Mediant™ 500L

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 enables seamless migration and PSTN fallback
- Support for BRI interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement

The AudioCodes **Mediant 500L Enterprise Session Border Controller (E-SBC)** and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 500L connects IP-PBXs to any SIP trunking service provider, scaling up to 60 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 4 BRI interfaces.

Vast mediation capabilities and proven interoperability

The Mediant 500L 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 500L 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 500L offers 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

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Specifications

Capacities			
Max. Signaling/Media Sessions	60	Max. SRTP/RTP Sessions	45
Max. Registered Users	200		
Telephony Interfaces			
Digital	1-4 BRI ports, network S/T interfaces, NT or TE termination		
Clock Source	5 ppm High Precision		
Network Interfaces			
Ethernet	4 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, 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		
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		
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users		
Transport Mediation	SIP over UDP/TCP/TLS, IPv4 / IPv6, RTP / SRTP (SDES)		
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)		
URI and Number Manipulations	URI user and host name manipulations, ingress and egress digit manipulation		
Vocoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB/WB, SILK-NB/WB, Opus-NB/WB		
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion		
NAT	Local and far-end NAT traversal for support of remote workers		
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		
Standalone Survivability	Maintains local calls in the event of WAN failure.		
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, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control		
Direct Media (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Voice Quality Monitoring	RTCP-XR, AudioCodes Session Experience Manager (SEM)		
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, third-party routing control through REST API		
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters		
Routing Features	Least-cost routing, call forking, load balancing, E911 gateway support, 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		
Regulatory Compliance			
Safety and EMC	IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 386		

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.

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/info
Website: www.audiocodes.com

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