AudioCodes Session Border Controller (SBC) Products

Mediant™ VE/SE 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 universal 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 uCPE 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.

AudioCodes **Mediant Server Edition (SE) Session Border Controller (SBC)** runs on AudioCodes certified Commercial Off-the-Shelf (COTS) servers, aimed at high-scale environments.

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

AudioCodes

Mediant™ VE/SE

Specifications

Capacities	Mediant VE	Mediant SE
Max. Signaling Sessions	10,000	32,000
Max. Media Sessions	10,000	24,000
Max. SRTP-RTP Sessions	10,000	16,000
Max. Transcoding	5,000 (media transcoding cluster)	-
Max. Registered Users	75,000	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, 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	
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		

Contact AudioCodes or an authorized AudioCodes reseller for a list of recommended server specifications

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC) 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 and 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, Value Added Applications and Professional Services. AudioCodes' underlying technology, VoIPerfectHD™, 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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