The bridge to possible

Data sheet
Cisco public

Cisco Unified Border Element Version 14

Contents

Product overview

Part of the Cisco® Collaboration Edge Architecture, Cisco Unified Border Element (CUBE) version 14 is an enterprise-class Session Border Controller (SBC) solution that makes it possible to connect and interwork large, midsize, and small business unified communications networks with public and private IP communication services.

As a licensed feature set of Cisco IOS® XE Software, CUBE has a wide range of capabilities that may be used to secure, monitor, and maintain business-critical connections and to ensure compliance with industry standards. Collectively, CUBE features provide exceptional flexibility when architecting highly available enterprise communications networks that save money and offer richer voice and video collaboration experiences to users.

Comprehensive interworking

As voice, video, and mobile communications systems converge to form more cost-effective, integrated collaboration solutions, the need to interwork diverse networks based on various protocols and security requirements increases. The CUBE SBC serves a critical role in linking these networks and provides a seamless experience for voice and video users.

CUBE is especially suited to facilitating:

- PSTN interconnect using Internet service provider SIP trunks, which allow rapid service delivery and the possibility of capacity pooling across locations.
- Migration from TDM to SIP public telephony trunk services. As a number of Cisco routers allow the concurrent use of voice gateway and CUBE features, a phased trunk migration is possible without requiring changes to the enterprise call control platform.
- Certified connection to Cisco and third-party cloud collaboration services, including Cisco Webex® Cloud Connected Audio (CCA and CCA-SP), Webex Calling Local Gateway, Cisco Hosted Collaboration Solution (HCS) and Direct Routing for Microsoft Phone System (Microsoft Teams), with normalization to customer collaboration systems. CUBE supports high-capacity SIP media connectivity to the Cisco Webex cloud to replace expensive TDM audio connections to conferencing services.
- Business-to-business voice and video system interconnect.
- Multi-tenant solutions that require customer-dedicated SIP trunks on a common platform.
- Codec interworking through the control of midcall codec renegotiation or transcoding.

Note: Cisco Unified Communications Manager customers requiring business-to-business video features should use Cisco Expressway™.

As CUBE terminates and re-originates signaling and media traffic, it is able to provide a secure demarcation between internal and external services, while interworking signaling protocols and encoded media streams between them. Further, CUBE provides a rich set of flexible session control features to secure and route traffic to different destinations and to apply policing and Quality-of-Service (QoS) policies.

Certain CUBE features may also be used with Cisco Communications Manager Express (CME) and Unified Survivable Remote Site Telephony (SRST) applications to connect with SIP trunk services.

Security and compliance

As networks become more interconnected, the need to secure information is of critical importance. Enterprises must comply with rapidly evolving industry standards for the proper handling and protection of sensitive and private information and for the proper auditing of commercial transactions. The comprehensive CUBE SBC feature set helps businesses achieve these requirements with:

- Flexible security rules that prohibit unauthorized connections.
- Behavior evaluation policies that can detect malicious call patterns, including Telephony Denial-of-Service (TDoS) attacks, and invoke an appropriate response – such as terminate, redirect, or record.
- Interworking of encrypted and non-encrypted communication streams.
- Replication of media streams for call recording solutions using either SIPREC or HTTP API.

Cloud communications services

Cloud call control products offer simple-to-provision-and-manage services. However, by their very nature, these services place a greater dependency on the wide-area connections required at customer sites. While additional bandwidth and redundant connectivity can mitigate this requirement, service providers can also use CUBE lineside features to ensure continued service delivery.

- CUBE registration proxy can manage periodic messaging from Cisco Multiplatform Phones (MPP) or third-party SIP endpoints, reducing demand on wide-area connections and permitting larger deployments of endpoints.
- Lineside survivability provides business continuity to SIP phones on a customer site should connectivity to the cloud service be interrupted.

Note: CUBE lineside features are offered for use with SIP-based, IP Centrex solutions (such as Cisco BroadCloud®). They cannot be used with Cisco Unified Communications Manager, where Expressway and Unified Survivable Remote Site Telephony products should be considered.

Contact center solutions

CUBE offers numerous features that may be used to architect and optimize fully featured contact center solutions. Examples include:

- Call Progress Analysis (CPA) for outbound calling campaigns
- Interactive Voice Response (IVR) solutions
- Media replication for call recording and analysis

Flexibility, Reliability, and Scale

Cisco offers industry-leading flexibility when it comes to deploying SBC functionality in almost any enterprise architecture. As CUBE is offered as part of Cisco IOS XE Software, it may be used concurrently with industry-leading IP networking, security, and QoS features. You can also choose from a wide range of host platforms to suit scale, performance, resiliency, and budget requirements (see Table 2).

In addition to physical hosts from Cisco Integrated Services Router (ISR), Cisco Catalyst® Edge Routers and Aggregation Services Router (ASR) product families, CUBE features are available for virtualized environments with the Cisco Cloud Services Router (CSR) and Catalyst Edge Software.

Stateful high availability with active/standby redundant pairs and clustering with Cisco Unified SIP Proxy allows enterprises to build highly scalable, business-critical solutions.

CUBE licensing models

The CUBE features described above are licensed to enable three principal use models:

- **Trunking** for service interconnect and protocol interworking. Trunk licenses are available for both standard (single node) and enhanced (high-availability and advanced features) network architectures to facilitate site-to-site and PSTN connectivity. Each trunk license enables a single call session in addition to a single forked media session for recording where required.
- **Lineside** to enhance the delivery of hosted SIP communications services. Previously only available for the Cisco 800 Series Routers through the NanoCUBE license, CUBE Lineside client licenses are now available for all platforms listed in Table 2. Each Lineside license enables registration proxy and survivability features for one local SIP endpoint.
- **Media Proxy** for advanced call recording and compliance solutions. Deployed independently from CUBE platforms configured for trunkside or lineside applications, CUBE Media Proxy allows corporate customers to meet compliance requirements by simultaneously recording or analyzing calls at up to five destinations simultaneously. Each Media Proxy license enables one forked media session in either standard or redundant configurations.

Smart Licensing

CUBE Smart Licenses allow for entitlement pooling and portability across all CUBE platforms registered to an organization's Cisco Smart Licensing account. Providing further flexibility, Cisco Smart Licensing also allows the borrowing of higher-entitlement CUBE licenses if required.

Starting with CUBE version 12.5 (Cisco IOS XE 16.10.1a), all platforms must be registered with a customer's Cisco Smart Software Management service account. For more information regarding Smart Licensing, see: https://www.cisco.com/go/smartlicensing

CUBE feature support

CUBE supports a comprehensive range of session control, security, interworking, and demarcation SBC features, many of which are detailed in Table 1.

Table 1. Cisco Unified Border Element features

Feature	Support details
Protocols[1]	• H.323 and SIP
Protocol and signal interworking[1]	• H.323 to H.323 (including Cisco Unified Communications Manager) • H.323 to SIP (including Cisco Unified Communications Manager) • SIP to SIP (including Cisco Unified Communications Manager and Cisco TelePresence®)
Media support	• RTP and RTCP • Binary Flow Control Protocol (BFCP) passthrough
Media interworking[1]	• SIP delayed-offer to SIP early-offer interworking for audio or video calls • H.323 Slow Start to H.323 Fast Start for audio calls
Media modes	• Media flow-through • Media flow-around
Signaling transport mode	• Transport Control Protocol (TCP) • Transport Layer Security (TLS) • User Datagram Protocol (UDP) • TCP, TLS and UDP interworking
Fax support	• T.38 fax relay • Fax pass-through • Fax over G.711
Modem support	• Modem pass-through • Modem over G.711
Dual-Tone Multifrequency (DTMF)[1]	• H.245 alphanumeric • H.245 signal • RFC 2833 /RFC 4733 • SIP notify • Key Press Markup Language (KPML) • Interworking capabilities include: ◦ H.323 to SIP ◦ RFC 2833/4733 to G.711 in-band DTMF[2] ◦ sip-info to rtp-nte interworking ◦ Various SIP-to-H.323 DTMF interworking options ◦ RFC 2833/4733 to KPML

Feature	Support details
Supplementary services[1]	• Call hold, call transfer, and call forwarding for H.323 networks using H.450 and transparent passing of Empty Capability Set (ECS) • SIP-to-SIP supplementary services (holds and transfers) support using REFER • SIP-to-SIP supplementary services (holds and transfers) support using REINVITE • H.323-to-SIP supplementary services for Cisco Unified Communications Manager with Media Termination Point (MTP) on the H.323 trunk • Multicast Music on Hold (MMoH) to Unicast MoH conversion • Call Progress Analysis (CPA)[2] to analyze far-end media (live versus recorded media) for outbound call centers
Internetworking	• Configurable SIP profiles to manipulate SIP message content, including header fields and Session Descriptor Protocol (SDP) attributes • Conditional SIP profiles, performing header modification dependent on header content • P-Asserted-Identity (PAI), P-Preferred-Identity (PPI), and Remote-Party-ID (RPID) internetworking • Unsupported Multipurpose Internet Mail Extensions (MIME)-type attachment pass-through • Unsupported SIP header pass-through • SDP attribute pass-through • Dial-peer bind (allows CUBE to connect to multiple service providers) • Incoming dial-peer match based on remote IP address • Assisted RTCP for Microsoft Lync/Skype for Business interoperability • Mid-call signaling block or pass-through when media changes • Early dialog UPDATE /183 consumption • Block incoming 180 and 183 signaling messages • Restrict video call to audio only • Media Anti-trombone • IPv4 to IPv6 interworking • Configurable SIP error codes • SIP error code pass-through
Call routing and dialing options	• E164-based dialing • Uniform Resource Identifier (URI)-based dialing • Routing based on nonsequential E164 and/or URI lists • Destination-based or source-based routing • Dial Peer Groups (Trunk Groups) (outbound routing determined by inbound dial pattern) • Server Groups to define order of selection of alternative or backup routing paths for outbound routing • Routing based on duple header variables (both AND OR logic) • Refer and call redirect consumption and pass-through • Outbound call load distribution with random or round robin schemes • Call re-routing based on network errors or error responses • P-called-party-ID support
Multitenancy, multi-VRF, and trunk realms	• Support for dial plan scenarios requiring either or both inter- and intra- IP VRF routing tables • Per-VRF-domain SIP user agent for multi-tenancy support (up to 100 VRFs) • Realm commonality of multiple trunks, even with different user agent definitions per trunk
Cisco Call Admission Control (CAC)	• CAC based on maximum number of calls per trunk (maximum number of calls) • CAC based on IP circuits • CAC based on total calls, CPU use, or memory use threshold • CAC based on bandwidth availability and call-spike detection

Feature	Support details
OPTIONS SIP message support	• Support for response to OPTIONS-PING messages with OPTION-PING groups based on session target • Support for generation of in-dialog OPTIONS-PING messages • Support for generation of out-of-dialog OPTIONS-PING messages to control dial-peer status
Media forking	• Media forking features for voice and video to integrate with media recording or analysis servers • API-based mechanisms for invoking media forking • Support for standard SIPREC media forking • Raw media forking using secure WebSockets for Cisco Contact Center solutions. (Requires Enhanced License) • Media Proxy mode for forking two to five concurrent media sessions[3] • Secure forking of a non-secure call
IP routing feature	• Support for Cisco IOS XE Software-based routing features, including Border Gateway Protocol (BGP), Enhanced IGRP (EIGRP), and Multiprotocol Label Switching (MPLS) • Support for Cisco IOS XE Software-based policy routing features • Support for Cisco IOS XE Software-based Access-Control-List (ACL) features
Voice-quality statistics	RTCP data from incoming and outgoing call legs used to provide: • Packet loss, jitter, and Round-Trip Time (RTT) • Per-call leg call-quality statistics
QoS	• IP precedence and Differentiated-Services-Code-Point (DSCP) marking • Per-call QoS packet marking
Network Address Translation (NAT) traversal	• NAT traversal support for SIP phones deployed behind non-Application Line Gateway (ALG) data routers • Stateful NAT traversal • ICE-Lite
Network hiding	• IP network privacy and topology hiding • IP network security boundary • Intelligent IP address translation for call media and signaling • Back-to-back user agent, replacing all SIP-embedded IP addressing • History information-based topology hiding and call routing
Number translation	• Number translation rules for Voice-over-IP (VoIP) numbers • URI-based dialing translations
Codecs	• OPUS low bitrate 6 kbps to very high-quality 510 kbps • G.711 mu-law and a-law • G.722 • G.723ar53, G.723ar63, G.723r53, and G.723r63 • G.726r16, G.726r24, and G.726r32 • G.728 • G.729, G.729A, G.729B, and G.729AB • Internet Low Bitrate Codec (iLBC) 13330 or 15200 bps • Internet Speech Audio Code (iSAC) 10 to 32 kbps • AAC-LD MP4A-LATM • Mid-call codec renegotiation and preservation • Narrowband Adaptive Multi-rate (AMR-NB) 4750-12200 bps

Feature	Support details
Transcoding[2]	• Transcoding between any two different families of codecs from the following list: ◦ G.711 a-law and mu-law ◦ G.729, G.729A, G.729B, and G.729AB ◦ iLBC ◦ G.722 ◦ OPUS (PVDM4 modules only) • Mid-call transcoder insert and drop
Transrating[2]	• Transrating of packetization rates for the following codecs: ◦ G.711 a-law and mu-law ◦ G.723 5.3/6/3 kbps ◦ G.729, G.729A, G.729B, and G.729AB ◦ G.722
Security	• Rogue SIP invite and rogue RTP packet detection with alerting • Configurable RTP port range • IP security (IPsec) • SRTP flow-through • Transport Layer Security (TLS) version 1.2, with exclusivity • SRTP-to-RTP and STRP-to-SRTP interworking with Next-Generation Encryption (NGE) cipher suites • Configurable SIP listening port • Disable unused transport mechanisms • SIP registration and digest authentication support • Various mechanisms for control of RTP and UDP packet flooding • Voice security policy application integration (via HTTP API) • Peer whitelisting /IP Trusted List • Silent discard of SIP messages from untrusted peers • Compatible with IOS Zone Based Firewall
Authentication, Authorization, and Accounting (AAA)	• AAA with RADIUS
Voice media applications	• Tool Command Language (TCL) scripts support for application customization • Web-based API to monitor and control signaling and media traffic (for external policy control)
API	• Web-based API compatible with Web Service Description Language (WSDL) development tools to support call monitoring and control, Call-Detail Records (CDRs), and serviceability attribute interaction with external application; specifically designed for voice-policy applications
Billing	• Standard CDRs for accurate billing available through: ◦ AAA records ◦ Syslog ◦ Simple Network Management Protocol (SNMP)
Line-side Registration Proxy	• Proxy registration of endpoints using the standard SIP registration process (including third-party SIP endpoints) for connecting with third-party hosted call-control services (e.g. Cisco BroadSoft®) • Local and PSTN survivability in the event of loss of WAN connectivity to a hosted call control • Proxy endpoint registration with 10 endpoints per SIP registration event
Inter-Cluster Lookup Service (ILS) routing	• Support for ILS routing to complement ILS dial-plan exchange between Cisco Unified Communications Manager clusters or to simplify call-routing complexity between multiple clusters

Feature	Support details
Video	
Protocols[1]	• H.323 and SIP
Rich media	• Simultaneous support for data, audio, and video
Signaling interworking	• SIP delayed-offer to SIP early-offer calls
Media	• Support for multiplex RTP calls (for Cisco TelePresence solution) • Simple Traversal of UDP through NAT (STUN) /Datagram TLS (DTLS) pass-through for telepresence
H.323-enhanced features[1]	• H.235 pass-through for secure calls • H.239 pass-through for picture-in-picture feature
QoS	• DSCP markings to prioritize video streams as they traverse the network
Data support	• T.120 data collaboration flow-around only
Camera control	• Far-End Camera Control (FECC)
Video suppression	• Terminate video media session for connection to audio-only sessions
Video codecs	• H.261 • H.263/H.263+ • H.264 • MPEG4
Network management	
Manageability, serviceability, and troubleshooting	• Resource usage monitoring over SIP trunk • Sortable dial peers • SIP session ID for end-to-end call tracing • SNMP per-call quality traps • SNMP and syslog SIP trunk status messages • DEBUG commands allowing user-selectable levels of debug information, from critical to verbose • DEBUG commands allowing user-selectable information for specific call characteristics • VoIPTrace continuous diagnostic capture
High availability	
High availability	• Inbox redundancy with Cisco ASR 1006 and ASR 1006-X • Box-to-box redundancy with Cisco 4000 Series ISRs, Catalyst Edge, Cisco ASR 1000, and CSR 1000V models (based on RG infrastructure) • Use of port channels to allow a connection to redundant switches • Requires Enhanced Trunk or Media proxy session license.

[1] H.323 features are deprecated from IOS XE 17.6.1 onwards. Refer to the [Product Bulletin] for further information.

[2] Requires DSP resource

[3] Media Proxy mode cannot be used concurrently with Voice Gateway, CUBE trunk or CUBE lineside features.

Router platform support

CUBE is developed as a component of Cisco IOS XE Software and runs on the platforms listed in Table 2. Maximum trunk session capacity and call processing performance is provided here for the purposes of comparison only. Operational capacity is dependent on various factors, such as call presentation rate, call type (for example, call center or standard IP telephony), transcoding, encryption, and media forking. Figures included in Table 2 are based on simple telephony calls and assume ideal conditions that optimize for either call volume or rate. Capacity figures are provided for platforms running IOS XE 17.3.1 or newer releases.

Certain CUBE deployment scenarios may require additional hardware for WAN termination or transcoding. If connected to the IP network through a WAN circuit, CUBE supports all connectivity methods and interface cards supported by the host router platform.

Virtualized CUBE (vCUBE) is available as a licensed feature for the Cisco Cloud Services Router (CSR 1000V) and Catalyst Edge 8000V software, allowing customers to use CUBE features in Network Functions Virtualization (NFV) environments.

CUBE features that require direct access to DSPs or voice interfaces are not available with vCUBE or ISR1100 models.

Table 2. CUBE Platform Support

Router platform	Minimum memory requirement	Maximum trunk sessions	Maximum sustainable call setup rate (Calls per second)
Cisco 1100 ISR[1]	Default	500	5
Cisco 4321 ISR	4 GB	500	4
Cisco 4331 ISR	4 GB	1000	10
Cisco 4351 ISR	4 GB	2000	13
Cisco 4431 ISR	8 GB	3000	15
Cisco 4451-X ISR	8 GB	6000	40
Cisco 4461 ISR	8 GB	10,000	55
C8200L-1N-4T	4GB	1500	9
C8200-1N-4T	8GB	2500	14
C8300-1N1S-6T	8GB	7000	40
C8300-1N1S-4T2X	8GB	8000	45
C8300-2N2S-6T	8GB	7500	42
C8300-2N2S-4T2X	16GB	10,000	55
Cisco CSR 1000V /C8000V 1vCPU[2]	4 GB	1000	5

Router platform	Minimum memory requirement	Maximum trunk sessions	Maximum sustainable call setup rate (Calls per second)
Cisco CSR 1000V /C8000V 2vCPU²	4 GB	3000	20
Cisco CSR 1000V /C8000V 4vCPU²	8 GB	6000	30
Cisco ASR 1001-X	16 GB	12,000	50
Cisco ASR 1002-X	16 GB	14,000	55
Cisco ASR 1006-X with RP3 and ESP100/ESP100X	16 GB	16,000	65
Cisco ASR 1004 /ASR 1006 / ASR 1006-X with RP2 and ESP40	16 GB	16,000	70

[1] ISR1100 4G/6G models for SDWAN do not support the CUBE feature set.

[2] Based on tests using Cisco UCS® C240 host with Intel® Xeon® 6132 2.60GHz processors running VMware ESXi 6.7. Performance quoted for 2 vCPU virtual machine also applies for vCUBE in Amazon Web Services.

Licensing options

CUBE may be purchased either through a Cisco Collaboration Flex Plan subscription, or perpetual licensing with Software Support Service (SWSS).

CUBE license subscriptions may be purchased using **A-FLEX-3, A-FLEX-3-EDU**, or **A-FLEX-3-FEDRAMP** offers with options listed in Table 3. For more information on ordering CUBE as a subscription, refer to the Cisco Collaboration Flex Plan Ordering Guide. Entitlements provided through Flex subscriptions are term-based and may be used with any version of CUBE. **Note:** Subscriptions for CUBE Line side licenses are not currently available.

Perpetual CUBE licenses may be ordered using the Smart License-enabled product codes in Table 4.

Cisco CUBE Version 14 licenses also provide entitlement to use all earlier, currently supported application versions (including those that use RTU licenses). All license options require at least one year of Software Support Service (SWSS).

Table 3. CUBE Subscription options

Product Code	Description
A-FLEX-ENH-CUBE	One CUBE trunk enhanced session subscription
A-FLEX-STD-CUBE	One CUBE trunk standard session subscription
A-FLEX-MP-CUBE	One Media Proxy stream subscription

Table 4. CUBE perpetual smart license ordering information

Product code	Description
L-CUBE	Top-level part number. Select this first
CUBE License options	
CUBE14-T-STD	One Standard Trunk Session License
CUBE14-T-ENH	One Enhanced Trunk Session License (Includes support for box-to-box redundancy)
CUBE14-L-STD	One Standard Lineside License
CUBE14-MP	Entitlement for one Media Fork with support for box-to-box redundancy
CUBE License upgrade option	
MIG-CUBE14-T-ENH	Upgrade one Version 12 or 14 Standard Trunk License to Version 14 Enhanced

Cisco environmental sustainability

Information about Cisco's environmental sustainability policies and initiatives for our products, solutions, operations, and extended operations or supply chain is provided in the "Environment Sustainability" section of Cisco's Corporate Social Responsibility (CSR) Report.

Reference links to information about key environmental sustainability topics (mentioned in the "Environment Sustainability" section of the CSR Report) are provided in the following table:

Sustainability topic	Reference
Information on product material content laws and regulations	Materials
Information on electronic waste laws and regulations, including products, batteries, and packaging	WEEE compliance

Cisco makes the packaging data available for informational purposes only. It may not reflect the most current legal developments, and Cisco does not represent, warrant, or guarantee that it is complete, accurate, or up to date. This information is subject to change without notice.

Cisco Capital

Flexible payment solutions to help you achieve your objectives

Cisco Capital makes it easier to get the right technology to achieve your objectives, enable business transformation and help you stay competitive. We can help you reduce the total cost of ownership, conserve capital, and accelerate growth. In more than 100 countries, our flexible payment solutions can help you acquire hardware, software, services and complementary third-party equipment in easy, predictable payments. Learn more.

Summary

Organizations large and small are realizing the value of SIP-based communication. The Cisco session border controller, CUBE, is helping these organizations take advantage of service providers' SIP services by providing voice and video connectivity for both trunk and lineside services. As such, CUBE is ideal for businesses of all sizes; it cost-effectively supports a variety of SIP services, whether premises-based or with hosted call control, with the added benefit that CUBE uses the customer's existing investment in Cisco routers.

For more information

For more information about the Cisco Unified Border Element (CUBE), visit https://www.cisco.com/go/cube or contact your local Cisco account representative.

Document history

New or revised topic	Described In	Date
Added subscription ordering and H.323 deprecation details		
Added information relating to newer Catalyst 8000 series products.		May 20, 2021
Added information relating to newer Catalyst 8000 series products.		Mar 15, 2021
Added information related to Catalyst Edge router platforms.		Nov 2020
Content updated for CUBE 14.0 (IOS XE 17.3.1)	Router platform support, new codec, Microsoft Phone System support and version 14 licensing	July 31, 2020
Content updated for CUBE 12.8 (IOS XE 17.2.1)	Router platform support	April 04, 2020
Added minimum memory requirements, new support for ISR1100 and updated scaling figures	Router platform support	Sep 27, 2019
Content fully updated for CUBE Version 12	Updated: CUBE Version 12	Feb 13, 2019

Americas Headquarters
Cisco Systems, Inc.
San Jose, CA

Asia Pacific Headquarters
Cisco Systems (USA) Pte. Ltd.
Singapore

Europe Headquarters
Cisco Systems International BV Amsterdam,
The Netherlands

Cisco has more than 200 offices worldwide. Addresses, phone numbers, and fax numbers are listed on the Cisco Website at **https://www.cisco.com/go/offices**.

Printed in USA C78-729692-13 08/21