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CHAPTER 1

Read Me First

Important Information about Cisco IOS XE 16

Effective Cisco IOS XE Release 3.7.0E (for Catalyst Switching) and Cisco IOS XE Release

[redacted]

[redacted]

CHAPTER [redacted]

Supported Platforms

CUBE is supported on various platforms running on Cisco IOS Software Releases and Cisco IOS XE Software Releases.

[redacted]

For informationM

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

For more information M infUrimied ʅ Ƃ Ƕ

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

Information about Virtual CUBE Support on Cisco CSR 1000V Series Routers

High Availability

Virtual CUBE uses Redundancy Group infrastructure for HA. HA is between two virtual CUBE CSR instances running on either the same host or across different hosts connected through a switch. Geographic stateful switchover is not supported.

Figure 7: Virtual CUBE High A

ESXi provides the virtual switch (vSwitch) functionality where it routes traffic internally between virtual machines and link to external networks.

Figure 8: vNICs Mapped to Cisco CSR 1000V Router Interfaces

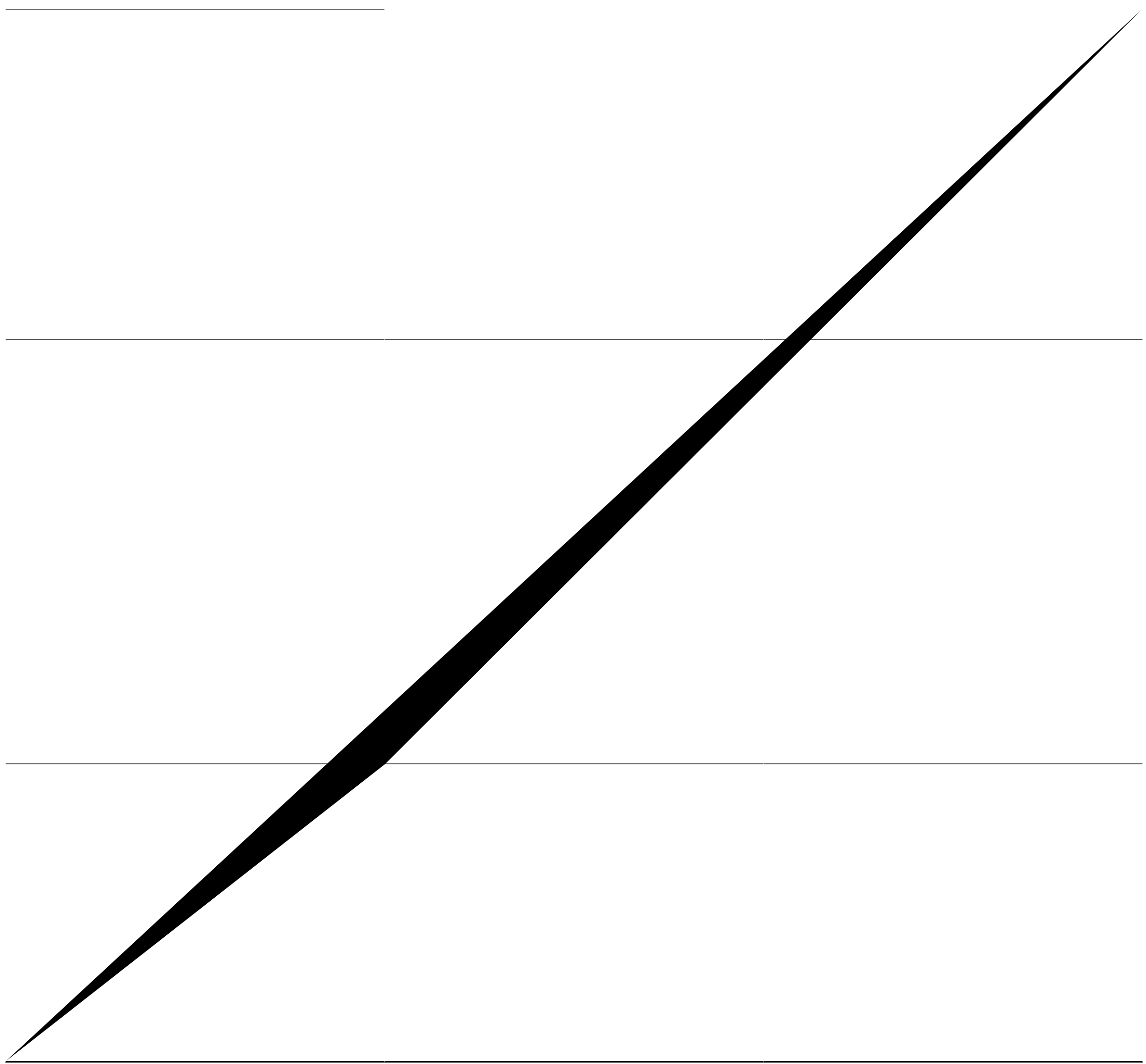
How to Enable Virtual CUBE on Cisco CSR 1000V Series Router

For details

CHAPTER

In CUBE, dial peers can also be classified as

A W



answer-address *ANI-string*

The

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, or for other types of manipulation. They are processed differently

DTMF relay prevents loss of integrity of DTMF digits caused by VoIP compressed codecs. The relayed DTMF is then regenerated transparently on the peer side.

Figure 12: DTMF Relay Mechanism

DTMF relay mechanisms supported on VoIP dial-peers are listed below based on the keywords used to configure them. The DTMF relay mechanism can be either out-of-band (H.323 or SIP) or inband (RTP).

h245-alphanumeric and h245-signal These two methods are available only on H.323 dial peers. This is an out-of-band DTMF relay mechanism that transports the DTMF

Payload type 96 and 97 are used for fax by default in

sip-notify

sip-kpml

sip-info

rtp-nte [digit-drop]

ciso-rtp

Multiple DTMF methods

H.323 gateways

Table 9: RTP-RTP Flow Around

Verifying DTMF Relay

SUMMARY STEPS

1. **show sip-ua calls**

```
Total SIP call legs:2, User Agent Client:1, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID                : 
```

```
   Calling Number          : 2017
   Called Number           : 1011
   CC Call ID              : 251
No.         Timestamp         Digit           Duration
=======================================================

Call 2
SIP Call ID                : 29BB98C-F01311E3-8297DE9F-78C438FF@10.86.176.119
   Call:
```

[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 show how codec negotiation is performed on CUBE. Two 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

[redacted]

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

Interface State	Result Using bind all or bind control Commands
	The call becomes a one-way call with media flowing in only one direction. The media flows from the gateway where the change [illegible]

Example:

```
Device# show
```

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
```

Connect Time

Table 22: Feature Information for Configuring Path of Media

CHAPTER [redacted]

SIP Profiles

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 20: SIP Profile

In order to customize SIP messaging in both directions, you can place and configure a CUBE with a SIP profile at the boundary of these networks.

In addition to network policy compliance, the CUBE SIP profiles can be used to resolve incompatibilities between SIP devices inside the enterprise network. These are the situations in which incompatibilities can arise:

- A device rejects an unknown header

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 class sip-profiles 10
Device(config-class)# request INVITE sdp-header Session-Owner modify
"CiscoSystems-SIP-GW-UserAgent  "-"
```

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

```
Device(config)# voice class sip-profiles 10
Device(config-class)# rule 1 request INVITE sdp-header Session-Owner modify
"CiscoSystems-SIP-GW-UserAgent  "-"
```

Convert "sip uri" to "tel uri" in Req-URI, From and To Headers of SIP INVITE Request Messages

For example, modify sip:2222000020@9.13.24.6:5060

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 **show running-config** command showing the SIP profiles in non-rule format:

```
Device#show running-config
!
```

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                              -.
```

Does not provide support for IPv4-IPv6 interworking cases with or without ANAT because Cisco UBE cannot operate in FA mode post tromboning.

Information About VoIP for IPv6

the SIP session is still active. Two header fields can be defined: Session-Expires, which conveys the lifetime of the session, and Min-SE, which conveys the minimum allowed value for the session timer.

For more information, refer to the SIP Session Timer Support section in the Cisco Unified Border Element SIP Support Configuration Guide.

Media Flow-Through (FT): In a media flow through mode, between two endpoints, both signaling and media flows through the IP-to-IP Gateway (IPIP GW). The IPIP GW performs both signaling and media interworking between H.323/SIP IPv4 and O

Flow-around Cisco UBE neither plays a part in mediP

RE-INVITE Consumption: The Re-INVITE/UPDATE consumption feature helps to avoid interoperability

Configurable Error Response Code in OPTIONS Ping: Cisco UBE provides an option to configure the error response code when a dial peer is busied out because of an Out-of-Dialog OPTIONS ping failure.

For more information, refer

A Cisco Unified Border Element

	Command or Action	Purpose
		Shuts down or enables VoIPv6 for the selected

CC Call ID

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **protocol mode** {**ipv4** | **ipv6** | **dual-stack** { {**ipv4** | **ipv6**
5.

DET0U

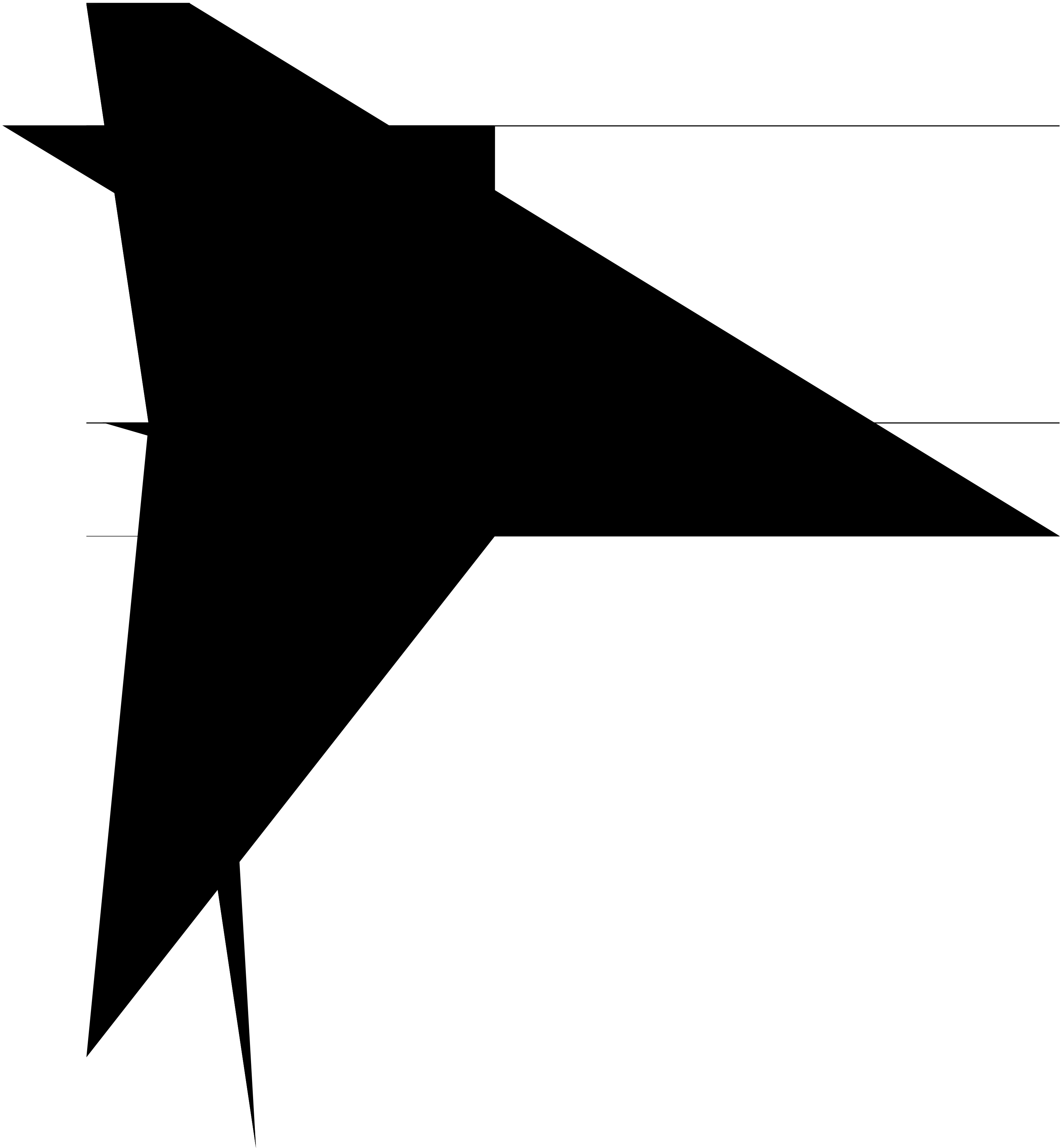
DETAILED STEPS

Configuring the SIP Server

SUMMARY STEPS

1. **enable**
2.

	Command or Action	Purpose
		Enables SIP gateways to register E.164 numbers



DETAILED STEPS

Configuring the RTP Port Range for an Interface

SUMMARY STEPS

1. **enable**
2.

Configuring Message Waiting Indicator Server Address

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4**: **destination-address** | **ipv6**: **destination-address** | **dns**: **host name**} **peer-tag** [**output-dial-peer-tag**]
5. **end**

DETAILED STEPS

Configuring Voice Ports

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port number*
4. **vmwi** [**fsk** | **dc-voltage**]
5. **end**

DETAILED STEPS

Configuring Cisco UBE Mid-call Re-INVITE Consumption

Configuring Passthrough of Mid-call Signalling

Perform this

performs both signaling and media interoperation between H.323/SIP IPv4 and SIP IPv6 networks (see the figure below).

Figure 25: IPv4 to IPv6 Media Interoperating Through Cisco IOS MTP

The Cisco UBE feature adds IPv6 capability to existing VoIP features. This feature adds dual-stack support on voice gateways and MTP, IPv6 support for SIP

Media Flow-Around

To enable all Session Initiation Protocol (SIP)-related debugging,

```
Port range not configured, Min: 16384, Max: 32767
                                                Ports        Ports        Ports
```

```
a=fmtp:18 annexb=no
a=rtpmap:19 CN/8000
a=ptime:20
```

```
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: <sip:6000@9.44.30.11>;party=called;screen=no;privacy=off
Contact: <sip:6000@9.44.30.11:5060>
```

Example:

```
Device# show
```

Example:

```
Device# debug ccsip messages
```

The

```
Call-ID: FB05CC74-B08E11E1-82C1F4DD-5665AA1B@2001:DB8:C18:2:223:4FF:FEAC:4540
Timestamp: 1339152794
CSeq: 101 INVITE
Allow-Events: telephone-event
Server: Cisco-SIPGateway/IOS-15.2.2.5.T
```

```
a=rtpmap:19 CN/8000
```

Allow-Events:

Feature Name	Releases	Feature Information
		The

PART II

Dial Peer Enhancements

[redacted]

CHAPTER

Matching Inbound Dial Peers by URI

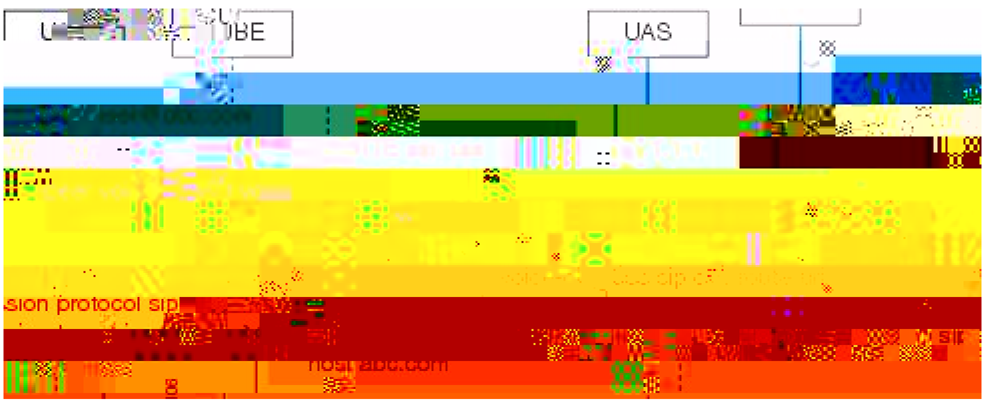
The Matching Inbound Dial Peers by URI feature allows you to configure selection of inbound dial peers by matching parts of the URI sent by remote (neighboring) SIP entity. The match can be done on different parts of the URI like hostname, IP address, DNS name. This feature M ch

DETAILED STEPS

SIP 2

```
dial-peer voice 101 voip
 session protocol
```

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**
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 25: 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

[redacted]

CHAPTER

Inbound Leg Headers for Outbound Dial-Peer Matching

The Inbound

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**

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)# dial-peer voice 100 voip
Device(config-dial-peer)# destination provision-policy 200
Device(config-dial-peer)# end

Device(config)# voice class uri 200 sip
Device(config-voice-uri-clas)# pattern 25054..

!Associates a Provision Policy with an Outbound Dial Peer.
The FROM SIP headers of the inbound leg is matched to select the below outbound dial peer.
Device(config)# dial-peer voice 200 voip
Device(config-dial-peer)# destination uri-from 200
Device(config-dial-peer)# end
```

[redacted]

CHAPTER [redacted]

Server Groups in Outbound Dial Peers

This feature configures a server group (group of server addresses) that can be referenced from an

Information About Server Groups in Outbound Dial Peers

You can now group

```
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

Table 30: Feature Information for Domain-Based Routing Support on the Cisco UBE

Feature Name	Releases	Feature Information
		The

With the introduction of the domain-based routing feature,

DETAILED STEPS

Example:

```
Device> enable
```

Step 2 **debug ccsip all**

Enables all SIP-related debugging.

Example:

```
Device# debug
```

```
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1297340108
```

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

Both the target

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

CHAPTER

Feature Information for VRF

The following table provides release

Information About Multi-VRF

The Multi-VRF feature allows you to configure and maintain more than one instance

Recommendations

For new deployments, we recommend a reboot of the router once all VRFs' are configured under interfaces.

DETAILED STEPS

	Command or Action	Purpose
	Example: At dial-peer level: `Device(config)#`**`dial-peer voice 1111 voip`**	

	Command or Action	Purpose
	CUBE supports a maximum of 54 VRFs. Hence, you can configure up to 54 mediM	

The following is example shows

```
Device(config-dial-peer)# video codec h264
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# session target ipv4:10.0.0.1
Device(config-dial-peer)# voice-class sip bind
```

Configure GigabitEthernet 0/1 that belongs

Using Server Groups with VRF

Whenever destination server group is used with VRF

To overcome this issue, the inbound dial-peers are filtered based on the incoming VRF and then followed by the regular inbound dial-peer matching. Now, the response is sent to the same VRF on which the request was received.

Consider the following configuration example output to understand the inbound dial-peer matching criteria used i ethei e

Prior to Cisco IOS 15.6(3)M and Cisco IOS XE Denali 16.3.1 releases, when an incoming call is received for the dialed number sending a query to the VRF name server. All IP addresses obtained from a VRF-specific name cache

Configuring VRF

```
Device# enable
Device# configure terminal
```

2222 voip up

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

long duration call detected:<y/n> long duration call duration :<sec> timestamp:<time>
LostPacketRate:<%> OutOfOrderRate:<%>
VRF:<%>
 MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
  last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
 <codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
 <codec> (payload size)
Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l>
dBm
 MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
        speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt:
```

```
<callID>  A/O  FAX T<sec>  Codec       type  Peer Address  IP
```

```
Device(config)# allow-connections sip to sip
Device(config)# redundancy-group 1
Device(config)# sip

Device(config)# redundancy
Device(config)# mode none
Device(config)# application redundancy
Device(config)# group 1
Device(config)# name raf-b2b
Device(config)# priority 1
Device(config)# timers
```

Example: Configuring HSRP High Availability with VRF

Below configuration example is applicable for

If an IP address is already assigned to an interface, then associating a VRF with interface will disable

The interface used fo M fo M fo M

Max Ports

No. of remote closu

[redacted]

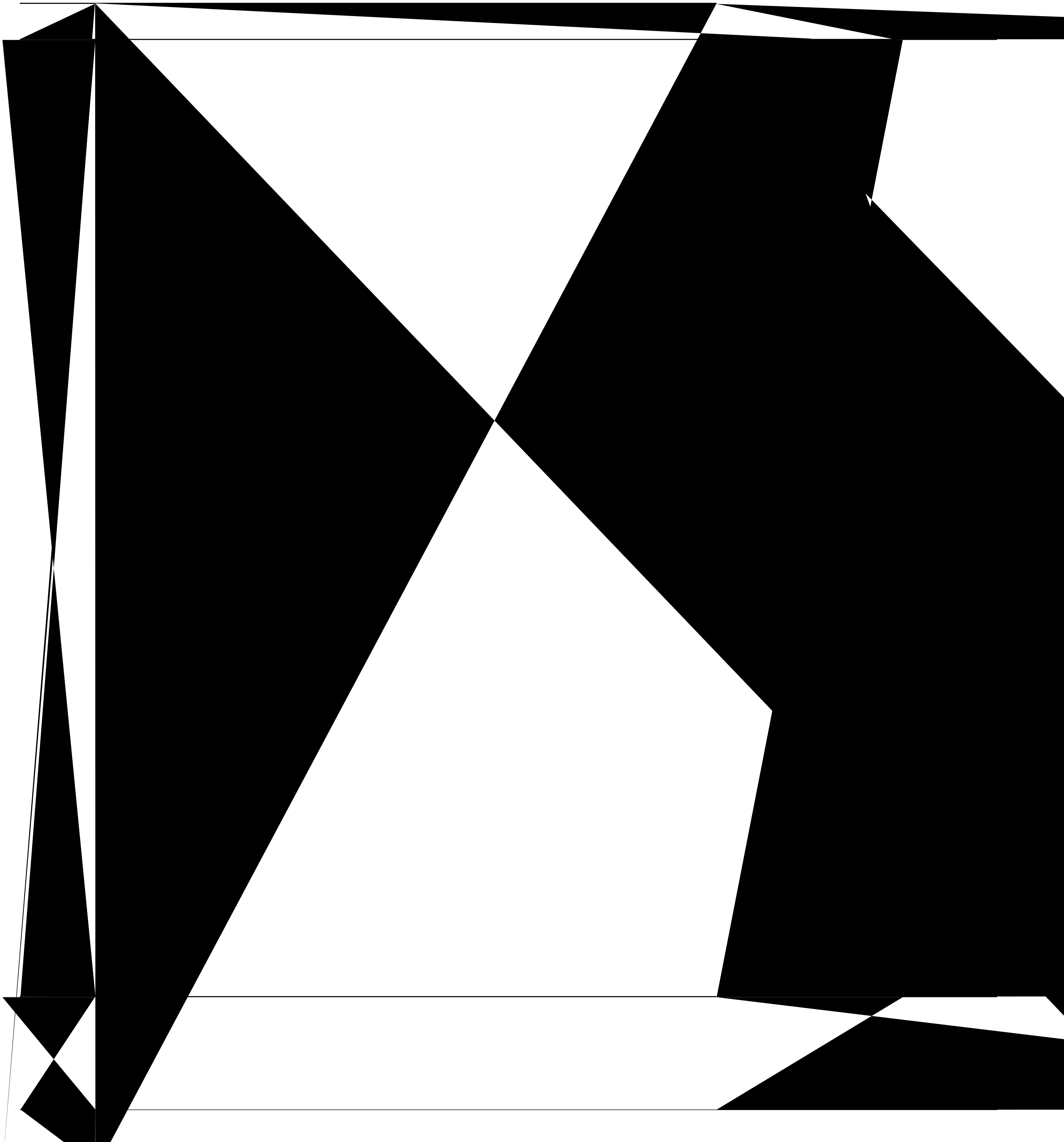
CHAPTER

Configuring Multi-Tenants on SIP Trunks

The feature

Information About Configuring Multi-Tenants on SIP Trunks

In the previous releases of Cisco IOS



PART IV

Codecs

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 duration call detected:<y/n> long duration call duration
```

[redacted]

CHAPTER

Transcoding

Transcoding is a process of converting one voice codec to another. For example, transcoding iLBC-G.711 or iLBC-G.729.

LTI based T

DSPFARM profile is associated to SCCP using the following commands:

```
Device(config)# voice-card 0/1
Device(config-voicecard)# dspfarm
Device(config-voicecard)# dsp services dspfarm
Device(config-voicecard)# exit

! Configuring dspfarm profile
Device(config)# dspfarm profile 1 transcode
Device(config-dspfarm-profile)# codec g711ulaw
Device(config-dspfarm-profile)# codec g711alaw
Device(config-dspfarm-profile)# codec g729r8
Device(config-dspfarm-profile)# maximum sessions 10

Device(config-dspfarm-profile)# associate application CUBE

Device(config-dspfarm-profile)# exitngine


Device(config)# interface ServiceM                    ١                                ى
```

```
 revocation-check none
 rsakeypair CUBE


!Authenticate the
```

```
associate application CUBE
```

Configuring SCCP-based T

Configuring Secure Transcoding

Configuring the Certificate Authority

Perform the steps described in this section to configure the certificate authority.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip http server**
4. **crypto pki server** *cs-label*
5. **database level**

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **crypto trustpoint** *name*
4. **enrollment url** *url*
5. **serial-number**
6.

Before You Begin

Before you register the secure universal transcoder to the Cisco Unified Border Element, you should associated SCCP to the secure DSPFARM profile, as described in the Associating SCCP to the Secure DSPFARM Profile, on page 326

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]

CHAPTER

Call Progress Analysis Over IP-to-IP Media Session

The Call

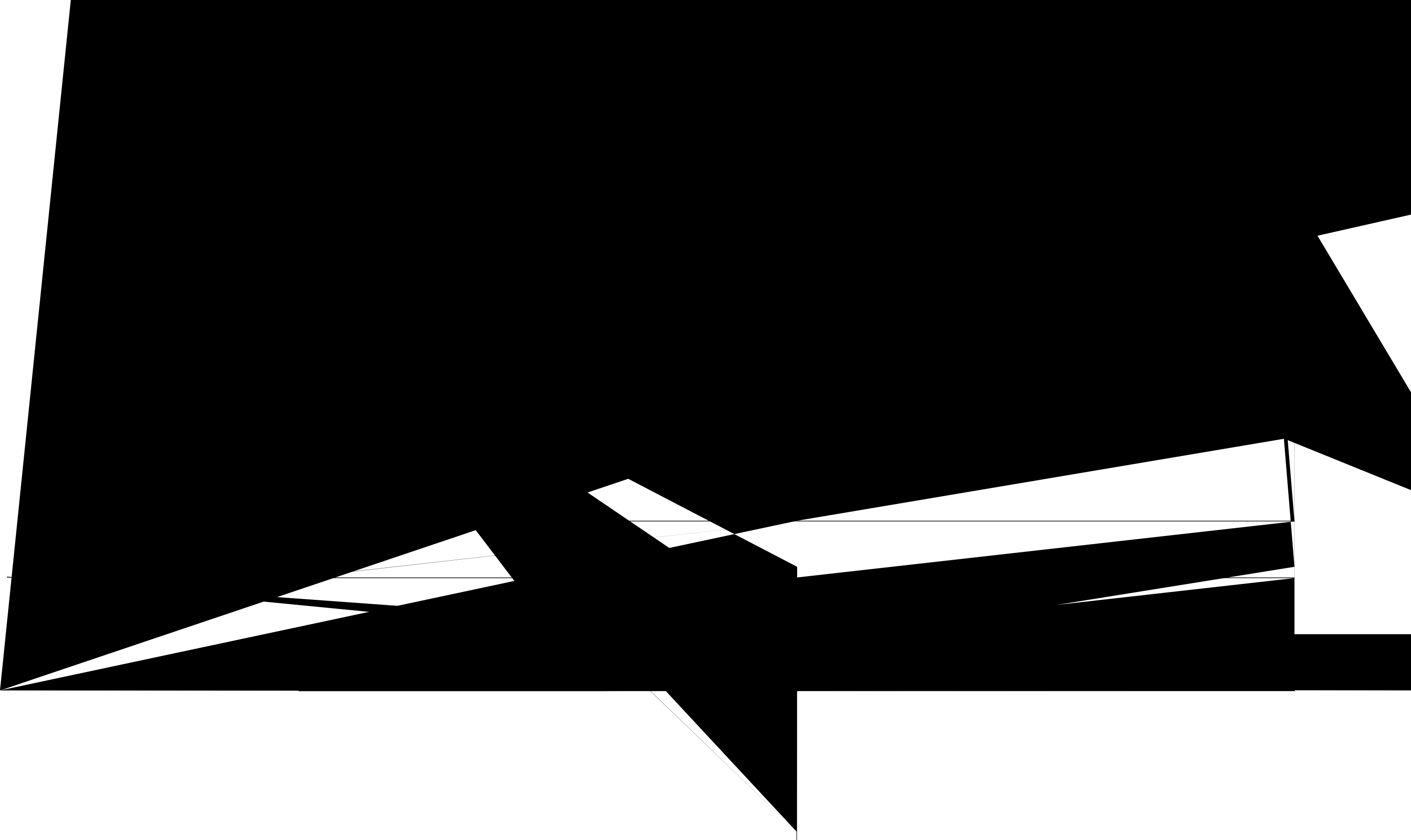
Table 37: Feature Information for Call Progress Analysis Over IP-IP Media Session

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 that analyzes the Real-Time Transport Protocol (RTP) voice stream



	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
```

PART VI

Video

[redacted]

CHAPTER [redacted]

Video Suppression

Video suppression feature allows pass-through of only audio and

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

PART VII

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	
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:

[redacted]

CHAPTER

SIPREC (SIP Recording)

The SIPREC (SIP Recording) feature supports media recording for Real-time Transport Protocol (RTP) streams in

Recording is not supported if RU	ɦ	Ω

The following figure illustrates a

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
a=ptime:20
a=sendonly
a=label:2

--uniqueBoundary
Content-Type: application/rs-metadata+xml
Content-Disposition: recording-session

<?xml version="1.0" encoding="UTF-8"?>
<recording xmlns="urn:ietf:params:xml:ns:recording:1">

    <stream stream_id="evyS5/1CEeSBOKsYHx7YVg==" session_id="evv2v/1CEeSBM6sYHx7YVg==">
        <label>1</label>
    </stream>
```

```
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 participant_id="vm+z2xM6EeWAIN4iOrLrag==">
        <nameID aor="sip:7774442214@10.104.54.52">
        </nameID>
    </participant>
    <participantsessionassoc participant_id="vm+z2xM6EeWAIN4iOrLrag=="
session_id="vACJ+xM6EeWAF94iOrLrag==">
        <associate-time>2015-06-16T08:44:32.869Z</associate-time>
    </participantsessionassoc>
    <participant participant_id="vm+z2xM6EeWAIN4iOrLrag==">
        <nameID aor="sip:7774442218@10.104.54.52">
        </nameID>
    </participant>
    <participantsessionassoc participant_id="vm+z2xM6EeWAIN4iOrLrag=="
session_id="vACJ+xM6EeWAF94iOrLrag==">
        <associate-time>2015-06-16T08:44:32.869Z</associate-time>
    </participantsessionassoc>
    <participant
```

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

Table 42: 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 43: 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

To capture the third-party GUID and forward it to the recording server, you need to copy a

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

SUMMAR

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

tar

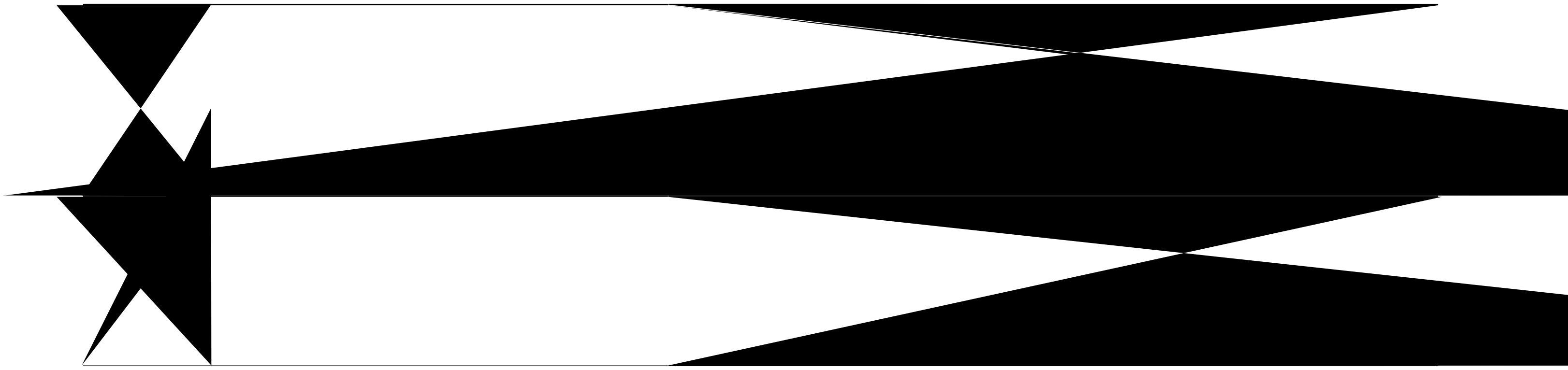
The following call

An example topology is as shown below where 4 CUCM applications are deployed. CUCM

COUNTRY_SPAIN

COUNTRY_SWITZERLAND

There is no difference in



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

CHAPTER

Example: Passing a Header Not Supported by CUBE

CUBE does not pass x-cisco-tip . However

CHAPTER [redacted]

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.

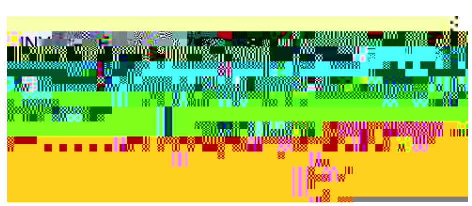
Given below is the original SIP message, where the INVITE has a non-routable value of 43565432A5. The actual phone destination number is 25555552 and is present i ¬ P To: SIP header.

Figure 35: Incoming SIP Message

Given below is the SIP message that is required. Note that 43565432A5 has changed to 25555552 in the SIP INVITE.

Figure 36: Modified SIP Message

Because CUBE is a back-to-back user agent, the incoming dial peer is matched to the outgoing dial peer. The SIP Profile configured below



Additionally, if you would like

[redacted]

CHAPTER

Manipulating SIP Status-Line Header of SIP Responses

The SIP status line is a SIP

Table 46: 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

CHAPTER

Table 47: Feature Information for Dynamic Payload Interworking for DTMF and Codec Packets Support

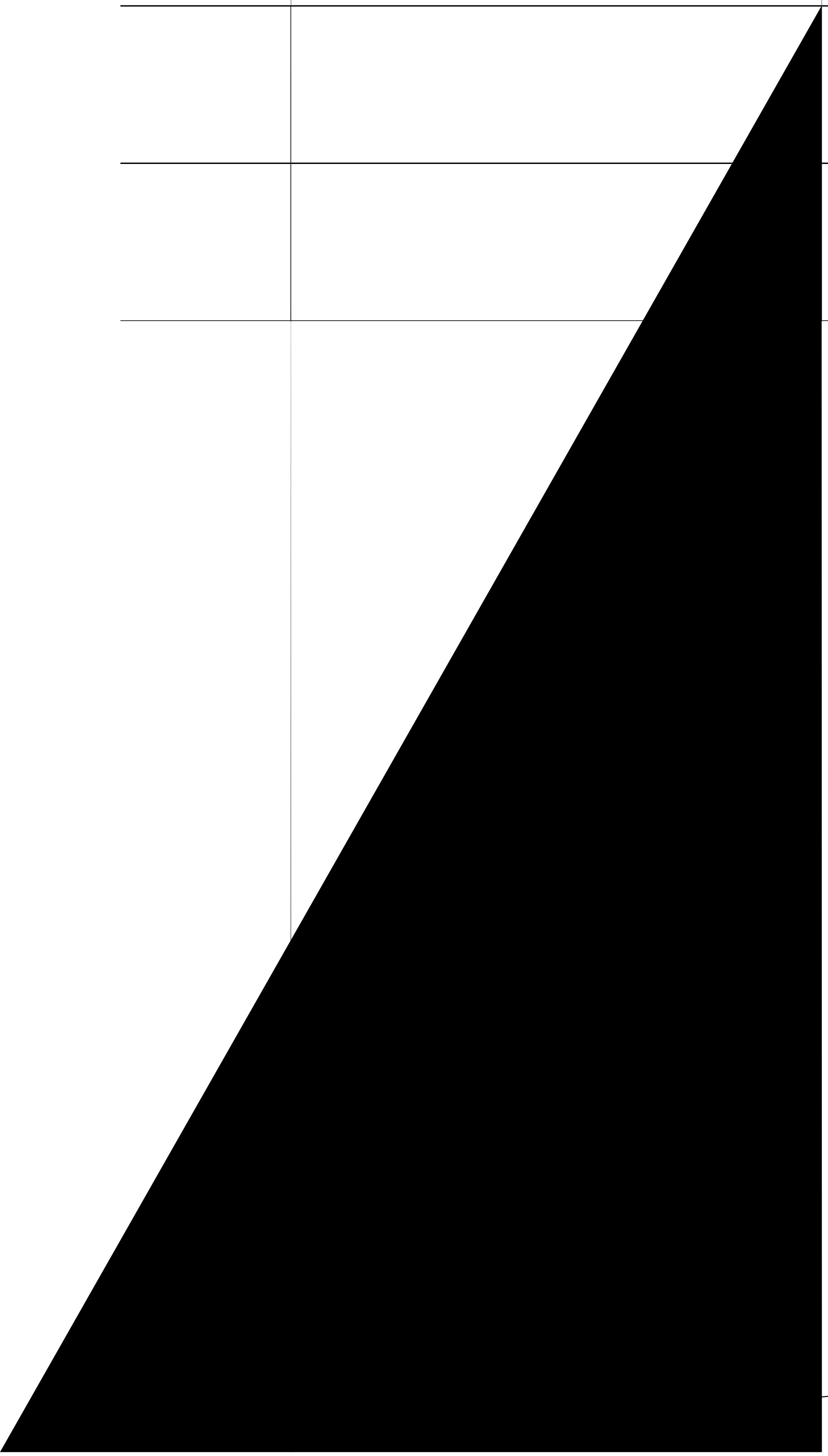
Symmetric and Asymmetric Calls

Cisco UBE supports dynamic payload type negotiation and interworking for all symmetric and asymmetric payload type

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

Perform this task to configure the pass through of DTMF or codec payload to the other call leg (instead of performing dynamic payload type interworking) feature



show voip rtp connection

PART

Table 48: Feature Information for Delayed-Offer to Early-Offer

Delayed-Offer to Early-Offer in Media Flow-Around Calls

Delayed-Offer to

DETAILED STEPS

Configuring Delayed Offer to Early Offer for Video Calls

SUMMAR

	Command or Action	Purpose
Step 3	**medial flow-around** Example: `Device(config-voi-serv)# media flow-around`	Enables media flow-around.

Step 4	Configure conversion of a delayed offer to an early offer:
	In dial-peer configuration mode
	voice-class sip early-offer forced
	In global VoIP SIP configuration mode
	early-offer forced
	
	Example:
	In dial-peer configuration mode:
	`Device (config) dial-peer voice 10 voip`
	`Device (config-dial-peer) voice-class sip early-offer forced`
	`Device (config-dial-peer) end`
	
	Example:
	In global V	

The **early-offer forced renegotiate [always]** command is used to configure this in global VoIP configuration mode (config-voi-ser

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

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *id* **voip**
4. **media transcoder high-density**
5. **end**

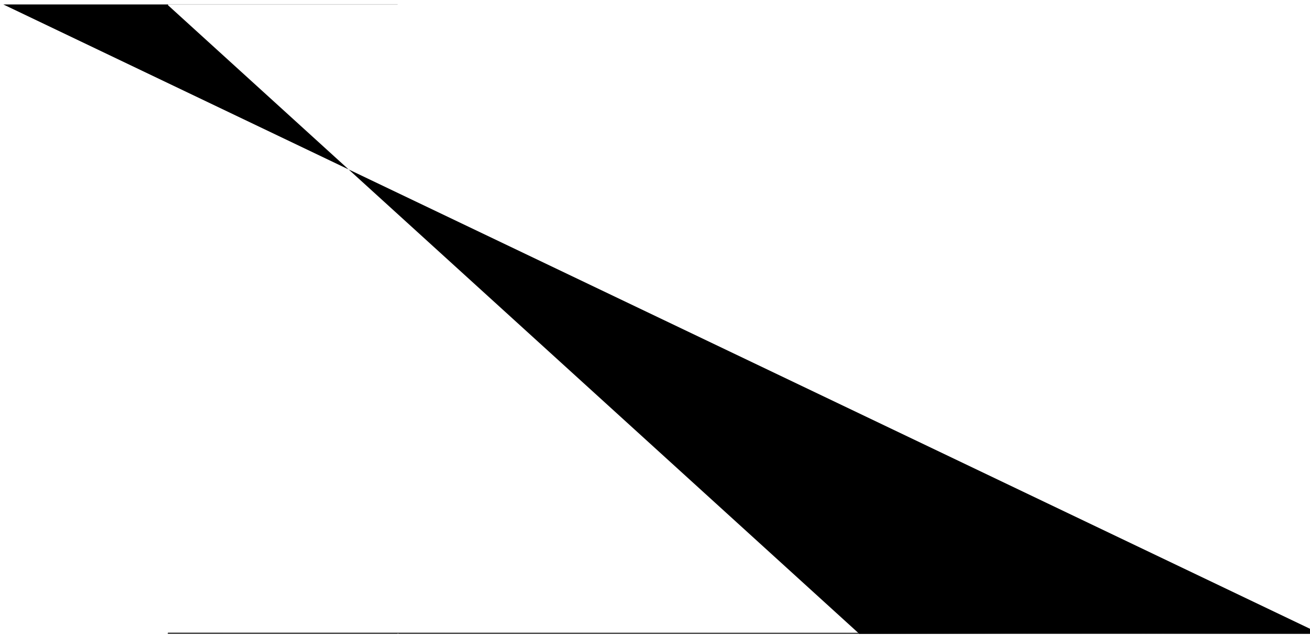
DETAILED STEPS

In the figure below, XIP1 is passed to CUCM1 when a 200 OK is received from SBC1. ACK from CUCM1 triggers new RE-INVITE with transcoding IP address and port number (XIP2) and this RE-INVITE has to be

Configuring High-Density Transcoding

To

[redacted]



[redacted]

Configuring H323-to-SIP Interworking

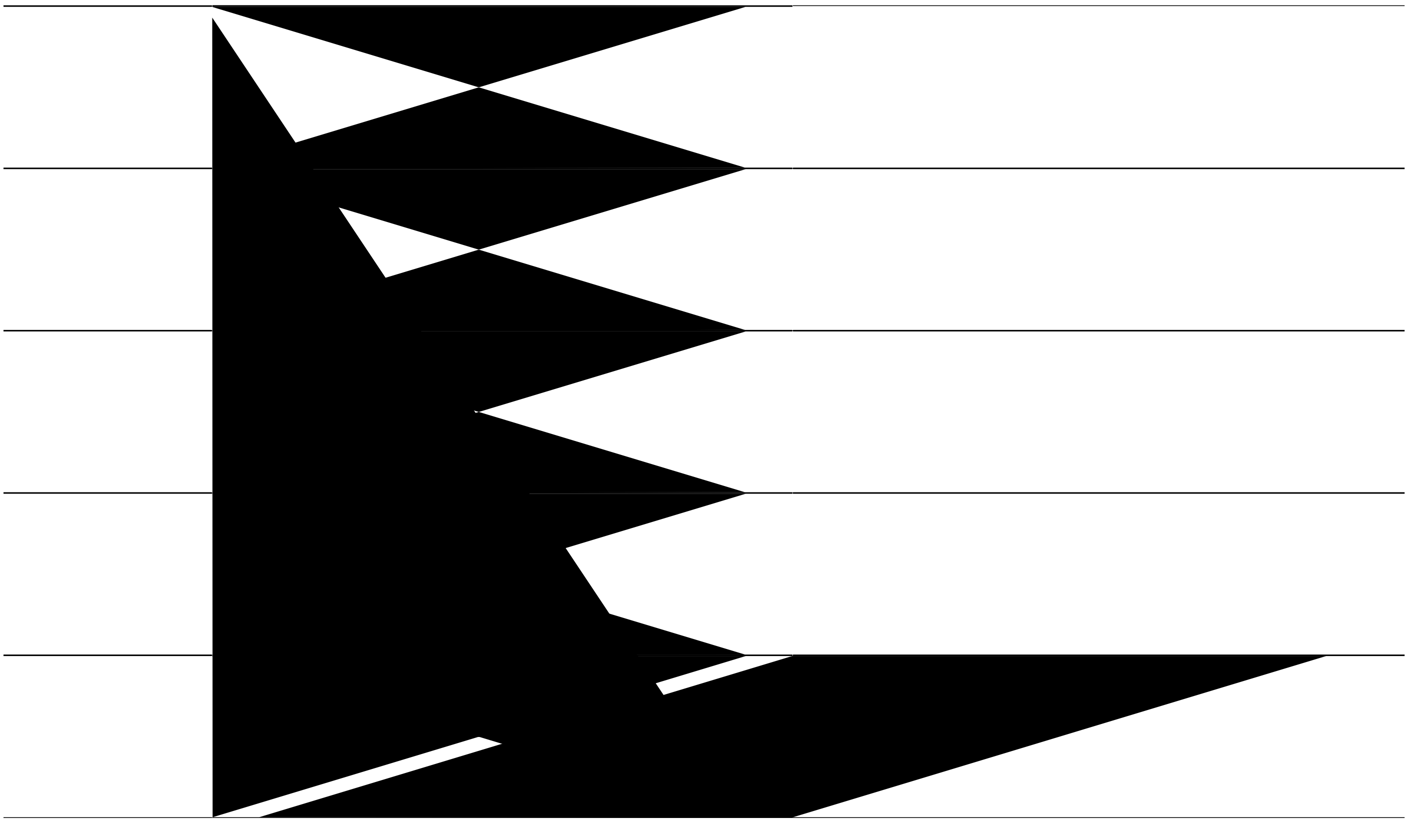
SUMMARY STEPS

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

T

Prerequisites

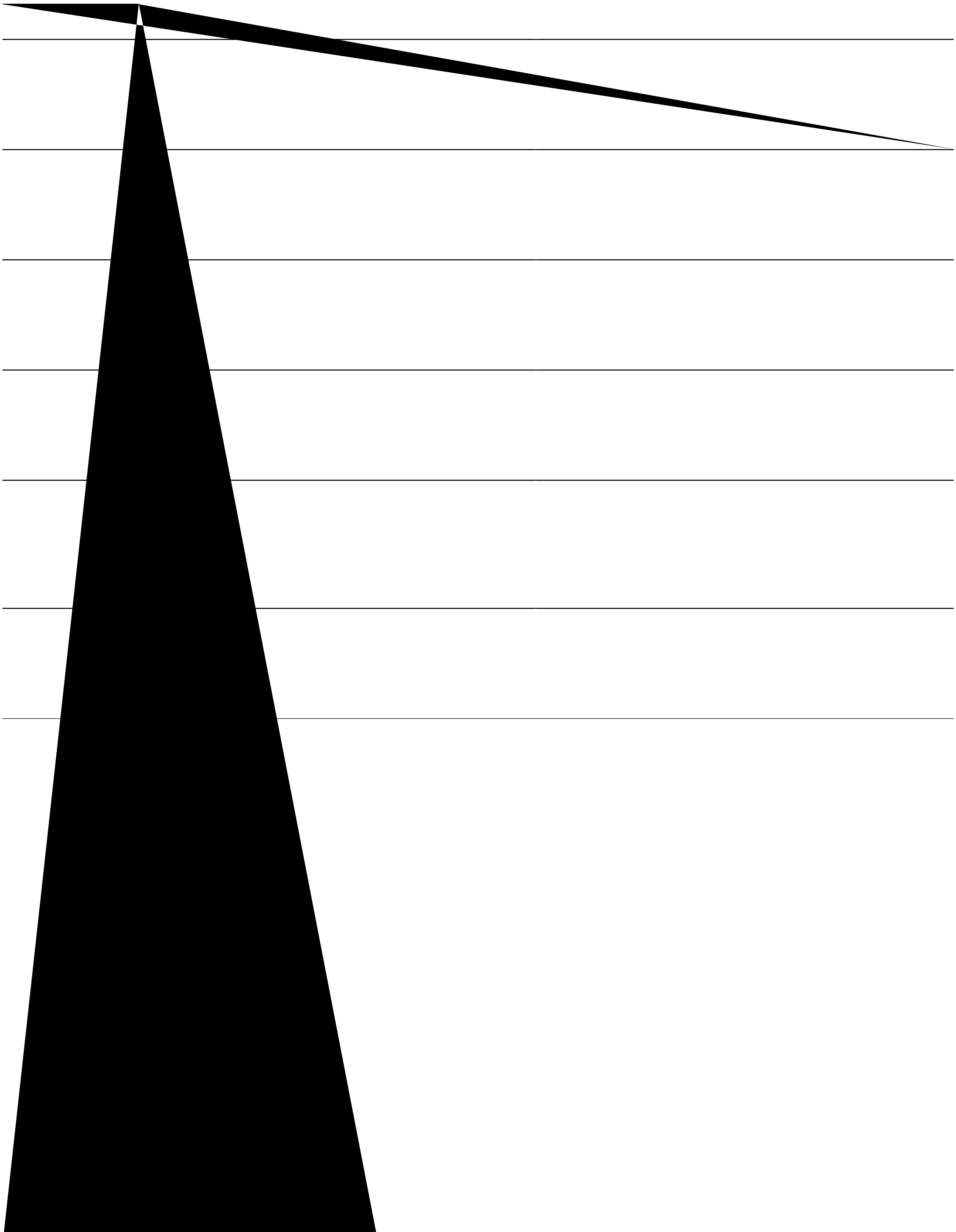
Enable CUBE



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



SUMMARY STEPS

1. **show call active**

T

[redacted]

CHAPTER

SRTP-SRTP Interworking

Cisco Unified Border Element (CUBE) supports secure calls betweenUnihs

Prerequisites for SRTP-SRTP Interworking

Cisco IOS XE

AES_CM_128_HMAC_SHA1_32

Figure 44: SRTP-SRTP Interworking

CUBE allows you to change the list of preference order of the cipher-suites. Cipher-suite preference can be configured globally

For call transfers involving REFER and 302 messages (messages that are locally consumed

SUMMARY STEPS

1. **enable**
2.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Apply crypto suite selection

SUMMARY STEPS

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

Configuration Examples

Example: Configuring SRTP-SRTP Interworking

The following example shows how to configure support for SRTP-SR

SIP Call ID   : 706E9

CHAPTER [redacted]

SRTP-RTP Interworking

The Cisco Unified Border Element (CUBE) Support for SRTP-RTP Interworking feature allows secure network to non-secure network calls and provides operationalto

R

SRTP-RTP interworking also connects SRTP enterprise networks with static IPsec over external networks, as shown inthe figure below

SUMMARY STEPS

	Command or Action	Purpose
	Example: `Device(config-dial-peer)#`	

DETAILED STEPS

SUMMARY STEPS

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

Troubleshooting T

```
0    : 5 12:50:14.326 IST Fri Jun 3 2011.2 +0 pid:0 Originate  connecting
 dur 00:01:19 tx:1653/271092 rx:2831/464284 dscp:0 media:0
 IP 10.45.34.252:2000 SRTP: on rtt:0ms pl:0/0ms lost:0/0/0 delay:0/0/0ms g711ulaw TextRelay: off
 media inactive detected:n media contrl rcvd:n/a timestamp:n/a
 long duration call detected:n long duration call duration:n/a timestamp:n/a
```

Configuration Examples for SRTP-RTP Interworking

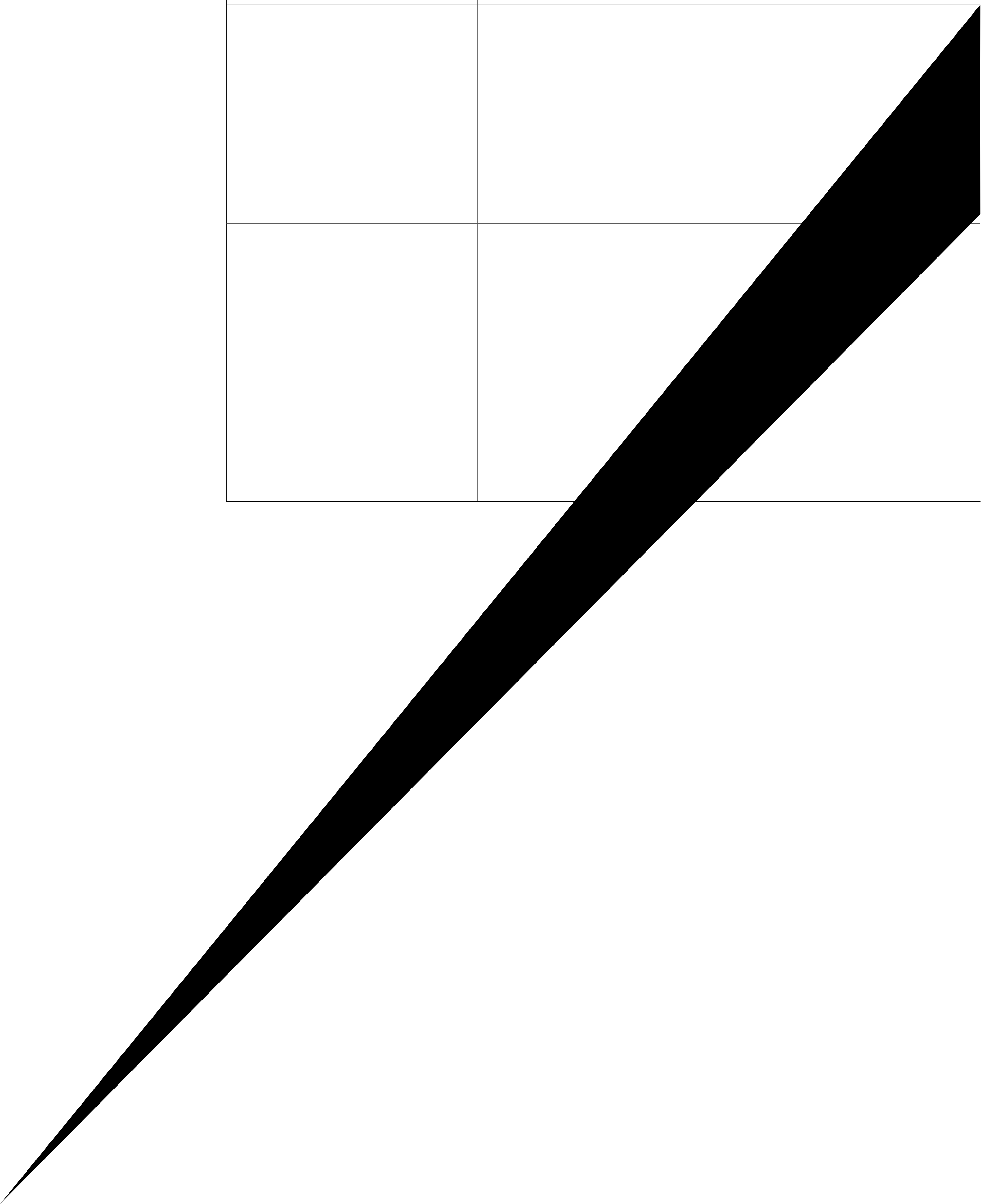
Example: Configuring Crypto Authentication (Global Level)

The following example shows how to configure Cisco UBE to support an SRTP connection using the AES_CM_128_HMAC_SHA1_80 crypto suite at the global level:

```
Device> enable
Device# configure terminal
```

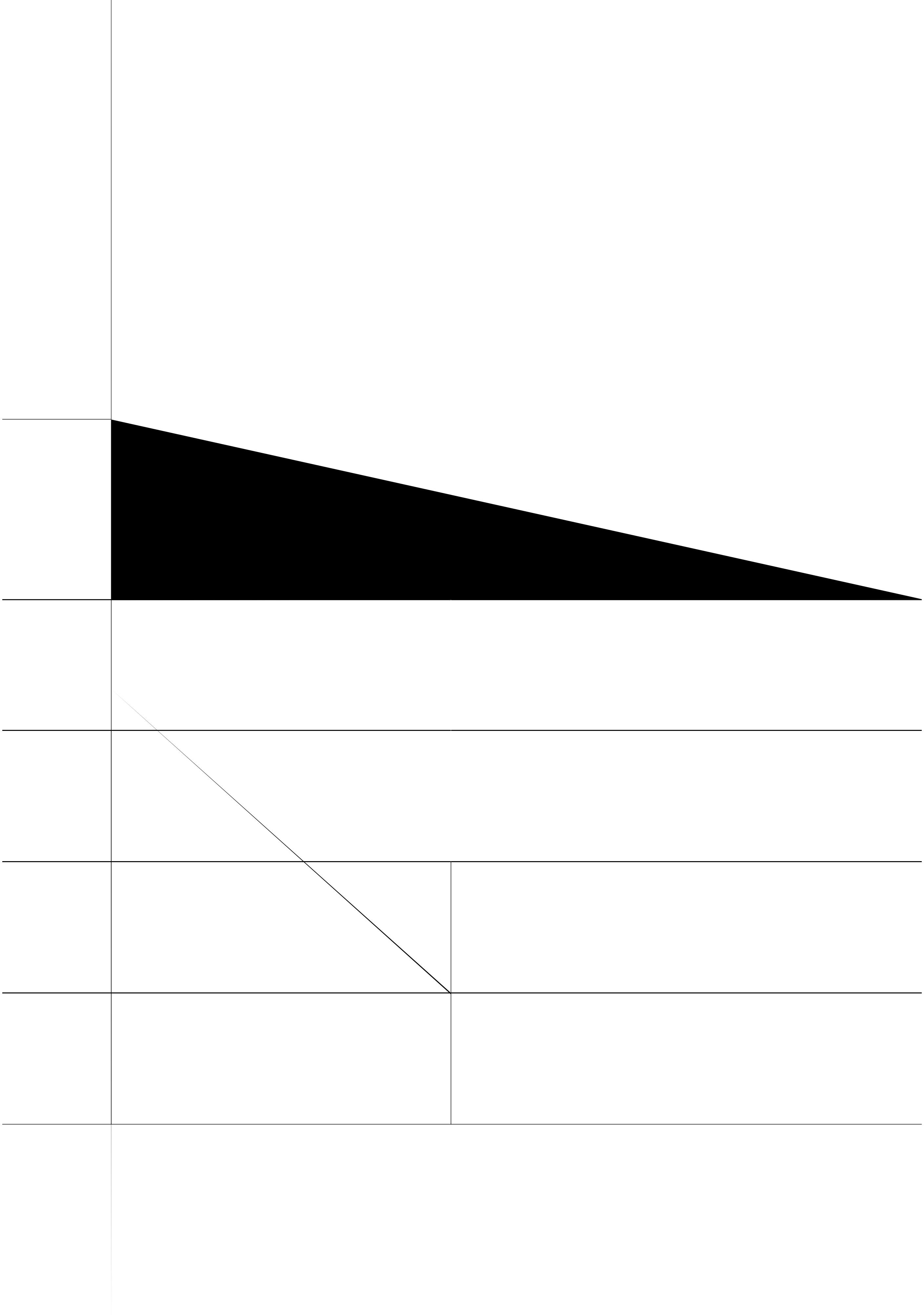
[redacted]

CHAPTER [redacted]



CUBE supports transparent passthrough of all (supported and unsupported) crypto suites.

Until Cisco IOS



Configuration Examples for SRTP-SRTP Pass-Through

Example for SRTP=SRTP Pass-Through

```
enable
configure terminal
dial-peer voice 201 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.102
incoming called-number 5550111
srtp
codec g711ulaw
end

dial-peer voice 200 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.101
incoming called-number 5550111
srtp
codec g711ulaw
end
```

Example for Pass-Through of Unsupported Crypto Suites for a specific dial peer

```
enable
configure
```

PART XII

High Availability

CHAPTER [redacted]

CUBE High Availability Overview

High Availability (HA) is a feature that ensures the availability of resources in a computer

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

Inbox redundancy	Supported only on ASR devices. Inbox

Clustering with load balancing Clustering

Licensing implications and

[redacted]

CHAPTER

Dolby Noise Reduction (NR) and Acoustic Shock Protection (ASP) are not supported.

All SCCP-based media resources (Conference bridge, Transcoding, HW MTP, and SW MTP) are not supported with Cisco

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

Figure 54: Call De-escalation

Media Forking with High Availability

Media forking with high availability is supported

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 recording Media Service Provider (MSP) session information. In the output shown, the fork leg details and the number of forking calls are displayed. Both the active and standby devices will have the same call information.

Example:

```
Device# show voip recmsp session

RECMSP active sessions:
MSP Call-ID                AnchorLeg Call-ID        ForkedLeg Call-ID
4441                       4440                     4442
Found 1 active sessions
```

Step 5

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 supported in Cisco U

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 XIII

[redacted]

CHAPTER [redacted]

Table 57: 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

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

PART XIV

SIP Protocol Handling

[redacted]

CHAPTER

Mid-call Signaling Consumption

The Cisco

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 Hold (MMOH) is not supported.

When

	Command or Action	Purpose
	midcall-signaling block In dial-peer configuration mode	

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 59: Feature Information for Mid-call Signaling

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

CHAPTER [redacted]

Consumption of Forked 18x Responses with SDP During Early Dialog

The Cisco Unified Border Element supports consumption of forked 18x responses with SDP

Table 60: Feature Information for Consumption of Multiple Forked 18x Responses with SDP During Early Dialog

If PRACK and UPDATE are supported and CUBE has to consume the forked 18x responses and initiate renegotiation after call connect, then the

DETAILED STEPS

SUMMARY STEPS

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

   In the dial-peer configuration mode

   voice-class

[redacted]

CHAPTER [redacted]

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

The Support for Pass-Through of Unsupported

[redacted]

CHAPTER

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 BorderUn l ent Un rUn seUn a m cent c Un nf

Table 63: Feature Information for PAID and PPID Headers on Cisco Unified Border Element (CUBE)

T

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

	Command or Action	Purpose
		Enables the vV

SUMMARY STEPS

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

Configuring Random-Contact Support on a Cisco Unified Border Element

To configure random-contact

[redacted]

SUMMARY STEPS

PART XV

SIP Supplementary Services

[redacted]

CHAPTER

Dynamic Refer Handling

When a dial-peer match occurs, CUBE passes the REFER message from an in leg to an out leg.

Table 65: Feature Information for Dynamic REFER Handling

Feature Name	Releases	Feature Information
		REFER Consume (Enhancements) provides additional configurations to conditionally forward the REFER message E

	Command or Action	Purpose
	Example5785.2/F1 10 Tf 1 0 0 1 92.899 634.966 Tm 5785.2/F1 10 Tf G0 0 1 303D3408.45m BHTc BQS8.0Tc B\c B\00R\.89 In Global VoIP configuration	

[redacted]

Table 67: Feature Information for Cause Code Mapping

Feature Name	Releases	Feature Information
		With the Cause Code Mapping feature, the NOTIFY message sent by CUBE to a Customer Voice Portal (CVP) contains a proper reason for failure of call transfer based on the information received by CUBE

Cause code mappings for cause code 19 and 21 require configurations mentioned in Configuring Cause Code Mapping, on page 646.

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

V

[redacted]

CHAPTER [redacted]

Cisco Unified Communications Manager Line-Side Support

Cisco Unified Communications Manager is an enterprise-class IP communications processing system. It extends

Line-Side Support for CUCM on CUBE

For an IP phone to register on a CUCM through CUBE, CUBE must be configured to do the following requirements.

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

```
Device(config)# enrollment terminal
Device(config)# revocation-check none

Device(config)# crypto pki certificate chain selfsignx
Device(config)# certificate self-signed 01
[Certificate data
```

Attaching Phone Proxy

PART XVII

Licensing

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

[redacted]

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

		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
!
```

```
subject-name cn=plutododsn
revocation-check none
rsakeypair selfsign
!
crypto pki trustpoint ccm155RSA
enrollment terminal
revocation-check none
!
!
crypto pki certificate chain ecdsacert1
certificate 07
```

```
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
```

[redacted]

CHAPTER [redacted]

CUBE Call Quality Statistics Enhancement

Call quality statistics in CUBE, such as packet loss, jitter, and round trip delay

Table 72: 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

which uses these values in statistics calculation. Calculated statistics such

Troubleshooting Call Quality Statistics

Use the following

RoundTripDelay

GapFillWithSilence

GapFillWithPrediction

GapFillW

IOS VQM, Voice/Audio Description Quality Metric	Description
round-trip-delay	The instantaneous round-trip delay. This may be obtained from the RTCP SR reports.
receive-delay	The minimum delay that will be applied to the packets received when using an adaptive jitter buffer.

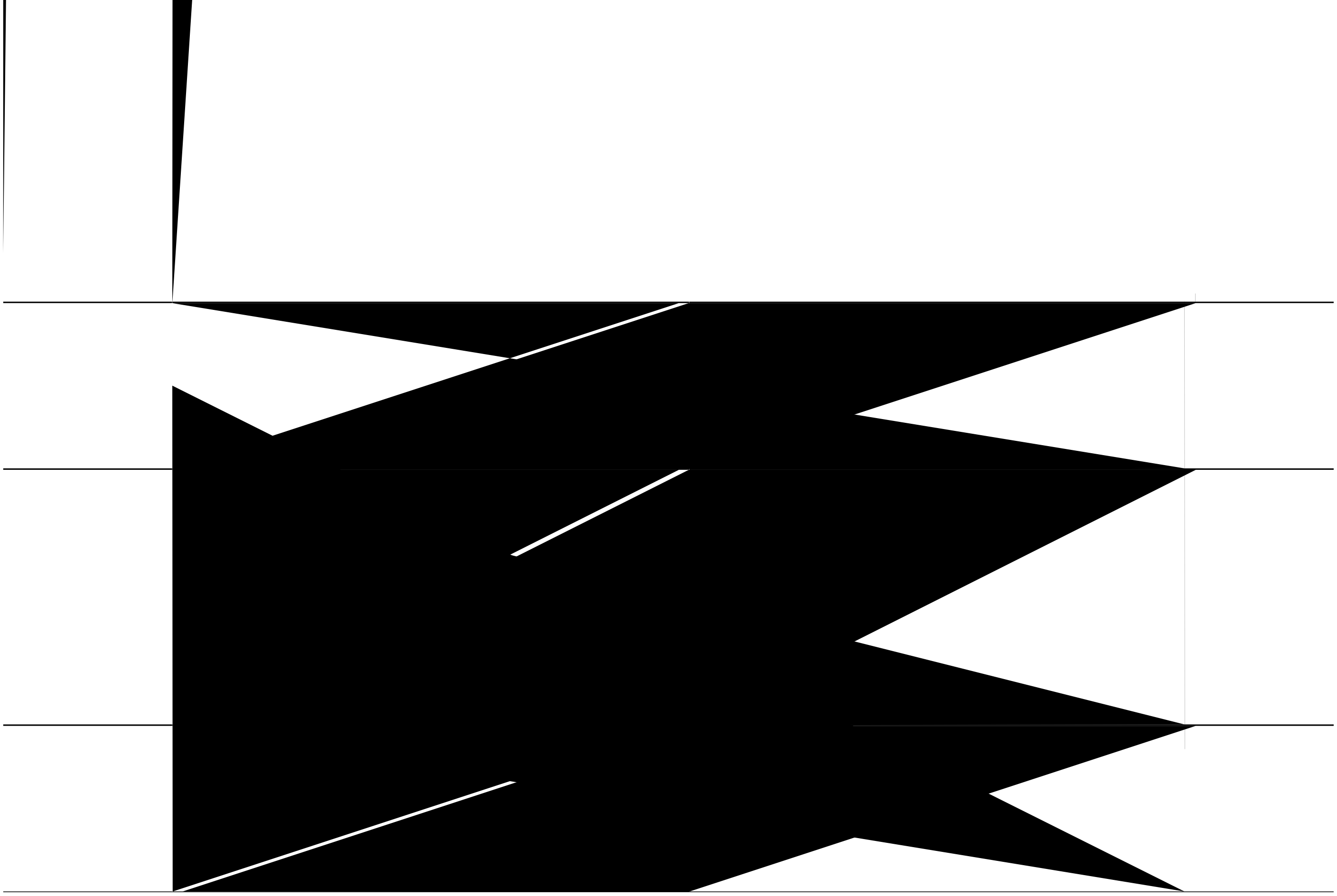


Table 74: Router Output Definitions for the show call active voice stats command

Example: CDR Enabled MOS Output

At the end

PART XX

Serviceability

Table 75: Feature Information for Session Identifier Support

WORD can be complete session identifier

.
.
SessionIDLocaluuid=4fd24d9121935531a7f8d750ad16e19

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

PART XXI

Appendixes

Additional

[redacted]

CHAPTER

Additional References

The following sections provide references related to the CUBE Configuration Guide.

Related References

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