Mediant™ CE/VE/SE

Mediant CE/VE/SE Software Session Border Controller (SBC)

AudioCodes **Mediant software session border controller (SBC)** is a highly scalable SBC solution supporting broad SIP interoperability, advanced media handling and robust security. AudioCodes Mediant software SBC enables enterprises and service providers to deliver voice services, such as SIP trunking and unified communications, via private or public clouds.

The Mediant software SBC is available in three variants to meet different customer deployment needs:

Mediant CE | a cloud-native SBC delivering high scalability and elasticity in virtualized cloud environments
Mediant VE | built for deployment in virtualized data centers, public clouds and NFV environments
Mediant SE | designed to run on commercial off-the-shelf servers (COTS) in high-scale communications environments

All Mediant software SBC variants provide the following:

Comprehensive SBC functionality and SIP interoperability
Shared code base with AudioCodes field-proven, hardware-based SBCs

Rapid cloud deployment
Optimized resource consumption for private and public clouds such as OpenStack and Amazon Web Services

NFV-ready
Proven interoperability with leading NFV orchestrators

Enhanced scalability
Easily scale from tens up to tens of thousands of concurrent sessions

High availability
1:1 active-standby configuration for business continuity

High performance and robust security
Built-in software-based media transcoding with support for encryption and protection from attacks

Qualified for leading UC and hosted telephony platforms
Supported environments include Microsoft Teams, Skype for Business and BroadSoft BroadWorks

Integrated WebRTC gateway
Simple and secure WebRTC deployment, supporting both signaling and media

Specifications

Capacities	Mediant CE	Mediant VE	Mediant SE
Max. Signaling Sessions	50,000	24,000	70,000
Max. Media Sessions	50,000	24,000	70,000
Max. SRTP-RTP Sessions	50,000	10,000	40,000
Max. Transcoding Sessions	27,000	12,000 *	30,000*
Max. Registered Users	100,000	75,000	500,000

Security	
Access Control	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
Encryption and Authentication	TLS, DTLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
Privacy	Topology hiding, user privacy
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access

Interoperability	
SIP B2BUA	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
SIP interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server
Transport Mediation	SIP over UDP/TCP/TLS/WebSocket, IPv4 / IPv6, RTP / SRTP (SDES/DTLS)
Header Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
URI and Number Manipulations	URI user and host name manipulations, ingress and egress digit manipulation
Transcoding and Vocoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729A/B, GSM-FR, AMR-NB, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion
WebRTC Controller	Interworking between WebRTC devices and SIP networks. Supports WebSocket, Opus, VP8 video coder, lite ICE, DTLS, RTP multiplexing, secure RTCP with feedback
NAT	Local and far-end NAT traversal for support of remote workers

Voice Quality and SLA	
Call Admission Control	Based on bandwidth, session establishment rate, number of connections/registrations
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintains local calls in the event of WAN failure.
Impairment Mitigation	Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection
Voice Enhancement	Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control
Direct Media	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption while avoiding media anchoring
Voice Quality Monitoring	RTCP-XR, AudioCodes Session Experience Manager (SEM)
High Availability	SBC high availability with two-box redundancy, active calls preserved
Quality of Experience	Access control and media quality enhancements based on QoE and bandwidth utilization
Test agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs

SIP Routing	
Routing Methods	Request URL, IP address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
Redundancy	Detection of proxy failures and subsequent routing to alternative proxies
Routing Features	Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization
SIPRec	IETF standard SIP recording interface

Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS, HTTP reverse proxy
Multi Tenancy	Advanced multi-tenant SBC partitioning
Deployment tools	VNFM/Stack manager (Mediant CE), HEAT templates, Cloud Formation
Auto-scaling (CE)	Automatic, REST API, CLI, Web UI

Mediant VE SBC Minimum Requirements			
Hypervisor	VMware® vSphere ESXi™ 5.x, Linux KVM, Microsoft Hyper-V	Virtual Resources	1 vCPU; 2 GB RAM; 10 GB Disk; Virtual NICS - 2 (Standalone)/3 (HA)

* With media transcoding cluster

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC) is a leading vendor of advanced voice networking and media processing solutions for the digital workplace. With a commitment to the human voice deeply embedded in its DNA, AudioCodes enables enterprises and service providers to build and operate all-IP voice networks for unified communications, contact centers and hosted business services. AudioCodes' wide range of innovative products, solutions and services are used by large multinational enterprises and leading tier one operators worldwide.

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