C3925-VSEC-CUBE/K9 Datasheet

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Overview

C3925-VSEC-CUBE/K9 is the Cisco 3925 router with Voice Sec and CUBE Bundle, including PVDM3-64, UC and SEC License PAK, and FL-CUBEE-25.

Quick Specs

Table 1 shows the Quick Specs of the C3925-VSEC-CUBE/K9.

Product Code	C3925-VSEC-CUBE/K9
Bundle	Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25
Services Performance Engine module	w/SPE 100
Rack Units	3U
Interfaces	3 integrated 10/100/1000 Ethernet ports with 2 ports capable of RJ-45 or SFP connectivity
Expansion Slot(s)	2 service module slots 1 Internal Services Module slot 4 onboard digital signal processor (DSP) slots 4 Enhanced High-Speed WAN Interface Card (EHWIC) slots
RAM	1 GB (installed) / 2 GB (max)
Flash Memory	256 MB (installed) / 8 GB (max)

Product Details

Figure 1 shows the appearance of the CISCO3925/K9. C3925-VSEC-CUBE/K9 is similar with it.

C3925-VSEC-CUBE/K9 provides Voice Sec and CUBE bundle.

Table 2 shows the Voice Security Bundle Features.

Authentication and Encryption Features	Media encryption of voice RTP streams using SRTP Exchange of RTP Control Protocol (RTCP) information using secure RTCP SRTP to RTP fallback for calls between secure and insecure endpoints Secure calls supported in Cisco Unified Survivable Remote Site Telephony (SRST) mode during WAN failover Compressed RTP (CRTP) supported with media encrypted calls using SRTP
Authentication and Encryption Algorithm	Supports AES-128 encryption algorithm Supports the HMAC secure hash authentication algorithm (SHA 1)

Signaling Authentication and Encryption Features	Gateway to Cisco Unified Communications Manager signaling and encryption uses IPSec for Media Gateway Control Protocol (MGCP), H.323 and SIP gateways IP phone to Cisco Unified Survivable Remote Site Telephony router signaling and encryption uses TLS (Transport Layer Security)
Protocol Support	MGCP 0.1 (supports MGCP gateways with Cisco Unified Communications Manager) H.323 (supported on H.323 gateways and CUBE; Cisco Unified Communications Manager interoperability is optional) Session Initiation Protocol (SIP) SCCP (Cisco Unified IP Phone) in SRST mode
Module Support	Any module that has PVDM2, PVDM3 and/or built-in DSP
Codec Support	G.711, G.729A, and G.729

Table 3 shows the 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 andSession Descriptor Protocol (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
Call routing and dialing options	E164-based dialing Uniform Resource Identifier (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
Cisco 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 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 recording	Media forking features for both voice and video to integrate with Cisco TelePresence Media Recording Servers Active (SIP-based) and passive (application programming interface [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 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 Adaptive Multirate (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
Security	Rogue SIP invite and rogue RTP packet detection Alerts for rogue packet activity IP Security (IPsec) Secure RTP (SRTP) Transport Layer Security (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 agent NANOCUBE 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
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 ISRs (Hot Standby Router Protocol [HSRP]-based) Note: Media is preserved for active calls at time of failover in each redundancy configuration listed.

Compare to Similar Routers

Table 4 shows the comparison of CISCO3925/k9 and C3925-VSEC-CUBE/K9.

Model	**CISCO3925/k9**	**C3925-VSEC-CUBE/K9**
Bundle	N/A	Cisco 3925 Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25
SPE	w/SPE 100	w/SPE 100
Interface	3 integrated 10/100/1000 Ethernet ports with 2 ports capable of RJ-45 or SFP connectivity	3 integrated 10/100/1000 Ethernet ports with 2 ports capable of RJ-45 or SFP connectivity
Slots	2 service module slots 1 Internal Services Module slot 4 onboard digital signal processor (DSP) slots 4 Enhanced High-Speed WAN Interface Card (EHWIC) slots	2 service module slots 1 Internal Services Module slot 4 onboard digital signal processor (DSP) slots 4 Enhanced High-Speed WAN Interface Card (EHWIC) slots
Rack Unit	3U	3U

Get more information

Do you have any question about the C3925-VSEC-CUBE/K9 router?

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Specification

C3925-VSEC-CUBE/K9 Specifications	
Manufacturer	Cisco Systems, Inc
Manufacturer Part Number	C3925-VSEC-CUBE/K9

Product Type	Router
Bundle	Cisco 3925 Voice Sec and CUBE Bundle, PVDM3-64, UC and SEC License P, FL-CUBEE-25
Form Factor	Desktop - modular - 3U
Connectivity Technology	Wired
Data Link Protocol	Ethernet, Fast Ethernet, Gigabit Ethernet
Network / Transport Protocol	TCP/IP, UDP/IP, RSVP, IPSec, PPPoE, L2TPv3
Routing Protocol	OSPF, IS-IS, BGP, EIGRP, DVMRP, PIM-SM, IGMPv3, GRE, PIM-SSM, static IPv4 routing, static IPv6 routing, policy-based routing (PBR)
Remote Management Protocol	SNMP, RMON
Encryption Algorithm	SSL, TLS
Authentication Method	RADIUS
Features	Firewall protection, NAT support, VPN support, MPLS support, Syslog support, content filtering, IPv6 support, Class-Based Weighted Fair Queuing (CBWFQ), Weighted Random Early Detection (WRED), Access Control List (ACL) support, Quality of Service (QoS), Dynamic Multipoint VPN (DMVPN)
Compliant Standards	IEEE 802.3, IEEE 802.1Q, IEEE 802.1ah, IEEE 802.1ag
DRAM Memory	1 GB (installed) / 2 GB (max)
Flash Memory	256 MB (installed) / 8 GB (max)
LED Status Lights Indicators	Link activity, power
Communications	
Type	Voice / fax module
Digital Ports Qty	64
Protocols & Specifications	ITU T.38
IP Telephony	
VoIP Protocols	H.323, SIP
Voice Codecs	G.711, G.722, G.723, G.728, G.729, G.729a, G.729ab, G.711u, G.711a, G.726, iLBC
IP Telephony Features	Call forwarding, automatic fax/modem detection and pass-through, call hold, call transfer, DTMF generation/detection
Connectivity Slots	
Router Interfaces	3 x 10Base-T/100Base-TX/1000Base-T - RJ-45 Management : 1 x console - RJ-45 Management : 1 x console - mini-USB Type B Serial : 1 x auxiliary - RJ-45 USB : 2 x 4 pin USB Type A 2 x SFP (mini-GBIC)
Expansion Slot(s)	4 (total) / 4 (free) x EHWIC 4 (total) / 3 (free) x PVDM 2 (total) / 1 (free) x CompactFlash Card 1 (total) / 1 (free) x ISM
Power Supply	
Power Device	Power supply - internal
Voltage Required	AC 120/230 V (50/60 Hz)
Dimensions / Weight / Miscellaneous	

Approximate Width	18.7 in
Depth	17.2 in
Height	5.2 in
Weight	39 lbs
Compliant Standards	CISPR 22 Class A, CISPR 24, EN55024, EN55022 Class A, EN50082-1, AS/NZS 60950-1, AS/NZ 3548 Class A, CAN/CSA-E60065-00, ICES-003 Class A, FCC CFR47 Part 15, EN300-386, UL 60950-1, IEC 60950-1, EN 60950-1, CSA C22.2 No. 60065, BSMI CNS 13438
System software	
Software Included	Cisco IOS Unified Communications, Cisco IOS Security, Cisco Unified Border Element Enterprise Edition (25 sessions)

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