AudioCodes Session Border Controller (SBC) Products

Mediant 9000 Session Border Controller

The **AudioCodes Mediant 9000 Session Border Controller (SBC)** is a high capacity member of AudioCodes' field-proven hardware-based SBC products, designed to offer service providers and enterprises a scalable SBC solution. The Mediant 9000 SBC supports wide-ranging SIP interoperability, delivering service assurance and enabling scalable, reliable and secured connectivity between different VoIP networks.

The Mediant 9000 SBC provides a perfect solution for service providers and large organizations such as contact centers, large data centers, hosted services and government institutions, where security, reliability and high performance are critical.

Benefits

- High Capacity SBC for Service Providers and Large Enterprise Deployments
- Offers comprehensive security, interoperability and reliability
- Delivers high service performance and voice quality
- Flexible licensing options for cost-effective scalability

Key Features

- Scalable to tens of thousands of SBC sessions
- Extensive SIP mediation capabilities
- Supports remote workers and mobile SIP clients
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Branch survivability during WAN failure
- Active/Standby High Availability

Extensive Mediation Capabilities and Proven Interoperability

The Mediant 9000 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 9000 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 9000 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
- IP contact centers
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems
- Residential VoIP

audiocodes

Mediant 9000

Specifications

Capacities			
Max. Signaling Sessions	50,000	Max. Media Sessions	50,000
Max. Registered Users	180,000	Max. SRTP-RTP Sessions	18,000
Max. Transcoding	25,000 (media transcoding cluster)		
Network Interfaces			
Ethernet	12x1Gb Ethernet ports		
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/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)		
Message 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		
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 (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		
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		
Physical / Environmental			
Dimensions	434.5mm x 622.3mm x 29.7mm (HxWxD)	Weight	19.2 kg (42.3 lb)
Mounting	19" mount	Operating Temperature	10° to 35°C
Power	Dual redundant 100-240V AC power supply/ Dual redundant -48 VDC power supply		
Regulatory Compliance			
FCC rating and Carrier grade	Class A, NEBS (GR-63-CORE & GR-1089-CORE) and ETSI certified		
Normative standards	CISPR 22; EN 55022; EN 55024; FCC CFR 47, Pt 15; ICES-003; CNS13438; GB9254; K22; K24; EN 61000-3-2; EN 61000-3-3; EN 60950-1; IEC 60950-1		

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.

International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive,
Somerset, NJ 08873
Tel:+1-732-469-0880
Fax:+1-732-469-2298

Contact us: www.audiocodes.com/contact
Website: www.audiocodes.com

Ref. # LTRM-30034 02/18 V.7