

AudioCodes Session Border Controller (SBC) Products

Mediant™ VE

Session Border Controller (SBC)

Benefits

- Meets demands for datacenter infrastructure harmonization and NFV
- Certified by Miercom for high performance and scalability under security attacks including Denial of Service, malformed SIP messages and rogue RTP packets
- Offers comprehensive interoperability and enhanced voice quality
- Deployable on private and public clouds such as OpenStack and Amazon Web Services (AWS)
- Proven interoperability with 3rd party NFV orchestration solutions
- Rapidly scale session capacity and quickly deploy new instances with AudioCodes' cloud licensing

Key Features

- Same code base as AudioCodes field-proven hardware-based SBCs
- Runs as a VNF in an NFV environment both on a virtual CPE and within service provider datacenters
- Runs on dedicated COTS servers and in virtualized environments
- High packet throughput through optimized network path
- Advanced voice quality monitoring and reporting
- Built-in media transcoding capability in the software
- Qualified for Microsoft Skype for Business/Lync SBC and BroadSoft BroadWorks environments
- Embedded signaling and media encryption hardware
- Media replication for recording through SIPREC
- High-availability 1:1 active-standby configuration ensures business continuity

AudioCodes **Mediant Virtual Edition (VE) Session Border Controller (SBC)** is designed to meet today's demands of enterprises and service providers looking to virtualize their infrastructure and harmonize their datacenters on commodity server hardware. The Virtual Edition SBC offers a comprehensive and scalable solution supporting SIP interoperability, media handling (including transcoding) and security.



Hypervisors: VMware, Hyper-V and KVM

Cloud environments: Openstack and Amazon Web Services (AWS)

The Mediant VE SBC was designed to meet ETSI NFV ISG requirements. It runs as an SBC VNF on leading NFV platforms and vE-CPE devices, as well as native servers, and is compatible with leading hypervisors and orchestration solutions. By offering a single scalable product, covering all capacity needs with one unified control and management interface, service providers can leverage its deployment and maintenance simplicity to introduce new communications services rapidly and cost-effectively.

Extensive Mediation Capabilities and Proven Interoperability

The Mediant VE SBC includes comprehensive media security and SIP normalization capabilities. It offers full interoperability with an extensive list of IP-PBXs, unified communications solutions and SIP trunking provider networks.

Security

The Mediant VE SBC provides robust protection for the IP communications infrastructure, preventing fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The Mediant VE SBC offers active/standby high availability and maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities result in minimum communications downtime.

Applications

- SIP trunking
- Hosted PBX & UC-as-a-Service (UCaaS)
- IP contact centers
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems



Mediant™ VE

SPECIFICATIONS

Capacities	Virtualized	Bare Metal	
Max. Signaling Sessions	6,000	32,000	
Max. Media Sessions	6,000	24,000	
Max. SRTP-RTP Sessions	4,000	16,000	
Max. Transcoding	350	-	
Max. Registered Users	30,000	120,000	
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 for SBC users		
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.729, GSM-FR, AMR-NB/WB, 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		
Stand Alone 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 (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Voice Quality Monitoring	RTCP-XR, AudioCodes Session Experience Manager (SEM)		
High Availability (Redundancy)	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		
Multi Tenancy	Advanced multi-tenant SBC partitioning		
Mediant VE SBC Minimum Requirements			
Hypervisor	VMware® vSphere ESXi™ 5.x, Linux KVM, Microsoft Hyper-V	Virtual NICs	2 (Standalone) 3 (High Availability)
Memory	2 GB	Virtual CPUs	1
Disk space	10 GB		

ABOUT AUDIOCODES

AudioCodes Ltd. (NasdaqGS: AUCD) designs, develops and sells advanced Voice over IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader focused on converged VoIP & data communications and its products are deployed globally in Broadband, Mobile, Enterprise networks and Cable. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers and Value Added Applications. AudioCodes' underlying technology, VoIPerfect HDTM, relies on AudioCodes' leadership in DSP, voice coding and voice processing technologies. AudioCodes High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.

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