

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



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 Microsoft Azure and Amazon Web Services, Google Cloud Platform



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

Certified SBC for Teams Direct Routing supporting media optimization



Integrated WebRTC gateway

Simple and secure WebRTC deployment, supporting both signaling and media

Specifications

Capacities			
	Mediant CE	Mediant VE	Mediant SE
Max. Signaling Sessions	40,000	24,000	70,000
Max. Media Sessions	40,000	24,000	70,000
Max. SRTP-RTP Sessions	40,000	10,000	40,000
Max. Transcoding Sessions	27,000	12,000 ¹	30,000 ¹
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	Automatic 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		
STIR/SHAKEN	STIR/SHAKEN support. Interworking with STI-AS/VS		
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/SCTP, 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 Gateway	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 One Voice Operations Center (OVOC)		
High Availability	SBC 1+1 high availability with active calls preservation		
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		
SBC Media Types	Audio(Video)Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)		
Recording Solutions	Lawful Interception (LI ²), SIPREC for both audio and video sessions		
Management			
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, HTTP reverse proxy One Voice Operations Center (OVOC), Session Detail Records (SDRs)		
Multi-Tenancy	Advanced multi-tenant SBC partitioning		
Deployment Tools	VNF/Stack manager (Mediant CE), HEAT templates, Cloud Formation		
Auto-scaling CE	Automatic, REST API, CLI, Web UI		
Cloud Environments			
Public Cloud	Azure, AWS, GCP		
Private Cloud	OpenStack, VMware® vSphere		
Mediant VE SBC Minimum Requirements			
Hypervisor	VMware® vSphere ESXi™ 5.5, & 6.5 and above, Linux KVM, Microsoft Hyper-V	Virtual Resources	1 vCPU; 2 GB RAM; 10 GB Disk; Virtual NICS - 2/3 (Standalone/HA)

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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¹ With media transcoding cluster
² Requires a dedicated software build