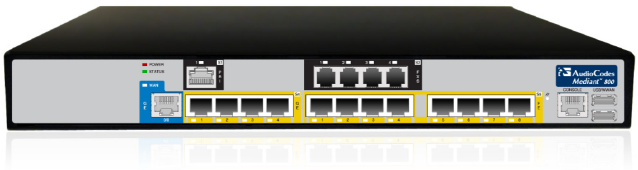
AudioCodes Session Border Controller (SBC) Products

Mediant™ 800

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
- 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 that connects to PSTN / PBX trunks for fallback and gradual enterprise migration to SIP
- Support for analog (FXO, FXS) and digital (PRI, BRI) interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- High Availability using two box redundancy
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- Optional Open Solution Network (OSN) Platform for hosting value-added applications

AudioCodes

The AudioCodes **Mediant 800 Enterprise Session Border Controller (E-SBC)** and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 800 connects IP-PBXs to any SIP trunking service provider, scaling up to 250 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 60 voice channels in a 1U platform.

Vast mediation capabilities and proven interoperability

The Mediant 800 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 800 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 800 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 (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™ 800

Specifications

Capacities			
Max. Signaling/Media Sessions	250		
Max. SRTP/RTP Sessions	180		
Max. Transcoding Sessions	45		
Max. Registered Users	800		
Telephony Interfaces			
Analog	4/8/12 FXS ports; 4/8/12 FXO ports		
Digital	1/2 span E1/T1; 4/8 BRI ports, network S/T interfaces, NT or TE termination		
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		
Clock Source	5 ppm High Precision		
Networking Interfaces			
Ethernet	4 GE or 4 GE + 8 FE 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 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, 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: SILK, G.711a/mu, G.723.1, G.729A/B, iLBC, AMR, G.726. Wideband: G.722, AMR-WB and SILK WB		
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		
Intelligent Voice	Multiple queues for granular prioritization of VoIP over other non-real time traffic types, Integrated Queuing and scheduling schemes (Strict Priority, Class based Prioritization queuing, fairness)		
Standalone Survivability	Maintain local calls in the event of WAN failure. Outbound calls use PSTN Fallback for external connectivity (including E911)		
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		
Voice Enhancement	Transrating, RTCP-XR, Acoustic echo cancellation		
Media De-anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)		
Redundancy	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		
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		
Multiple LANs	Support for up to 12 separate LANs		
SIPRec	IETF standard SIP recording interface		
OSN Server Platform (Optional)			
Single Chassis Integration	Embedded, open Network Solution Platform for third-party services		
Memory	Up to 16 GB RAM	Storage	HHD or SSD
Physical / Environmental			
Dimensions	1U x 320mm x 345mm (HxWxD)	Weight	Approx. 5.95lb (2.7kg) loaded with OSN
Mounting	Desktop or 19" rack mount	Power	100-240V 1.5A 50-60 Hz
Operating Temperature	5°-40° C		
Regulatory Compliance			
Telecommunications	TIA/EIA-IS-968 (FXO, T1) interface, ETSI ES203 021 (FXO interface), TBR-4 (ISDN over E1 interface), TBR13/13 (E1 lines), TBR-3 (BRI interface)		
Safety and EMC	IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 386		
Environmental Storage	ETS300019-2-1 class T1.2		
Transportation	ETS300019-2-2 class T2.3	Operating	ETS300019-2-3

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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