Mediant™ 800

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 800 enterprise session border controller (E-SBC)** and media gateway offers a complete connectivity solution for small-to-medium sized enterprises.

Scaling up to 400 concurrent sessions, the Mediant 800 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 800 supports up to 124 voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN.

400 SBC Sessions I 124 TDM Sessions I 1+1 High Availability I Certified SBC for Teams Direct Routing I Supports OPUS and SILK

Comprehensive interoperability
Proven interoperability with SIP trunks, SIP platforms and IP cloud services

Hybrid functionality
True hybrid SBC and gateway platform for gradual migration, low CAPEX and reduced space and power footprints

Enhanced security
Robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft

Superior voice quality
Advanced capabilities for optimizing and monitoring voice service quality

High resiliency
High availability using 1+1 redundancy, local branch survivability and PSTN fallback

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Specifications

Capacities				
	Max. Signaling	**Max. RTP/SRTP Sessions**	**Max. Transcoding Sessions**	**Max. Registered Users**
Mediant 800B	250	250/250	57	1500
Mediant 800C	400	400/250	114	2000

Telephony Interfaces	
Analog	4 FXS/FXO ports
Digital	Up to 4 E1/T1 interfaces; 4/8 BRI Ports
Clock Source	5 ppm High Precision
Digital PSTN Protocols	Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS- 100 and others. Different CAS protocols, including MFC R2, E&M immediate start, E&M delay dial/start and others.
Network 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 (Intrusion Detection System)
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	Automatic 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 and more
Registration and Authentication	SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
Transport Mediation	Mediation between SIP over UDP/TCP/TLS/WebSocket, IPv4/IPv6, RTP/SRTP (SDES/DTLS)
Header Manipulation	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
Number 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.729A/B, GSM-FR, AMR-NB, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB, iLBC
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, DTLS, RTP multiplexing.
NAT	Local and far-end NAT traversal for support of remote workers, ICE full and lite support (RFC 8445)
Voice Quality and SLA	
Call Admission Control	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
Voice Monitoring and Enhancement	Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, fixed and dynamic voice gain control, packet loss concealment, dynamic programmable jitter buffer, silence suppression/comfort, noise generation, RTP redundancy, broken connection detection
Direct Media	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
High Availability	SBC high availability with two-box redundancy, active calls preserved
Test Agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Call Handling	
Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
Querying External Databases	Destinations based on customized queries of ENUM, LDAP, HTTP server (REST API)
Available Destinations	Configured SIP peers, registered users, IP address, request URI
Advanced Features	Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization
SBC Media Types	Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)
SIPREC	IETF standard SIP recording interface, supporting both audio and video SBC sessions
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC)

Physical/Environmental			
Dimensions	1U x 345mm x 320mm (HxWxD)	**Weight**	Approx. 5.95lb (2.7kg) loaded with OSN
Mounting	Desktop or 19" rack mount	**Operating Temperature**	5°-40° C
Power	Internal AC power supply rated: 100-240 VAC ~50- 60Hz 1.5A maximum (Optional) Additional 12V 10A DC power, via an AudioCodes external AC/DC power adaptor		

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

80 Kingsbridge Rd - Piscataway, NJ 08854
Tel: +1-732-469-0880

www.audiocodes.com

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