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

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Specifications

Capacities			
Max. Signaling/Media Sessions	24,000		
Max. SRTP-RTP Sessions	16,000		
Max. Registered Users	120,000		
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		
Encryption and Authentication	TLS, 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 OAM interfaces		
Intrusion Detection System	Detect and mitigate VoIP attacks, prevent Theft of Service and unauthorized access.		
Interoperability			
SIP B2BUA	Full SIP transparency, mature & broadly deployed SIP stack		
SIP interworking	3xx redirect, REFER, PRACK, Session Timer, Early media, Call hold, Delayed offer		
Registration	Registration and authentication on behalf of an IP-PBX		
Transport Mediation	SIP over UDP to SIP over TCP or SIP over TLS, IPv4 to IPv6, RTP to SRTP		
Header Manipulation	Ability to add/modify/delete headers using advanced regular expressions		
URI and Number Manipulations	URI User and Host name manipulations. Ingress & Egress Digit Manipulation		
Coder normalization	Coder enforcement and re-prioritization		
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	Maintain local calls in the event of WAN failure		
Transparent Media	Low latency, unprocessed payload transfer		
Media De-anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Redundancy	High availability with two box redundancy, Active calls preserved		
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)		
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		
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, Coder type etc)		
Redundancy	Detect proxy failures and route to alternative proxies		
Routing Features	Least cost routing, call forking, load balancing		
SIPRec	IETF standard SIP recording interface		
Physical / Environmental			
Dimensions	1U x 434mm x 698mm (HxWxD)	Weight	19.2 kg (42.3 lb)
Mounting	19” mount	Power	Dual redundant 100-240V AC power supply/ Dual redundant -48 VDC power supply
Operating Temperature	10° to 35°C		
Regulatory Compliance			
FCC rating	Class A		
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		
Carrier grade	NEBS (GR-63-CORE & GR-1089-CORE) and ETSI certified		

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 & 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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