

### 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 | 124 TDM Sessions | 1+1 High Availability | 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

## Specifications

Capacities				
	Max. Signaling	Max. RTP/SRTP Sessions	Max. Transcoding Sessions	Max. Registered Users
Mediant 800B	250	250/250	57	1500
Mediant 800C	400	400/300	114	2000
Telephony Interfaces				
Analog	4/8/12 FXS ports; 4/8/12 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, V34, packet-time conversion, V.150.1			
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, lite ICE, DTLS, RTP multiplexing.			
NAT	Local and far-end NAT traversal for support of remote workers			
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, RTPC-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			



**International Headquarters**  
 1 Hayarden Street, Airport City  
 Lod 7019900, Israel  
 Tel: +972-3-976-4000  
 Fax: +972-3-976-4040

**AudioCodes Inc.**  
 200 Cottontail Lane, Suite A101E,  
 Somerset, NJ 08873  
 Tel:+1-732-469-0880  
 Fax:+1-732-469-2298

Contact us: [www.audiocodes.com/contact](http://www.audiocodes.com/contact)  
 Website: [www.audiocodes.com](http://www.audiocodes.com)

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