Data Sheet

Cisco Unified Border Element Version 10.0.2

Product Overview

You can do a lot at the edge of your network with Cisco® Unified Border Element. You can make the transition from TDM to SIP trunking easier, tailoring the architecture to meet your network and business needs. You can significantly decrease toll charges for web conferencing. You can stop telephony denial-of-service attacks at your gate. You can give remote workers access to collaboration - without the hassle of a VPN. Workers from your company can join others in B2B telepresence meetings with access as easy as sending an email message. If outgoing calls from your contact center don't connect with a person, you can keep those calls from cluttering up agents' time. You can more easily access cloud collaboration services, and providers can offer these services more cost-effectively. Cisco Unified Border Element gives you all these capabilities and more.

Part of the Cisco Collaboration Edge Architecture, Cisco Unified Border Element (CUBE) is an enterprise-class session border controller that connects large, midsize, and small business unified communications networks to the IP PSTN. In addition to session border control, CUBE has many other capabilities including simple and cost-efficient collaboration beyond the enterprise firewall. Put all together, these functions give you unprecedented flexibility as to how you can architect your network.

CUBE use cases include:

1. IP PSTN Connectivity via SIP Trunking
2. Teleworker VPN-less Cisco SIP phone registration services
3. Cloud connectivity solutions for Cisco and third-party hosted call control
4. Cisco WebEx® Cloud Connected Audio
5. Business-to-business telepresence, voice, and video interconnect
6. Voice and video recording
7. Call-center and interactive-voice-response (IVR) solutions
8. Voice policy

Let's look at these use cases in more detail.

IP PSTN Connectivity with SIP Trunking

The benefits of unified communications and collaboration services will continue to grow as voice, video, and mobile services become more pervasive elements of integrated collaboration solutions. However, these enhancements will be available only if you deploy end-to-end real-time IP communications for both inter- and intracompany voice and video services based on Session Initiation Protocol (SIP). This deployment will require transitioning your service provider network interconnect from time-division multiplexing (TDM) circuits to SIP trunking.

The session border controller (SBC) has become a critical network component for scaling and securing unified communications networks. The SBC makes it possible to expand end-to-end IP connectivity for real-time voice and video through service provider SIP trunks or through a secure SIP session over the Internet to directly connect two enterprise networks.

Cisco Unified Border Element (CUBE) supports the transition to SIP trunking by enabling Cisco or third-party IP call/session control to connect to and interoperate with service provider SIP trunk services. CUBE terminates and reoriginates both signaling (H.323 and SIP) and media streams (Real-Time Transport Protocol [RTP] and Real-Time Control Protocol [RTCP]) to provide secure border interconnection services between IP networks. Using CUBE, Cisco customers can save on their current network services, simplify their network architectures, and position their networks for ongoing enhancements in collaboration services.

As an integrated Cisco IOS® Software application, CUBE runs on a broad range of Cisco router platforms, including the Cisco 800, 2900, 3900, and 4000 Series Integrated Services Routers (ISRs) as well as the Cisco ASR 1000 Series Aggregation Services Routers (ASRs). The breadth of Cisco router platforms that support the CUBE feature license means that it provides unsurpassed price/performance scalability compared to other enterprise SBCs. This scalability translates into network design flexibility for enterprise, midsize, and small businesses. It also means operational efficiencies and a broader serviceable market for service providers that include CUBE as part of their SIP trunk managed or hosted services.

CUBE performs the following functions between the enterprise and service provider networks:

- **Session control:** The capability to offer flexible trunk routing, Call Admission Control (quality of service [QoS]), and resiliency and call accounting for the SIP sessions processed by the SBC
- **Interworking:** The capability to interconnect different signaling methods and media encoding variants for both voice and video sessions
- **Demarcation:** The capability to act as a distinct demarcation point between two networks for address and port translation and to facilitate troubleshooting
- **Security:** The capability to intelligently allow or disallow real-time traffic between networks, and to encrypt the real-time traffic as appropriate for the application

CUBE also enables **flexible SIP interconnect:** As a Cisco IOS Software feature set that runs on a broad range of Cisco routers (see complete list in Table 2, later in this document), CUBE enables great flexibility in how you design your SIP trunk network. Of course, this means your existing inventory of Cisco routers (as per the list in Table 2) can be repurposed for use as a CUBE platform. As a result you can deploy CUBE in centralized, distributed, or hybrid (combination of centralized and distributed) SIP architectures. This flexibility will be invaluable as collaboration use cases, such as conferencing, video, and mobility, evolve to place increased demands on the enterprise network. Many service providers have recognized their customers' need to adapt their SIP trunk architecture as their collaboration services evolve, and they now offer multilocation SIP trunk pooling services. CUBE provides unsurpassed SBC price/performance flexibility so you can take advantage of these service offerings.

For SIP network deployments that require a centralized architecture, CUBE can be deployed to support highly scalable, highly available configurations referred to as CUBE Clusters. CUBE Clusters are created by combining CUBE with Cisco Unified SIP Proxy (CUSP), and can support up to 64,000 concurrent sessions in a single cluster.

CUBE also enables **simplified TDM-to-SIP migrations** using the Cisco ISRs. For most enterprises, the transition from TDM trunk services to SIP trunk services requires careful planning because this transition must be achieved while maintaining a functional voice network. This planning involves, among other things, addressing the service provider requirement for number portability and the IT requirement for dial-plan revisions. CUBE can simplify this transition, particularly for Cisco customers who already use the Cisco TDM Gateway on the Cisco 2900 and 3900 Series ISR series. These routers can be easily upgraded-without requiring any additional hardware-to support the CUBE feature license, and they can concurrently support SIP trunking and TDM trunking. As a result, you can transition your voice network to SIP trunking while retaining the existing TDM gateway functions. As you become familiar and confident with SIP trunking, you can phase out the TDM Gateway function, or retain it as a high-availability redundant network strategy.

Teleworker VPN-less SIP Phone Registration Services

Cisco Unified Border Element (CUBE) will allow remote Cisco SIP phones (Cisco Unified IP Phone 6900, 7800, 7900, 8800, 8900, and 9900 Series models) to establish a secure application layer connection for both signaling (SIP-Transport Layer Security [TLS]) and media (Secure RTP [SRTP]) through the Internet and then perform a proxy registration of these phones with Cisco Unified Communications Manager, Cisco Business Edition, or Cisco Hosted Collaboration Solution (Cisco HCS).

Cloud Connectivity Solutions for Cisco and Third-Party Hosted Call Control

- **Cisco Hosted Collaboration Solution (HCS) remote SIP phone registration services:** Using CUBE, enterprises of all sizes can enjoy the collaboration benefits of Cisco HCS over the Internet. Using SIP TLS and SRTP, CUBE phone proxy services securely deliver all the capabilities of Cisco HCS without requiring a VPN. CUBE supports Cisco Unified IP Phone 6900, 7800, 7900, 8800, 8900, and 9900 Series models of SIP phones. These services can include registration pass-through, voice-quality metrics, and 911 preemption.
- **Third-Party Hosted Call Control remote SIP phone registration services:** Many service providers offer hosted call-control services to their small customers based on cloud-based private-branch-exchange (PBX) software, such as Broadsoft. Using the NANOCUBE licensing (refer to the “CUBE Licensing Options” section later in this document), Cisco 800 Series ISRs can be included as part of these hosted call-control services to perform gateway functions at the customer premises, such as registration pass-through, voice quality metrics, and 911 preemption.

Cisco WebEx Cloud Connected Audio

Cisco WebEx Cloud Connected Audio (CCA) is a cost-effective audio conferencing solution for Cisco WebEx meetings and is enabled by CUBE. It extends the functions of Cisco Unified Communications Manager and incorporates on-premises equipment to connect your organization to a Cisco WebEx Cloud data center via dedicated peering connections. CUBE provides the high capacity SIP media connectivity to the WebEx® cloud to replace the TDM audio connection. It allows SIP sessions from all enterprise sites to connect to the Cisco WebEx cloud and avoids media hairpinning to provide the most efficient network usage.

Business-to-Business Telepresence, Voice, and Video Interconnect

CUBE provides secure connectivity for Cisco TelePresence® deployments over SIP, either on a point-to-point basis over the Internet between two private enterprise networks or from a private enterprise network to a telepresence service provider.

Voice and Video Recording

CUBE supports voice-recording solutions by providing various mechanisms to invoke media forking on a per call basis. One method, which is SIP-based and is derived from the SIPREC standardization, sends a forked SIP invite to the target recording application server, which can either accept or reject the call. An alternative method is an HTTP-based application programming interface (API) that allows the recording server to instruct CUBE to perform the media forking, and can toggle the media forking through the duration of an active call.

Call-Center and IVR Solutions

CUBE provides numerous features to improve the performance of your enterprise call center. For example, it supports midcall codec renegotiation, with either internal transcoding or external endpoints. CUBE also supports SIP-based Call Progress Analysis for outbound call-center solutions, and runs concurrently on the ISR with the Cisco IOS Software-based VoiceXML client to support IVR integration.

Voice Policy

CUBE, as part of its complete SBC security function, supports policy-led evaluation of phone calls. This capability is becoming increasingly critical as incidents of telephony denial of service (TDoS) are more prevalent, as evidenced by the formal public warnings of such attacks being given by the U.S. Federal Bureau of Investigation (FBI) and Department of Homeland Security (DHS). CUBE enables highly flexible and granular voice-policy solutions to identify specific patterns of calling activity from either internal users (employees) or external callers and to take appropriate action when those patterns occur, including call termination, call redirection, and call recording. This voice-pattern recognition helps ensure that the full capacity of the enterprise voice network is used to support the business as intended.

CUBE Licensing Options

CUBE can be licensed according to the following three different licensing options:

- **Standard CUBE Licensing:** Standard licensing is used for CUBE licensing with all modular Cisco ISR and Cisco ASR 1000 Series platforms to deliver the full range of CUBE functions, but without high availability. This license is per-session, and a session is defined as a two-way call through CUBE regardless of the number of media sessions involved in that call (that is, a video call with four media lines or sessions or an audio call with two media lines or sessions).
- **Redundant CUBE Licensing:** Redundant licensing is used for CUBE with all modular Cisco ISR and Cisco ASR 1000 platforms to deliver the full range of CUBE functions, including high availability with call preservation between an active and standby CUBE. This licensing option also allows for license transfer between two geographically distributed CUBE platforms, as in the scenario of a dual data center deployment strategy, where CUBE service redundancy without call preservation is satisfactory.
- **NANOCUBE Licensing:** NANOCUBE licensing is used only for Cisco 800 Series ISR and Cisco Service Provider Integrated Access Device (SP-IAD) platforms typically as part of a third-party (not based on Cisco Hosted Collaboration Solution (HCS)) cloud hosted call-control solution. This licensing option also supports other CUBE features, except to the extent that such features require additional hardware platform support, such as DSPs for transcoding.

More information about these licensing options can be obtained in the CUBE Ordering Guide available at: http://www.cisco.com/go/cube.

CUBE Feature Support

CUBE supports protocols such as H.323 and SIP for both voice and video. Table 1 lists the features supported for voice and video.

Table 1. Cisco Unified Border Element Features (CUBE Versions Include 9.5.1 or Later)

Feature	Support Details
Protocols	• H.323 and SIP
Protocol and signal interworking	• 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) • SIP to SIP (including Cisco TelePresence® calls)
Media support	• RTP, RTCP, and Binary Floor Control Protocol (BFCP) • Sub-RTCP for media statistics
Media interworking	• 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	• TCP • User Datagram Protocol (UDP) • TCP-to-UDP interworking
Fax support	• T.38 fax relay • Fax pass-through • Fax over G711
Modem support	• Modem pass-through • Modem over G711
Dual-tone multifrequency (DTMF)	• H.245 alphanumeric • H.245 signal • RFC 2833 • SIP notify • Key Press Markup Language (KPML) • Interworking capabilities include: ◦ H.323 to SIP ◦ RFC 2833 to G.711 in-band DTMF* ◦ Various SIP-to-H.323 DTMF interworking options ◦ RFC 2833 to KPML
Supplementary services	• 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
Internetworking	• Configurable SIP profiles to manipulate SIP message content, including header fields and SDP attributes • P-Asserted-Identity (PAI), P-Preferred-Identity (PPI), and Remote-Part-ID (RPID) internetworking** • Unsupported Multipurpose Internet Mail Extensions (MIME)-type attachment pass-through** • Unsupported SIP header pass-through** • Dial-peer bind (allows Cisco Unified Border Element to connect to multiple different service providers) • Incoming dial-peer match based on remote IP address • Assisted RTCP for Microsoft Lync Interoperability

Feature	Support Details
Call Routing/Dialing Options	• E164-based dialing • URI-based dialing • Routing based on nonsequential lists (either E164 or URI or both) • 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
Call Admission Control (CAC)	• 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 • Resource Reservation Protocol (RSVP)
OPTIONS SIP message support	• Support for response to OPTIONS-PING messages with OPTIONS 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 recording	• Media forking features for both voice and video to integrate with Cisco TelePresence Media Recording Servers • Active (SIP-based) and passive (API-based) mechanisms for invoking media forking
IP Routing feature	• Support for Cisco IOS Software-based routing features, including Border Gateway Protocol (BGP), Enhanced IGRP (EIGRP), and Multiprotocol Label Switching (MPLS) • Support for Cisco IOS Software-based policy routing features • Support for Cisco IOS Software-based access-control-list (ACL) features
Voice-quality statistics	• Packet loss, jitter, and round-trip time (RTT) • Per-call leg call-quality statistics • Flexible NetFlow call-quality statistics and information • Sub-RTCP statistics collection
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 • IPv4-to-IPv6 translation
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 • Uniform Resource Identifier (URI)-based dialing translations
Codecs	• G.711 mu-law and a-law • G.722 and G.722.2 • 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) • Midcall codec renegotiation • AMR wideband • AAC-LD
Transcoding**	• 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 • Midcall transcoder insert and drop

Feature	Support Details
Security	• Rogue SIP invite and rogue RTP packet detection • Alerts for rogue packet activity • IP Security (IPsec) • SRTP • TLS • SRTP-to-RTP interworking
Authentication, authorization, and accounting (AAA)	• AAA with RADIUS
Voice media applications	• Tool Command Language (Tcl) scripts support for application customization • VoiceXML 2.0 script support for application customization • Web-based API to monitor and control signaling and media traffic
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)
Lawful intercept[**]	• Provision of replicated packets to third-party mediation device
Remote Phone Proxy Sessions	• Termination of SIP-TLS and SRTP with registration pass-through to allow SIP-based endpoints, including Cisco Unified IP Phone 7900, 8900, and 9900 models and Jabber® Voice Client, to connect from remote sites through the Internet without requiring IPsec VPN to Cisco Unified Communications Manager, Cisco Business Edition, or Cisco HCS (Not included with NANOCUBE license)
Line-side Back to Back User AgentNANOCUBE Sessions	• Termination of Cisco Shared Port Adapter (SPA) and other third-party SIP endpoints with registration pass-through and survivability for use with third-party hosted call-control service provider services
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
Video	
Protocols	• H.323 and SIP
Cisco endpoints supported	• Cisco Unified Video Advantage (UVA) and Cisco TelePresence endpoints
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	• 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 codecs	• H.261 • H.263 • H.264
Network Management	
Manageability and serviceability	• Resource usage monitoring over SIP trunk • SNMP per-call quality traps • SNMP and syslog SIP trunk status messages

Feature	Support Details
High Availability	
High availability	• Inbox redundancy on Cisco ASR 1006 • Box-to-box redundancy on Cisco ASR 1000 (based on RG Infrastructure) • Box-to-box redundancy on Cisco Integrated Services Routers (ISR) routers (HSRP-based) **Note:** Media is preserved for active calls at time of failover in each redundancy configuration listed.

** Requires digital signal processors (DSPs) and is available only on the Cisco 2900 Series, Cisco 3900 and 3900E Series, and Cisco 4000 Series Integrated Services Routers and Cisco ASR 1000 Series Aggregation Services Routers.

Router Platform Support

CUBE is developed as a component within Cisco IOS Software and runs on the following platforms:

- Cisco 800 Series ISRs (Cisco 880 Series and Cisco 892F and 897 models)
- Cisco Service Provider Integrated Access Device (SP-IAD)
- Cisco 2900 Series ISRs (Cisco 2901, 2911, 2921, and 2951)
- Cisco 3900 Series ISRs (Cisco 3925 and 3945)
- Cisco 3900E Series ISRs (Cisco 3925E and 3945E)
- Cisco 2900 Series ISRs (Cisco 2901, 2911, 2921, and 2951)
- Cisco 4000 Series ISRs (Cisco 4321, 4331, 4351, 4431, and 4451)
- Cisco ASR 1000 Series Routers (Cisco ASR 1001-X, ASR 1002-X, ASR 1004, and ASR 1006 (RP2))

A minimum of 64 MB of flash memory and 256 MB of DRAM and a minimum of one Fast Ethernet port for an external interface are required.

CUBE may require additional hardware for connectivity to the public switched telephone network (PSTN) and WAN and for transcoding capabilities. If connected to the IP network through a WAN connection, CUBE supports all the WAN connectivity methods and interface cards that the underlying router platform supports.

Note: For transcoding, additional DSPs are required.

More information about transcoding is available at:
http://www.cisco.com/en/US/products/ps5854/products_qanda_item0900aecd8016c2c7.shtml.

Product Specifications and Session Capacities

Table 2 shows platform memory specifications to support CUBE. In this table, the maximum capacities for each of the SIP session types are shown. CUBE supports three session types, as follow:

1. SIP Trunk Session-SIP trunk registration and connectivity to Service Provider
2. Phone Proxy Session-Secure remote phone registration to Cisco Unified Communications Manager, Cisco Business Edition, and Cisco Hosted Collaboration Solution (HCS)
3. NANOCUBE Session-Allows for SIP endpoint access to hosted third-party call control

Table 2. Platform Support, Product Specifications, and Session Capacity and Session Types

Router Platform with Latest CUBE Versions	Flash Memory	DRAM	Maximum SIP Trunk Sessions**	Maximum Phone Proxy Sessions**	Maximum NANOCUBE Sessions**
Cisco 881 ISR* **Cisco 886V ISR*** **Cisco 887V ISR***	Fixed Configuration	Fixed Configuration	15	Not Supported	15
Cisco 888E ISR* **Cisco 888 ISR***	Fixed configuration	Fixed configuration	25	Not Supported	25
Cisco 892F ISR*	Fixed configuration	Fixed configuration	50	Not Supported	50
Cisco 897VA ISR	Fixed configuration	512 MB upgradable to 1024 MB	50	Not Supported	50
SPIAD2901-8FXS/K9*	256 MB	512 MB	80	Not Supported	100
SPIAD2911-16FXS/K9* **SPIAD2911-24FXS/K9***	256 MB	512 MB	100	Not Supported	125
Cisco 2901 ISR	256 MB	1GB	100	50	-
Cisco 2911 ISR	256 MB	1 GB	200	100	-
Cisco 2921 ISR	256 MB	1 GB	400	250	-
Cisco 2951 ISR	256 MB	1 GB	600	400	-
Cisco 3925 ISR	256 MB	1 GB	800	500	-
Cisco 3945 ISR	256 MB	1 GB	950	500	-
Cisco 3925E ISR	256 MB	2 GB	2,100	1,400	-
Cisco 3945E ISR	256 MB	2 GB	2,500	1,500	-
Cisco 4321 ISR	256 MB	512 MB	100	100	-
Cisco 4331 ISR	256 MB	512 MB	400	250	-
Cisco 4351 ISR	256 MB	2 GB	1,000	500	-
Cisco 4431 ISR	256 MB	4 GB	3,000	1,500	-
Cisco 4451 ISR	Refer to data sheet	8 GB	6,000	3,000	-
Cisco ASR 1001	Refer to data sheet	8 GB	10,000	3,600	-
Cisco ASR 1001-X	Refer to data sheet	8 GB	10,000	3,600	-
Cisco ASR 1002-X	Refer to data sheet	16 GB	12,000	3,600	-
Cisco ASR 1004 and ASR 1006 (RP2)	Refer to data sheet	16 GB	16,000	6,000	-

*These Router platforms support NANOCUBE licensing (see Licensing Section).

** Maximum trunk sessions mentioned are for simple trunk calls (7 messages per leg) and it may vary depending on call/message rate.

** Maximum sessions assumes that only the specified session is configured. Simultaneous use of other CUBE session types will reduce session maximums one for one.

Ordering Information

This product is orderable through bundles, as shown in Table 3.

Table 3. Cisco Unified Border Element (CUBE) ISR Bundles

Part Number (SKU)	Product Description	Technology Package	Additional HW	Flash/DRAM
C2901-VSEC-CUBE/K9	Cisco 2901 Voice Sec and CUBE Bundle, PVDM3-16, UC and SEC License P, FL-CUBEE-25	SL-29-UC-K9 and SL-29-SEC-K9	None	256MB 512MB
C2911-VSEC-CUBE/K9	Cisco 2911 Voice Sec and CUBE Bundle, PVDM3-16, UC and SEC License P, FL-CUBEE-25	SL-29-UC-K9 and SL-29-SEC-K9	None	256MB 512MB
C2921-VSEC-CUBE/K9	Cisco 2921 Voice Sec and CUBE Bundle, PVDM3-32, UC and SEC License P, FL-CUBEE-25	SL-29-UC-K9 and SL-29-SEC-K9	None	256MB 512MB
C2951-VSEC-CUBE/K9	Cisco 2951 Voice Sec and CUBE Bundle, PVDM3-32, UC and SEC License P, FL-CUBEE-25	SL-29-UC-K9 and SL-29-SEC-K9	None	256MB 512MB
C3925-VSEC-CUBE/K9	Cisco 3925 Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25	SL-39-UC-K9 and SL-39-SEC-K9	None	256MB 1GB
C3945-VSEC-CUBE/K9	Cisco 3945 Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25	SL-39-UC-K9 and SL-39-SEC-K9	None	256MB 1GB
C3925E-VSEC-CUBEK9	Cisco 3925E Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25	SL-39-UC-K9 and SL-39-SEC-K9	None	256MB 1GB
C3945E-VSEC-CUBEK9	Cisco 3945E Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25	SL-39-UC-K9 and SL-39-SEC-K9	None	256MB 1GB

This product is also orderable as an add-on or upgrade to Cisco routers by following three simple steps:

1. Select a Cisco router based on performance requirements (refer to Table 2).
2. Select a Cisco IOS Software image with Cisco Unified Border Element (CUBE) feature support (all IP Voice and later images support some components of the CUBE). On the Cisco 2900, 3900, and 3900E and 4000 Series platforms, which use a universal software image, select the Unified Communications package.
3. Select the appropriate Cisco IOS Software feature license.

To order software only, follow steps 2 and 3 and refer to Table 4 when selecting feature license part numbers.

Table 4. Ordering Information for Feature Licenses

Part Number (SKU)	Description
FL-CUBEE-5(=)	Feature license applicable to the Cisco 2900, 3900 and 4000 Series platforms for 5 simultaneous IP-to-IP Gateway sessions
FL-CUBEE-25(=)	Feature license applicable to the Cisco 2900, 3900 and 4000 Series platforms for 25 simultaneous IP-to-IP Gateway sessions
FL-CUBEE-100(=)	Feature license applicable to the Cisco 2900, 3900 and 4000 Series platforms for 100 simultaneous IP-to-IP Gateway sessions
FL-CUBEE-500(=)	Feature license applicable to the Cisco 2900, 3900 and 4000 Series platforms for 500 simultaneous IP-to-IP Gateway sessions
FL-CUBEE-1000(=)	Feature license applicable to the Cisco 3900 Series platforms for 1000 simultaneous IP-to-IP Gateway sessions
FL-CUBE-4(=)	Feature license applicable to the Cisco 2800 and 3800 Series platforms for 4 simultaneous IP-to-IP Gateway or Gatekeeper sessions
FL-CUBE-25(=)	Feature license applicable to the Cisco 2800 and 3800 Series platforms for 25 simultaneous IP-to-IP Gateway or Gatekeeper sessions
FL-CUBE-100(=)	Feature license applicable to the Cisco 2800 and 3800 Series platforms for 100 simultaneous IP-to-IP Gateway or Gatekeeper sessions
FLSASR1-CUE-100(=)	Cisco Unified Border Element 100 Sessions for ASR1000 Series
FLSASR1-CUE-500(=)	Cisco Unified Border Element 500 Sessions for ASR 1000 Series

Part Number (SKU)	Description
FLSASR1-CUE-1K(=)	Cisco Unified Border Element-1000 Sessions for ASR 1000 Series
FLSASR1-CUE-4K(=)	Cisco Unified Border Element 4000 Sessions for ASR 1000 Series
FLSASR1-CUE-16K(=)	Cisco Unified Border Element 16,000 Sessions for ASR 1000 Series
FL-NANOCUBE	NANOCUBE license available only for 800 Series ISR routers and SP-IAD 2900 routers
FL-SL-IPV-POL-100=	Voice Policy/Voice Security Feature License for 100 Sessions (additional SRE server blade required)
FL-SL-IPV-POL-1K=	Voice Policy/Voice Security Feature License for 1,000 Sessions (additional SRE server blade required)
FL-SL-IPV-POL-10K=	Voice Policy/Voice Security Feature License for 10,000 Sessions (additional SRE server blade required)

Downloading the Software

After ordering a feature license, visit the Cisco Software Center to download the Cisco IOS Software. Table 5 provides the software image name and software feature set available with each platform.

Table 5. CUBE Software Feature Set and Software File

Platform	Software Image Name	Software Feature Set
Cisco 2901, 2911, and 2921 platforms	c2900-universalk9-mz	Universal Image
Cisco 2951 platform	c2951-universalk9-mz	Universal Image
Cisco 3925 and 3945 platforms	c3900-universalk9-mz	Universal Image
Cisco 4321, 4331, and 4351 platforms	ISR4300-universalk	Universal Image
Cisco 4432 and 4451 platforms	isr4400-universalk	Universal Image
Cisco ASR 1000 Series Aggregation Services Router	SASR1R1-AESK9-21SR	Cisco ASR Advanced Enterprise Services

Note: Cisco 2800 and 3800 Series ISRs can support CUBE with Cisco IOS Software Release 15.1(4) M and earlier. The end-of-life notice for these platforms can be found at the following link: http://www.cisco.com/en/US/prod/collateral/routers/ps5853/qa_c67-631674_ps5854_Products_End-of-Life_Notice.html.

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 line-side service offerings. As such, CUBE is ideal for businesses of all sizes; it cost-effectively supports a variety of SIP services, whether with premises-based call control or 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 http://www.cisco.com/go/cube or contact your local Cisco account representative.

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