AudioCodes Mediant 500 base Enterprise Session Border Controller and Media Gateway M500-ESBC

Product Name: AudioCodes Mediant 500 base Enterprise Session Border Controller and Media Gateway M500-ESBC
Manufacturer: AudioCodes
Model Number: M500-ESBC

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Please Note: Additions including sessions, remote implementation support and licenses/ software are available. If you require any of these additions, please enquire using the button above.

The AudioCodes Mediant 500 is a compact, high-performance VoIP connectivity solution for small enterprises and branch office locations. AudioCodes Mediant 500 Features

• 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, low CAPEX and reduced space and power footprints
• Enhanced security, robust perimeter defence 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

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.
AudioCodes Mediant 500 Technical Specification
Capacities

• Max. Signaling: 250
• Max. Registered Users: 1,500
• Max. RTP/SRTP Sessions: 200

Telephony Interfaces

• Digital: Single E1/T1 interface
• Analog: 5 ppm High Precision
• Clock Source: Various ISDN PRI protocols such as EuroISDN, North American NI-2, LucentTM 4/5ESS, NortelTM DMS- 100 and others. Different CAS protocols, including MFC R2,

Network Interfaces

• Ethernet: 4 GE interfaces configured in 1+1 redundancy or as individual ports

Security

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• 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, 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, 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
• High Availability SBC high availability with two-box redundancy, active calls preserved

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 prioritisation

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• SIPREC IETF standard SIP recording interface

Management

• OAM&P Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS

Physical and 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

Please Enquire