AudioCodes Session Border Controller (SBC) Products

Mediant™ 3000 Hybrid E-SBC and Media Gateway

The **AudioCodes Mediant 3000 Enterprise Session Border Controller (E-SBC)** and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 3000 connects IP-PBXs to any SIP trunking service provider, scaling up to 1008 concurrent SBC sessions. It offers superior performance in connecting any SIP to SIP environment, legacy TDM-based PBX systems to IP networks, and IP-PBXs to the PSTN, supporting up to 2016 voice channels in a modular 1U platform.

Benefits

- Fully integrated device for secured SIP trunking and PSTN access
- Hybrid SBC and Media Gateway platform lowers CAPEX and reduces space and power footprints
- Software license scalability from 252 up to 1008 SBC sessions
- Extensive interoperability and partnerships that extend across multiple vendor devices and protocol implementations
- Offers comprehensive security, interoperability and reliability
- Delivers high service performance and voice quality
- Branch office survivability in the event of a WAN Outage

Key Features

- Rich and powerful SIP normalization and routing mechanisms for seamless interoperability
- Hybrid SBC enables seamless migration and PSTN fallback
- Perimeter defense against denial of service, fraud and eavesdropping
- Carrier-Grade Simplex and High-Availability configurations
- VoIP quality monitoring and enforcement
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.

Vast mediation capabilities and proven interoperability

The Mediant 3000 supports a wide range of voice coders and is capable of transcoding between narrowband and wideband voice coders, providing SIP normalization, fax handling, gain control and numerous additional media processing features. It offers certified interoperability with leading unified communications solutions and SIP trunking providers.

Security

The Mediant 3000 provides robust protection for the IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The Mediant 3000 E-SBC supports high-availability configurations with reliable, “1+1” redundancy of all system components, ensuring no loss of active sessions during failure time. The Mediant 3000 maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities (including PSTN fallback with E911) 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

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Mediant™ 3000

Specifications

Capacities			
Max. Signaling/Media Sessions	1008	Max. Transcoding Sessions	1008
Max. SRTP/RTP Sessions	882	Max. Registered Users	Up to 3000 or 5000 (HW config dependent)

Telephony Interfaces	
PSTN	1 OC-3 or STM-1 APS optical links, 1 to 3 T3 (DS3) electrical links, up to 63/84 E1/T1 links
Network Interfaces	
Ethernet	Dual Redundant 100/1000 Base-T Ethernet ports and additional two Dual Redundant 100 Base-T Ethernet ports for OEM and Control (Available on the E1/T1 configuration only)
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/Authentication	TLS, SRTP, HTTPS, SSH, IPSec, IKE, SNMPv3, Client/Server SIP Digest authentication, RADIUS Digest
Privacy	Topology Hiding, User Privacy
Traffic Separation	Physical separation (on E1/T1 configuration only) or VLAN 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, V.34 Fax
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
Signal Conversion	DTMF/RFC 2833, Inband/T.38 Fax, Packet-time Conversion, V.150.1
NAT	Local and Far End NAT traversal for support of remote workers
Transcoding and Vocoders	Coder normalization, including transcoding, coder enforcement and re-prioritization. Extensive vocoder support: Wireline: G.711a/mu, G.723.1, G.726, G.727, G.729A/B/E, EG.711; Wireless:GSM-FR, GSM-EFR, MS-GSM, AMR, iLBC, EVRC, EVRC-B; Wideband: AMR-WB, G.722, SPEEX, RTA
ITSP and PBX support	Interoperable with many SIP trunk Service Providers and PBX vendors, such as AT&T, Verizon and Microsoft Lync
Hybrid PSTN mode	Connect to TDM PBXs or PRI/CAS trunks for least-cost routing or fallback. Also useful for gradual enterprise migration to SIP. Support for T1/E1/J1, T3, OC-3, STM-1 physical interfaces
Signal Detection	Voice Activity, Call Progress Tone, and Answering Machine
Voice Quality and SLA	
Call Admission Control	Based on bandwidth, session establishment rate, number of connections/registrations
Standalone Survivability	Maintain local calls in the event of WAN failure
Transparent Media	Low latency, unprocessed payload transfer
Impairment Mitigation	Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/ Comfort Noise Generation, RTP redundancy, broken connection detection
Media De-anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)
Test agent	Remote verification of connectivity, voice quality and SIP message flow between SIP UAs
Gain control	Fixed & dynamic voice gain control
Redundancy	Carrier grade “Five 9s” availability with 1+1 hardware redundancy, Active calls preserved
SIP Routing	
Routing Methods	Request URL, IP Address, FQDN, ENUM, advanced LDAP
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, Coder type etc)
Routing Features	Least cost routing, call forking, load balancing
SIPRec	IETF standard SIP recording interface
Hardware Specifications	
Dimensions	(HxWxD) 88mm x 482.6mm x 296.8mm, 4-slot, 2U cPCI chassis
Weight	Approx. 35.27 lb (16 kg), fully loaded
Power	48 V DC Dual Feed, with up to 2 DC Power modules or 100–240 V AC redundant Dual Feed
Regulatory Compliance	
Telecommunication Standards	FCC part 68, TBR4 and TBR13
Safety and EMC Standards	UL 60950 FCC part 15 Class A CE Mark (EN55022 Class A, EN60950, EN 55024, EN300 386)
Environmental Specifications	NEBS level 3: GR-63-Core, GR-1089-Core, Type 1 & 3, ETS300 019

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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