

### Hybrid SBC and Media Gateway

The AudioCodes **Mediant 500 enterprise session border controller (E-SBC)** and media gateway is a compact, high performance VoIP connectivity solution for small enterprises and branch office locations.

Scaling up to 250 concurrent sessions, the Mediant 500 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

The Mediant 500 also supports up to 30 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.



250 SBC Sessions | 30 TDM Sessions | 1+1 High Availability | WebRTC Gateway



#### Comprehensive interoperability

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



#### Hybrid functionality

True hybrid SBC and gateway platform for gradual migration 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 Ethernet redundancy

## Specifications

Capacities			
Max. Signaling	250	Max. RTP/SRTP Sessions	200
Max. Registered Users	1,500		
Telephony Interfaces			
Digital	Single E1/T1 interface		
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 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, 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 OAMP interfaces		
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		
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users		
Transport Mediation	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP (SDES)		
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		
SIP Interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer		
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion		
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 Routing			
Routing Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth		
Querying External Databases	Routing based on customized queries of ENUM, LDAP, HTTP server (REST API)		
Route To	Configured SIP peers, registered users, IP address, request URI		
Advanced Routing Features	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization		
SIPREC	IETF standard SIP recording interface		
Management			
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, OVOC		
Physical/Environmental			
Dimensions	43.7 (1U) x 310 x 210 mm (HxWxD)		
Weight	4.4 lb (2.0kg)		
Mounting	Desktop or 19" rack mount		
Power	100-240V, 50-60 Hz, 0.8A		
Environmental	Operational: 0 to 40°C (41 to 104°F); Storage: -25 to 70°C (-13 to 185°F) Relative Humidity: 10 to 90% non-condensing		



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

**AudioCodes Inc.**  
 27 World's Fair Drive,  
 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)

©2018 AudioCodes Ltd. All rights reserved. AudioCodes, AC, HD VoIP, HD VoIP Sounds Better, IPmedia, Mediant, MediaPack, What's Inside Matters, OSN, SmartTAP, User Management Pack, VMAS, VoIPerfect, VoIPerfectHD, Your Gateway To VoIP, 3GX, VocaNom, AudioCodes One Voice and CloudBond are trademarks or registered trademarks of AudioCodes Limited. All other products or trademarks are property of their respective owners. Product specifications are subject to change without notice.