

Introduction

This document answers frequently asked questions about the Cisco Unified Border Element (CUBE).

Q1. What are the benefits of using Cisco Unified Border Element ?

A:

- Multiple physical interconnects, intelligent OAM, Call Admission Control, Billing etc
- Security Demarc: FireWall, DOS protection, VPN, etc
- Signaling, Protocol & Media Interworking: H.323/SIP, transcoding, DTMF etc.
- Media QoS control and monitoring
- Interoperability: various network elements, CUCM, etc.
- Co-existence/co-operation with TDM trunking

Q2: What type of Calls are supported in CUBE ?

A: CUBE supports,

- IP-to-IP Calls
- TDM Voice Calls
- Gatekeeper
- Routing

Q3: How the Media is handled in CUBE ?

A: Media is handled in 2 different modes

- Media Flow-Through
- Media Flow-Around

Q4: What Platforms are supported for CUBE?

A: Cisco CUBE is an Integrated application with Cisco IOS software.

CUBE functionality is supported in Cisco 2600XM, 2691, Cisco ISR 2800, 3800 series, Cisco VXR 7200, Cisco XR 12000, AS5400XM Universal gateways and the Service Provider Gateways.

Q5: Do we need Digital signal processors (DSPs) for CUBE?

A: Digital Signal Processors (DSPs) are only required for calls with dissimilar codecs

Q6: What is the licensing requirements of the CUBE?

A: If you have any existing Cisco Gateway then you can download the Cisco IOS supported version for running CUBE on your Gateway. Also you need to buy license for the no. of sessions in CUBE for different platform. Say for an example if you want to have a CUBE supporting 500 sessions , you need to buy the license for this amount.

Q7: How to have a basic setup between CUCM and CUBE?

A: The layout will be CUCM-sip-Cube-sip-provider.

- First you need to create the SIP trunk on the CUCM and point it to the cube
- Bind your sip signalling to the appropriate interfaces on the CUBE
- Enter the respective dial-peers on the cube for inbound and out bound calls to and from the CUCM

Q8: Do CUCM need special license for CUBE or it will be just a SIP trunk without license?

A: No, there is no special license for connecting CUCM to CUBE..

Q9: WE have a CUBE for a SIP connection. I know there is a license that we purchased from Cisco. Is there a command to enter on the CUBE to view the licenses?

A: The CUBE licenses are not entered into the router. As of now, they are paper licenses.

Q10: Whether CUBE supports Voice and Data traffic simultaneously?

A: Yes. CUBE is in IOS based platform hence it can handle both Voice and Data traffic.

Q11: Does CUBE have to be run on a dedicated device/router?

A: Not Necessary. It depends on the voice traffic you want to route using CUBE. For large enterprise its recommended to use standalone device for CUBE functionality. If the traffic is not high you can run other services along with the CUBE application for voice on the same platform simultaneously.

Q12: Do we need DSP resources in CUBE if the SIP trunk from city (external) to the SIP trunk of CUCM will be used?

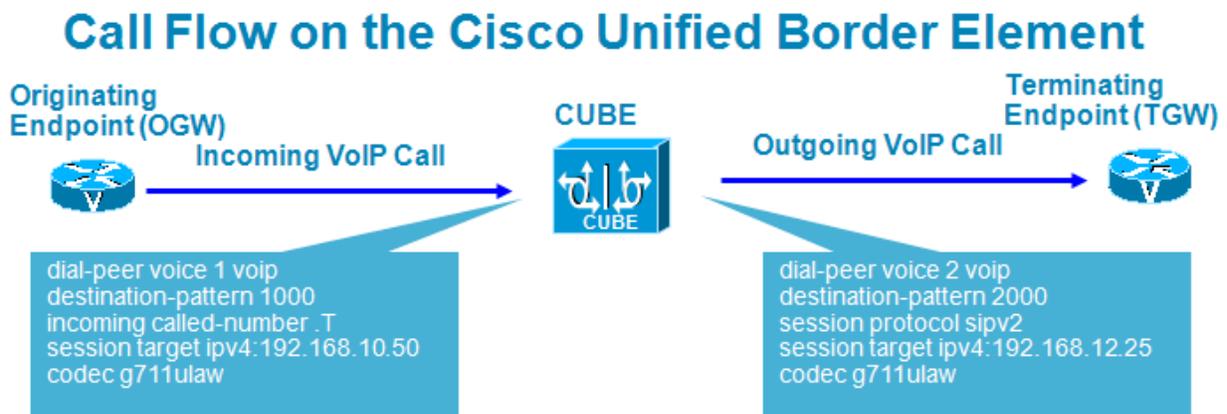
A: DSP resources are required to simply connect to SIP Trunk. DSP resources are required if you plan on doing transcoding between g711 to g729 codec. For conferences if all parties are using g711 codec, then you can use the software resources in UCM to do the conferencing. If one of the parties is using g729 codec and if that needs to be conferenced, you will need to use voice gateways for hardware conferencing which will require DSPs. Digital Signal Processors (DSPs) are only required for calls with dissimilar codecs.

Q13: Will SRST work on CUBE ?

A: Yes, SRST and CUBE can be co-located on the same router.

Q14: Explain the Call flow in CUBE.

A:



1. Incoming VoIP setup message from OGW to CUBE
2. This matches an inbound VoIP dial peer 1 for characteristics such as codec, VAD, DTMF method, protocol etc
3. CUBE then looks up called number in setup and matches outbound VoIP dial peer 2
4. Outgoing VoIP setup message from CUBE to TGW

Q15: How to enable the IP-IP Calls and Protocol support in CUBE ?

A: Enabling the IP-to-IP Calls

```
CUBE#config t
CUBE(config)# voice service voip
```

CUBE(conf-voi-serv)#allow-connections h323 to h323
CUBE(conf-voi-serv)#allow-connections h323 to sip
CUBE(conf-voi-serv)#allow-connections sip to h323
CUBE(conf-voi-serv)#allow-connections sip to sip

It is mandatory to have Incoming and Outgoing VoIP Dial-peers with required parameters like Protocol, Transport, Codec, CAC, QoS, etc

Q16: Whether CUBE supports Universal Transcoding ?

A: Yes CUBE supports Universal Transcoding, any to any Voice codec. Example. iLBC to G.711 or iLBC to G.729

List of codecs supported

- g711alaw 64Kbps
- g711ulaw 64Kbps
- g723r53 5.3Kbps
- g723r63 6.3Kbps
- g723r63 6.3Kbps
- g729 (all variants) 8Kbps
- iLBC

Q17: What is CAC?

A: CAC is Call Admission Control. It control number of calls based on resources and bandwidth and Proactively reserve resources for good quality video calls. Also it ensure traffic adheres to QoS policies within each network. CUBE can provide six different CAC mechanisms

Q18: Is CP-PWR-CUBE-4= compatible with the 7900 series phones?

A: No this power cube is for the new 8900/9900 phones, compatible with Communications Manager. The power tip is different size, won't fit in other phones.

Q19: Im trying to find a way to route inbound SIP calls, based on their source.

For example:

SIPCarrier1 is coming from 1.1.1.1

SIPCarrier2 is coming from 2.2.2.2

If calls originate from SIPCarrier1, I want them to be passed to my UCM Cluster.

If call originate from SIPCarrier2, I want them to be passed to my Asterisk box.

I've played around with incoming called-number, trying to make it work with IP addresses, rather than phone numbers, but no luck.

Any ideas on how to enable source based routing?

A: Yes this can be achieved by Carrier based Routing. Refer this link for more details.

[VoIP Gateway Trunk and Carrier Based Routing Enhancements](#)

Q20: CUBE Sizing, how many simultaneous calls/sessions are supported on each Router model to terminate a SIP Trunk at the service provider end?

A: Refer the given table for more details.

Enterprise Size	SIP Trunk Sessions	Redundancy Recommendation	Platform Recommendation
Small	<100	None	Single 2901
	100-200	None	Single 2911
	200-500	Optional	Single 2951
Medium	500-1000	Recommended	No redundancy: Single 3900 Local redundancy: Dual HSRP 3900 Geo Redundancy: Dual 3900
	1000-2500	Must-have	No redundancy: Single 3900E Local redundancy: Dual HSRP 3900E Geo redundancy: Dual 3900E
Large	2500-5000	Must-have	Inbox redundancy: Single ASR1004/6 RP2 Geo redundancy: Dual ASR1004/6 RP2
Very Large	5000+	Must-have	Inbox redundancy: Single ASR1006 RP2 Geo redundancy: Dual ASR1006 RP2

Platform	CUBE Sessions
C880/C890 SKUs (Future)	5-15 ←
2801	55
2811	110
2821	200
2851	225
3825	400
3845	500
AS5000XM	600
2901	100
2911	200
2921	400
2951	600
3925	800
3945	950
3925E	2100 ←
3945E	2500 ←
ASR1002/1004/1006 RP1	1750
ASR1004/1006 RP2	15000

Q21: What are the types of CUBE licenses available?

A: Two types of Cisco Unified Border Element licenses are available:-

- Pay-as-You-Grow, or session count, license
- Platform, or flat, license

Cisco Unified Border Element licenses are available in both types on select platforms. Cisco Gatekeeper licenses are available in the platform license type only.

- Pay-as-You-Grow License

This type of license covers the right to use the feature as well as the maximum session count allowed. An example is the FL-CUBE-25 license, which allows up to 25 sessions.

This license is designed to allow a specific number of sessions (or calls) on a platform. Purchase only as many sessions as are required in your deployment. You can add more licenses later as your needs expand, thus offering pay-as-you-grow benefits. Session licenses are stackable. These licenses are available on select platforms as given in Table 3, and are available on all software images. Examples include the FL-CUBE-4 and FL-CUBEE-100 licenses.

- Platform License

This type of license covers the right to use the feature up to the maximum session count supported on the chosen platform. An example is the FL-INTVVSERV-2811 license, which allows the maximum number of sessions the Cisco 2811 platform supports.

These licenses are available on select platforms. These licenses require a software image. Examples include the FL-INTVVSERV-2801 and FL-GK-3945 licenses.

- Cisco Unified Border Element Licenses

For Active/Standby redundant configurations, use the "Redundant-Platform" licenses. For all other configurations, including single-platform Active/Active load balancing, and Inbox redundant configurations, use the "Single-Platform" licenses.

Q: Can we Integrate CUBE with Siemens OSV. Can CUBE do it? What are the element in specific needed for Integration and deployment ? Please share a sample configurations?

A: We have validated the integration with Siemens HiPath 4000. More details on the integration is:

<http://www.cisco.com/en/US/solutions/collateral/ns340/ns414/ns728/ns833/698642.pdf>

Q22: I am trying to troubleshoot the Multicast MOH on Cube, how to do it?

Following commands will help you to analyse and troubleshoot the MOH issue you are facing on CUBE.

Debug commands:

- debug ccm-manager music-on-old all
- debug voip rtp
- debug ccsip all

Show commands:

- show version
- show running configuration

When MMOH is being streamed, please collect the following output on CUBE:

- show ccm-manager music-on-hold
- show voip rtp connections
- show call active voice compact

Note: Please collect the logs using buffered logging mechanism. You can configure the following:

no logging queue-limit

logging buffered 80000000

no logging rate-limit

no logging console

Q23: Debug commands on CUBE

DEBUG COMMANDS - Make sure to clear logg before call is made & get show logg after call is done

1. H.323 - H.323 Scenarios

debug h225 asn1

debug h225 events

debug h245 asn1

debug h245 events

debug h225 q931

debug cch323 all

debug voip ipipgw

debug voip ccapi inout

2. H.323 - SIP Scenarios

debug h225 asn1

debug h225 events

debug h245 asn1

debug h245 events

debug cch323 all

debug voip ipipgw

debug voip ccapi inout

debug ccsip all

3. SIP - SIP Scenarios

debug ccsip all

```
debug voip ccapi inout
```

4. Transcoder related scenarios

Apart from the debugs mentioned above based on the scenario

```
debug dspfarm all
```

```
debug sccp messages
```

Verifying Fundamental Cisco Unified Border Element Configurations

To verify Cisco Unified Border Element feature configuration and operation, perform the following steps (listed alphabetically) as appropriate.

Step 1 show call active video

Use this command to display the active video H.323 call legs.

Step 2 show call active voice

Use this command to display call information for voice calls that are in progress.

Step 3 show call active fax

Use this command to display the fax transmissions that are in progress.

Step 4 show call history video

Use this command to display the history of video H.323 call legs.

Step 5 show call history voice

Use this command to display the history of voice call legs.

Step 6 show call history fax

Use this command to display the call history table for fax transmissions that are in progress.

Step 7 show crm

Use this command to display the carrier ID list or IP circuit utilization.

Step 8 show dial-peer voice

Use this command to display information about voice dial peers.

Step 9 show running-config

Use this command to verify which H.323-to-H.323, H.323-to-SIP, or SIP-to-SIP connection types are supported.

Step 10 show voip rtp connections

Use this command to display active Real-Time Transport Protocol (RTP) connections.

This FAQ Document was created from the Cisco Unified Border Element related discussions in Cisco Support Community

Related Information

- [SIP-to-SIP Connections on a Cisco Unified Border Element](#)
- [Cisco Unified Border Element \(CUBE\) / SIP Trunking Solutions](#)
- [CUBE High Availability\(HA\) configuration on ASR Platform](#)
- [CUBE High Availability\(HA\) configuration on ISR-G2](#)