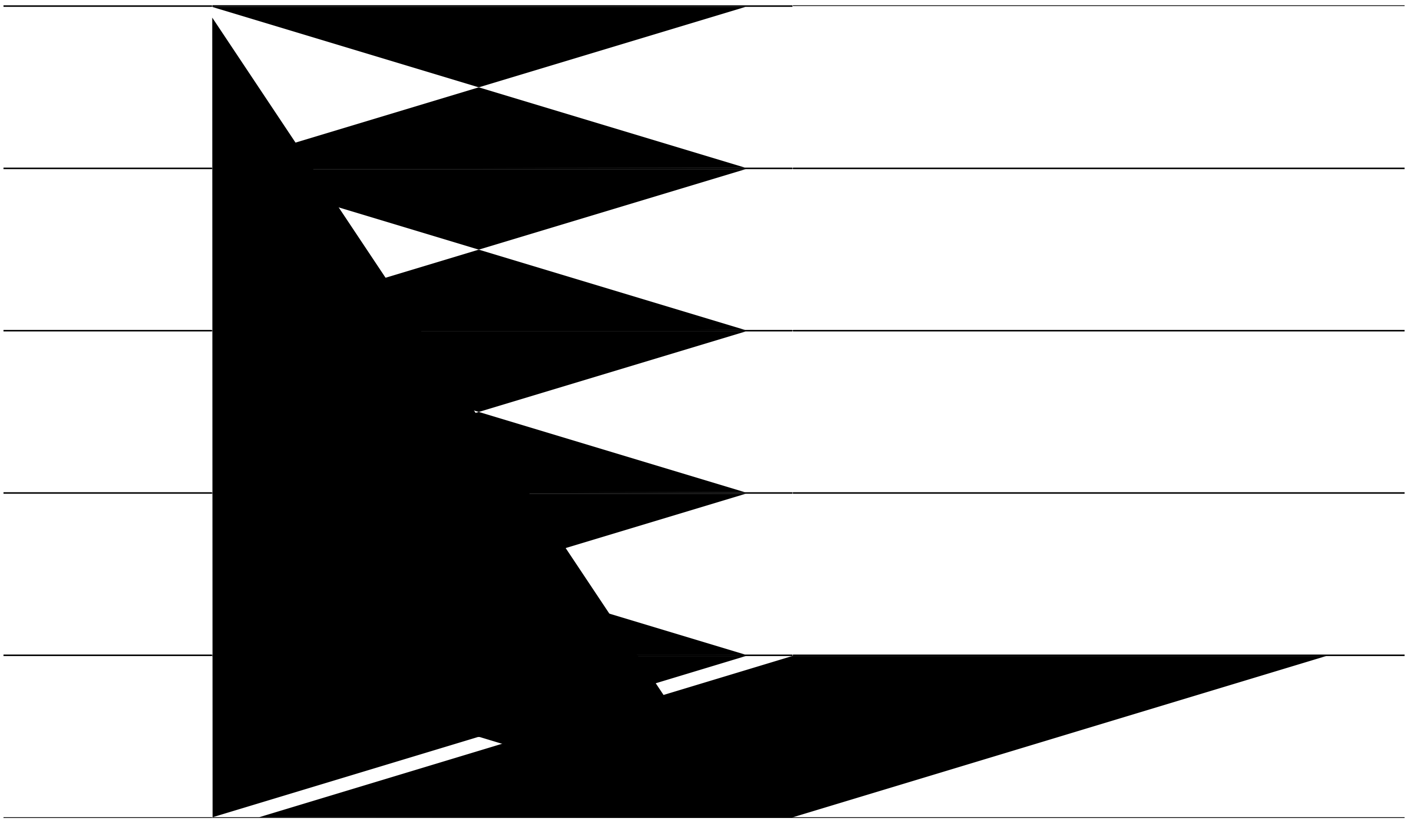
SUMMARY STEPS

1. **enable**
2. **configure terminal**

T

Prerequisites

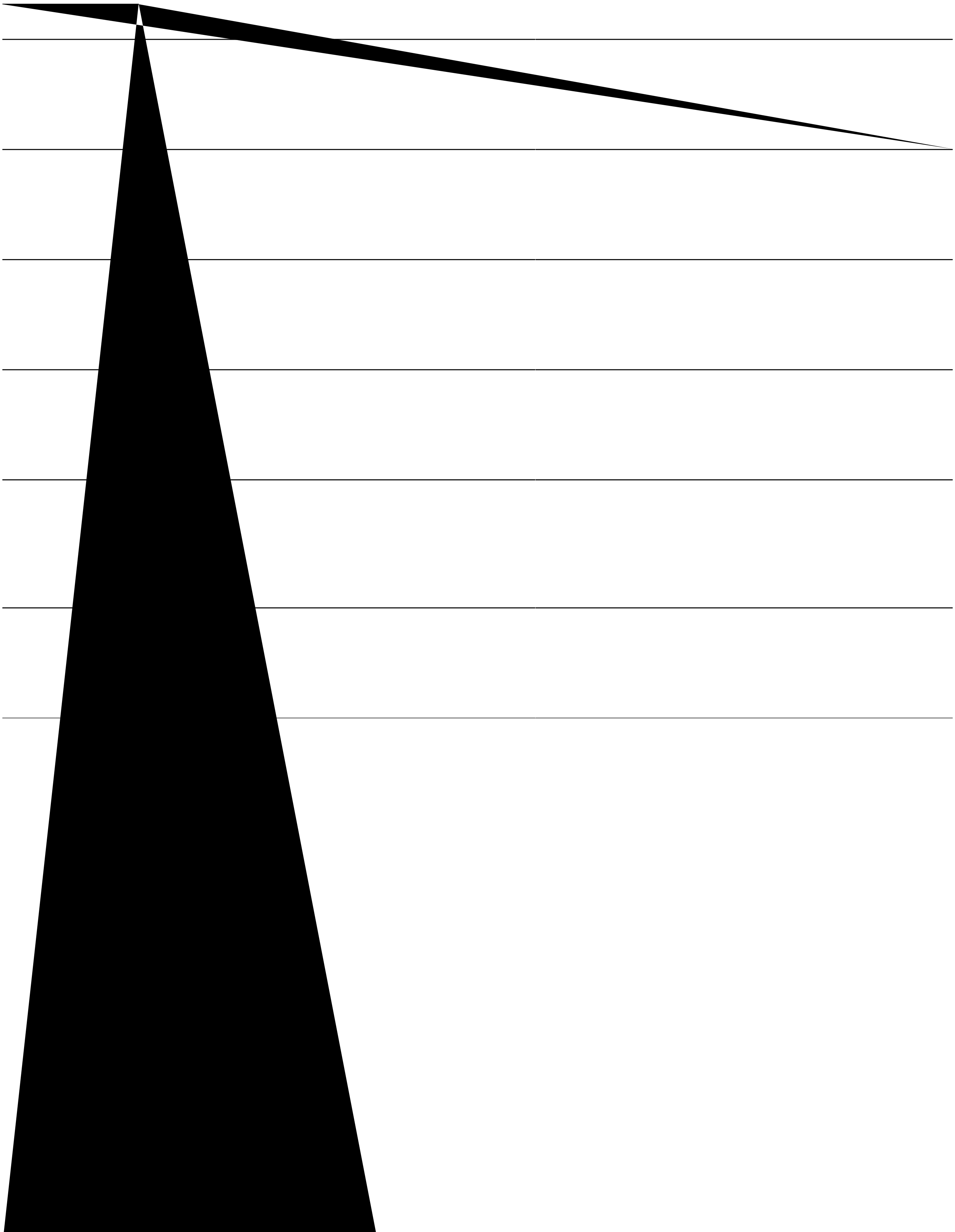
Enable CUBE



The following examples show a configuration with more reserved calls than the default

```
  ip circuit max-calls 1000
  ip circuit carrier-id AA reserved-calls 200
!
voice source-group 1
 carrier-id source AA
 carrier-id target AA
```

If there is no incoming source carrier ID, the default circuit is used because there is no match in the voice source group.



SUMMARY STEPS

1. **show call active**

T

PART X

High Availability

Route Processor Redundancy

Route Processor Redundancy (RPR) allows you to configure a standby RP. When you configure RPR, the standby RP loads the Cisco IOS

Clustering with load balancing Clustering

CHAPTER 40

DSP High Availability Support

Cisco Unified Border Element (CUBE) DSP High Availability support is added for Box-to-Box and Inbox configurations. Earlier, calls that required DSP resources were not checkpointed. As a result, both the media and signaling sessions were not preserved after switchover resulting in call failure.

> **Note** DSP HA is supported only for SIP-to-SIP calls.

[redacted]

CHAPTER [redacted]

If the Cisco UBE switchover happens at any instance, then video calls will be preserved before de-escalation and audio calls will

Note Use the same hardware for both

Enables privileged EXEC mode.

Example:

```
Device> enable
```

Step 2 **show call active voice compact**

Displays a compact version of call information

Monitoring Media Forking with High Availability

Perform this task to monitor media forking calls with high availability on active and standby Cisco UBE devices. The **show** commands can be entered in any order.

SUMMARY STEPS

1. **enable**
2. **show call active voice compact**
3. **show voip rtp connections**
4. **show voip recmsp session**
5. **show voip rtp forking**
6. **show voip rtp forking**

DETAILED STEPS

Step 1 **enable**

Enables

Displays active

Voice HA RF Client ID: 1345
Voice HA RF Client SEQ: 128
My current

debug voip rtp

```
standby 0 track 2 decrement 10
standby 0 name SB
```

Example: Configuring the Interfaces for ASR Devices

[redacted]

CHAPTER

CVP Survivability TCL support with High Availability

Call survivability features are

Prerequisites

CVP survivability TCL application is configured on incoming dial-peer

Restrictions

If there is a courtesy callback (CCB) registered with CVP, then post switchover, CCB is not supported.

PART XI

ICE-Lite Support on CUBE

[redacted]

CHAPTER [redacted]

Table 50: Feature Information for ICE-Lite Support on CUBE

High Availability Support with ICE

High availability (HA) is supported only for audio calls that use ICE.

LocalIP 10.104.45.107   port 8004 type

```
30                     RUNNING
35                     RUNNING
36                     COMPLETED
```

Step 6 **show voip ice global-stats**

The following sample output

```
CALL-ID              ICE-STATE
------------------------------
57                   RUNNING
58                   RUNNING
Total number of sessions: 2
```

Step 6 **show voip ice global-stats**

The following sample output displays the global ICE statistics.

Example:

```
Device# show voip ice global-stats

Interactive Connectivity Establishment(ICE) global stats:
Total Rx Stun BindingRequests        : 47
Total Tx Stun BindingSuccessResponses: 43
Total Tx Stun BindingErrorResponses  : 4
```

The following are the sys

004029: *Aug 8 14:25:30.876 IST:

```
nonce                              : Not Set/Present
Xormapped Address                  : Not Set/Present
Server                             : Cisco
ICE       ج Xo‗                        ١                    ج Server                        ج :
```

```
004167: *Aug  8 14:25:30.913 IST: Finger Print                  : Not Set/Present
###STUN Message structure End###

004168: *Aug  8 14:25:30.913 IST: //-1/xxxxxxxxxxxx/STUN/Detail/stunSendMsg: Sent Bind Response,
Free the transaction
004169: *Aug  8 14:25:30.913 IST: //58/91300134802E/STUN/Detail/cisco_stun_process_send_msg_event:
STUN message Sent
```

Troubleshooting ICE-Lite Support on CUBE

Y

CHAPTER

Mid-call Signaling Passthrough - Media Change

Passthrough media change method optimizes or consumes mid-call, media-related signaling within the call. Mid-call signaling changes will be passed through only when bidirectional media like T.38 or video is added. The command **midcall-signaling passthru media-change** needs to be configured

A call

Multicast Music On

	Command or Action	Purpose
	Example: In Global VoIP SIP configuration poQ d	

Configuring Mid Call Codec Preservation

This tasks disables codec negotiation in the middle of a call and preserves the codec negotiated before the call.

SUMMARY STEPS

1. **enable**
2. **configure**

[redacted]

CHAPTER

Early Dialog UPDATE Block

This feature

Table 52: Feature Information for Mid-call Signaling

	Releases	Feature Information

Important Characteristics of Early Dialog UPDATE Block

The following are a few important characteristics of Early Dialog UPDATE block:

If vcc codec is of

	Command or Action	Purpose
		Exits VoIP SIP configuration mode and enters

Information About Pass-Through of Unsupported Content Types in SIP INFO Messages

The Support for Pass-Through of Unsupported

[redacted]

CHAPTER [redacted]

Support for PAID PPID Privacy PCPID and PAURI Headers on the Cisco Unified Border Element

P-Preferred Identity and P-Asserted Identity Headers

Incoming Header	Outgoing Header	Configuration Notes
PAID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer. Note PAID headers will be given priority and RPID headers will be created using the PAID header information.
RPID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer.
PPID	RPID	Enables PAID headers on incoming dial-peer and RPID headers on outgoing dial-peer. Note PPID headers will be given priority and RPID headers will be created using the PPID header information.

Privacy

If the user is subscribed to a privacy service, the Cisco Unified Border Element can support privacy using one of the following methods:

- Using prefixes

The NGN dial plan can specify prefixes to enable privacy settings. For example, the dial plan may

Random Contact Support

The Cisco Unified Border Element can

Table 55: Feature Information for PAID and PPID Headers on Cisco Unified Border Element (CUBE)

T

Prerequisites for Support for PAID PPID Privacy PCPID and PAURI Headers on the Cisco Unified Border Element

Cisco Unified Border Element

Cisco IOS Release 12.4(22)YB or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

Cisco IOS XE Release 3.1S or a later release must

Configuring P-Header Translation on an Individual Dial Peer

To configure P-Header translation on an individual dial peer, perform the steps in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip asserted-id** *header-type*
5. **exit**

DETAILED STEPS

Configuring P-Called-Party-Id Support on a Cisco Unified Border Element

To configure P-Called-Party-Id support

	Command or Action	Purpose
		Enables the vV

Ղ

SUMMARY STEPS

1. **enable**

Configuring Random-Contact Support on a Cisco Unified Border Element

To configure random-contact

[redacted]

SUMMARY STEPS

PART XIII

SIP Supplementary Services

CHAPTER [redacted]

Dynamic Refer Handling

When

Table 57: Feature Information for Dynamic REFER Handling

This

[redacted]

Table 59: Feature Information for Cause Code Mapping

1 CUBE consumes the REFER that

Note Cause code mappings for cause code 19 and 21 require configurations mentioned in Configuring Cause Code Mapping, on page 516.

Note This mapping is only for the REFER consume scenario and not for REFER passthrough.

Configuring Cause Code Mapping

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3.

	Command or Action	Purpose
Step 5	**end** Example: `Device(config-sip-ua)# end`	Exits to privileged EXEC mode.

V

PART XIV

Call Progress Analysis

[redacted]

CHAPTER

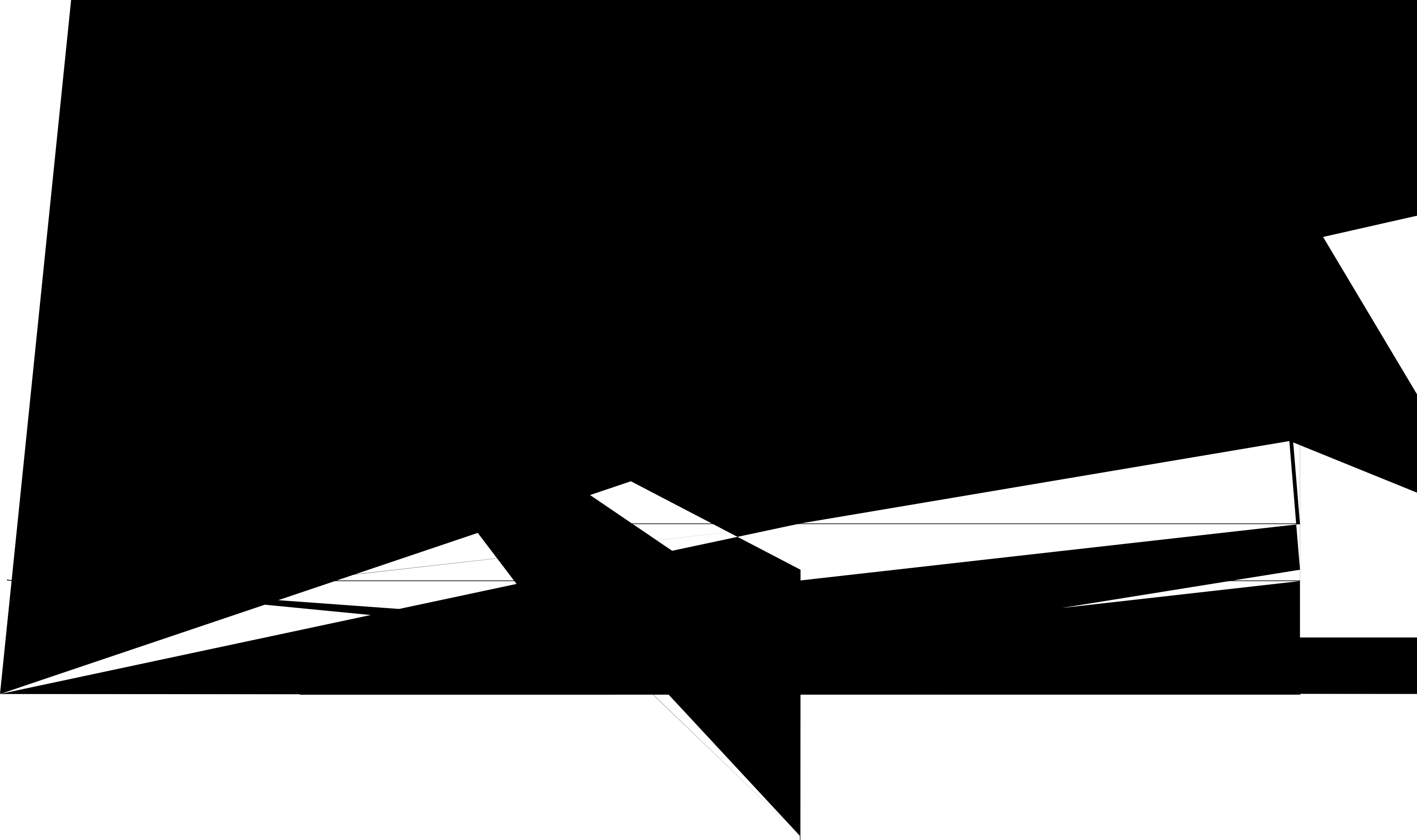
Table 61: Feature Information for Call Progress Analysis Over IP-IP Media Session

CPA call record is not supported for "180 without SDP" and "Direct Call Connect (without 18x)" call flows from Service Provider.

Information About Call Progress Analysis Over IP-IP Media Session

Call Progress Analysis

Call progress analysis (CPA) is a DSP algorithm



	Command or Action	Purpose
	cpa threshold active-signal *signal-threshold* Example:	(Optional) Sets the threshold (in decibels) of an active signal that is related to the measured noise floor level.

```
Resource Provider : FLEX_DSPRM   Status : UP
Number of Resource Configured : 4
Number of Resources Out of
```

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go

Line-Side Support for CUCM on CUBE

For an IP phone to

When Line Side Support for CUCM on CUBE is configured, predefined SIP

Login to Cisco Unified OS Administration and Security and Certificate Management, download the CUCM key (the CallManager.pem file), and copy and paste the CUCM key to

Creating a CTL File

SUMMAR

Configuring a Phone Proxy

SUMMARY STEPS

1. **voice-phone-proxy** *phone-proxy-name*
2. **voice-phone-proxy file-buffer** *size*
3. **tftp-server-address** [**ipv4** *server-ip-address* | *domain-name*]
4. **ctl-file** *ctl-filename*
5. **access-secure**
6. **complete**

DETAILED STEPS

Attaching a Phone Proxy to a Dial Peer

SUMMARY STEPS

1. **dial-peer voice** *tag* **voip**
2. **phone-proxy** *phone-proxy-name* **signal-addr ipv4** *ipv4-address* **cucm ipv4** *ipv4-address*
3. **session protocol sipv2**

Example:

```
CUBE# show dial-peer voice 5678 | section voice class sip extension

voice class sip extension = cucm,
```

Displays if **extension cucm** has been configured for the dial peer.

Example:

```
CUBE# show dial-peer voice 5678 | section voice class sip extension

voice class sip extension = none,
```

Displays if **extension cucm** has been removed for the dial peer using the **no** form of the command.

Step 3 **show dial-peer voice**

Example:

```
Device# show dial-peer
```

```
CAPF sessions: 0
Config status: complete
SIP dial-peers associated:
   Name
   ---------------
   3
   dialpeer4
-------------------------------------------------------------
```

Step 5 **show voice class phone-proxy sessions**

Example:

```
Device# show voice class phone-proxy sessions

Phone-Proxy 'phone_proxy_ipad':
                    Source                             Destination
 ------------------------------ Sessions of Dial-peer 5             -------------------------------
|Ac      --  peer
```

Example: Configuring a Phone Proxy

```
Device(config)# crypto pki certificate chain ccm1
Device(config)# certificate ca 55C2FCBFBAC552B7C6CED497D4AD33F8
[Certificate
```

```
ipv4 172.18.110.120 port 8443
Device(config-phone-proxy)# service-map server-addr ipv4 10.50.209.215 port 8080 acc-addr
ipv4 172.18.110.120 port 8080
Device(config-phone-proxy)# service-map server-addr ipv4 10.50.209.215 port 3804 acc-addr
ipv4 172.18.110.120 port 3804
Device(config-phone-proxy)# complete

Device(config)# voice-phone-proxy tftp-address ipv4 10.50.209.100
Device(config-phone-proxy)# port-range 40000 50000
Device (Config)# voice-phone-proxy tftp-address ipv4 172.18.110.120
Device(config-phone-proxy)# port-range 40000 50000
Device(config-phone-proxy)# voice-phone-proxy file-buffer size 60
```

Attaching fiM Ω

Attaching Phone Proxy

[redacted]

CHAPTER

Cisco Unified Communications Manager Express for call processing

Cisco Unity® Express hardware module and licenses for voicemail, integrated messaging, and interactive voice response.

Digital signal processors

Geographic Redundancy

Scenarios Covered   Box-to-Box and

CUBE Licensing FAQs

Q. Is CUBE Licensing enforced?

No

PART XVII

Security

CHAPTER

Feature Name	Releases	Feature Information

a third party entity. When a call is made, a TLS handshake is initiated between CUCM and CUBE, and the IOS PKI infrastructure is used to exchange certificates signed

How to Configure SIP TLS Support on CUBE

Configuring SIP TLS on CUBE

SUMMARY STEPS

1. **enable**
2. **configur**

		Purpose

	Command or Action	Purpose
Step 19	**voice service** {**pots**| **voatm** |**vofr**|**voip**} Example: `Router(config)# voice service voip`	Specifies a voice encapsulation type and enters voice service VoIP configuration mode.
		Enters this command in SIP configuration mode to enable the TLS portcor ati

to overcome this error

Configuration Examples for SIP TLS Support on CUBE

Example: SIP TLS Support on CUBE

```
show running-config
Building configuration...



Current configuration : 10894 bytes
!
! Last configuration change at 23:19:20 IST Wed Aug 19 2015
! NVRAM config last updated at 20:25:45 IST Tue Aug 18 2015
!
version 15.6
service timestamps debug datetime msec localtime show-timezone
service timestamps log datetime msec localtime show-timezone
```

```
subject-name cn=plutododsn
revocation-check none
rsakeypair selfsign
!
crypto pki trustpoint ccm155RSA
enrollment terminal
revocation-check none
!
!
crypto pki certificate chain ecdsacert1
certificate
```

```
crypto pki certificate chain ccm155RSA
certificate ca 4E23E56C7339CC679FD444D77F7A463F
  308203AB 30820293 A0030201 0202104E 23E56C73 39CC679F D444D77F 7A463F30
  0D06092A 864886F7 0D01010B 0500306A 310B3009 06035504 06130249 4E310E30
  0C060355 040A0C05 63697363 6F310D30 0B060355 040B0C04 73727467 31143012
  06035504 030C0B50 4C55544F 2D435543 4D313112 30100603 5504080C 096B6172
  6E617461 6B613112 30100603 5504070C 0962616E 67616C6F 7265301E 170D3135
  30383034 31333431 35315A17 0D323030 38303231 33343135 305A306A 310B3009
  06035504 06130249 4E310E30 0C060355
```

```
shutdown
!
interface GigabitEthernet0/0
ip address
```

```
 transport tcp tls v1.2
connection-reuse
crypto
```

PART XVIII

Voice Quality in CUBE

[redacted]

CHAPTER

CUBE Call Quality Statistics Enhancement

Call quality statistics in CUBE, such as packet loss, jitter, and round trip delay can be added to the call detail record (CDR), and these voice metrics can be calculated in IOS. For more information, refer to Voice Quality Enhancements on Cisco Unified Border Element.

The call quality statistics feature

Table 67: Feature Information for Call Quality Statistics Enhancement

Feature Name	Releases	Feature Information
		Call quality statistics in CUBE, such as packet loss, jitter, and round trip delay can be

Troubleshooting Call Quality Statistics

Use the following

PART XIX

Serviceability

WORD can be complete session identifier

.
.
SessionIDLocaluuid=4fd24d9121935531a7f8d750ad16e19

```
SCCP call-legs: 0
```

PART XX

Appendixes

[redacted]

Related Topic	Document Title
Related Application Guides	*Cisco Unified Communications Manager and Cisco IOS Interoperability Guide* *Cisco IOS SIP Configuration Guide* Cisco Unified Communications Manager (CallManager) Programming Guides
	Cisco IOS Debug Command Reference, Release 15.3. *Tr*

MIBs

Technical Assistance

[redacted]

CHAPTER

Glossary

Glossary