Mediant™ 500L

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 500L 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 60 concurrent sessions, the Mediant 500L connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 500L supports up to 8 voice channels 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.

60 SBC Sessions | 8 TDM Sessions | Branch Survivability | 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
Local branch survivability and PSTN fallback with E911

Mediant™ 500L

Specifications

Capacities			
Max. Signaling	60	**Max. RTP/SRTP Sessions**	60
Max. Registered Users	200		

Telephony Interfaces	
Digital	1-4 BRI ports, network S/T interfaces, NT or TE termination
Analog	Up to 4 FXS and 4 FXO ports
Clock Source	5 ppm High Precision
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 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, 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, 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
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
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, EMS
Physical/Environmental	
Dimensions	51 x 296 x 160 mm (2 x 11.65 x 6.3 in.) (HxWxD)
Weight	670g
Mounting	Desktop
Power	Single universal AC power supply 100-240V, 50-60 Hz, 12V/3A or 12V/5A
Environmental	Operational: 5 to 40° C (41 to 104°F); Storage: -25 to 85°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
Website: www.audiocodes.com