

AudioCodes Session Border Controller (SBC) Products

Mediant™ 1000 Hybrid E-SBC and Media Gateway



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
- Scalable “pay-as-you-grow” modular architecture
- 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
- Modular support for analog and digital TDM interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- Optional Open Solution Network (OSN) server module for hosting value-added applications

The **AudioCodes Mediant 1000 Enterprise Session Border Controller (E-SBC)** and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 1000 connects IP-PBXs to any SIP trunking service provider, scaling up to 150 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 192 voice channels in a modular 1U platform.

Vast mediation capabilities and proven interoperability

The Mediant 1000 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 1000 provides robust protection for IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The Mediant 1000 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

Mediant™ 1000

SPECIFICATIONS

Capacities			
Max. Signaling/Media Sessions	150	Max. Transcoding Sessions	96
Max. SRTP/RTP Sessions	120	Max. Registered Users	600
Telephony Interfaces			
Modularity and Capacity	6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)		
Digital Module	Up to 6 E1 or 8 T1/J1 spans provided on trunk modules. Each module supports 1, 2, or 4 E1/T1/J1 spans, with an option of PSTN fallback		
Digital PSTN Protocols	Supporting various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. It also supports different variants of CAS protocols, including MFC R2, E&M immediate start, E&M delay dial / start and others.		
BRI Module	Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN fallback. Providing S/T interfaces; NT or TE termination; 2W per port (power supplied)		
Analog Module	Up to 24 FXS/FXO interfaces, provided on 4 ports FXO / FXS modules, ground / loop start		
Media Processing Module	Up to 4 Media Processing modules (MPM), providing additional DSP resources		
Network Interfaces			
Ethernet	Up to 6 GE interfaces configured in 1+1 redundancy or as individual 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/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, 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: NarrowBand: G.711a/mu, G.723.1, G.729A/B, iLBC, EVRC, AMR, G.726, G.727, GSM FR, MS GSM, GSM EFR, QCELP. Wideband: G.722		
Media Processing	60 Conferencing legs (3 way or N-way), play, record to IP or PSTN using Netann (RFC4240), MSCML (RFC5022)		
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		
SIP Routing			
Routing Methods	Request URL, IP Address, FQDN, ENUM, advanced LDAP		
Advanced Routing Criteria	QoS, bandwidth, SIP message (SIP request, Coder type etc)		
Routing Features	Least cost routing, call forking, load balancing		
OSN Server Platform			
Single Chassis Integration	Embedded, Open Network Solution Platform for third-party services		
Memory	Up to 8GB RAM	Storage	HHD or SSD
Hardware Specifications			
Dimensions	1U x 320mm x 345mm (HxWxD)		
Weight	Approx. 5.95lb (2.7kg) installed with OSN		
Mounting	Desktop or 19" mount		
Power	Single power supply 100-240V, 50-60 Hz, 1.5A max. optional redundant power supply		
Environmental	Operational: 0 to 40° C (32 to 104° F); Storage: -20 to 70° C (-4 to 158° F) Relative Humidity: 10 to 85% non-condensing		
Regulatory Compliance			
Telecommunication Standards	TIA/EIA-IS-968, TBR-4, TBR-13, and TBR-21		
Safety and EMC Standards	UL60950-1; FCC 47 CFR part 15 Class B CE Mark (EN55022 Class B, EN60950-1, EN55024, EN300 386, EN61000-3-2/3/3)		
Environmental Specifications	ETS 300019-2-1 Storage T1.2, ETS 300019-2-2 Transportation T2.3 ETS 300019-2-3 Operating T3.2		

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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