AudioCodes M500L MSBR DSL POTs (ADSL / VDSL) and EFM;4 port FE switch

Product Name: AudioCodes M500L MSBR DSL POTs (ADSL / VDSL) and EFM;4 port FE switch
Manufacturer: -
Model Number: M500L-G-A1GECS

AudioCodes M500L MSBR DSL POTs (ADSL / VDSL) and EFM;4 port FE switch (M500L-G-A1GECS)
The AudioCodes M500L-G-A1GECS is a compact, high performance VoIP connectivity solution for small enterprises and branch office locations. The M500L-G-A1GECS connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment as well as scaling up to 60 concurrent sessions.
AudioCodes M500L-G-A1GECS Key Features

ï¿½ 60 SBC Sessions
ï¿½ 8 TDM Sessions
ï¿½ Branch Survivability
ï¿½ Supports OPUS and SILK
ï¿½ Comprehensive interoperability
ï¿½ Hybrid functionality
ï¿½ Enhanced security
ï¿½ Superior voice quality
ï¿½ High resiliency

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.
AudioCodes M500L-G-A1GECS Technical Specification
Capacities

ï¿½ Max. Signaling: 60
ï¿½ Max. Registered Users: 200
ï¿½ Max. RTP/SRTP Sessions: 60

Telephony Interfaces

ï¿½ Digital: 1-4 BRI ports, network S/T interfaces, NT or TE termination
ï¿½ Analog: Up to 4 FXS and 4 FXO ports
ï¿½ Clock Source: 5 ppm High Precision

Network Interfaces

ï¿½ Ethernet: 4 GE interfaces configured in 1+1 redundancy or as individual ports

Security

ï¿½ Access Control: DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
ï¿½ VoIP Firewall: RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
ï¿½ Encryption/Authentication: TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
ï¿½ Privacy: Topology hiding, user privacy
ï¿½ Traffic Separation VLAN/physical interface separation for multiple media, control and OAMP

AudioCodes M500L MSBR DSL POTs (ADSL / VDSL) and EFM;4 port FE switch

interfaces

Interoperability

ï¿½ SIP B2BUA: Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
ï¿½ SIP Interworking: 3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer and more
ï¿½ Registration and Authentication: SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
ï¿½ Transport Mediation: Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP (SDES)
ï¿½ Header Manipulation: Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
ï¿½ Number Manipulations: Ingress and egress digit manipulation
ï¿½ SIP Interworking: 3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer
ï¿½ Signal Conversion: DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, packet-time conversion
ï¿½ NAT: Local and far-end NAT traversal for support of remote workers

Voice Quality and SLA

ï¿½ Call Admission Control: Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
ï¿½ Packet Marking: 802.1p/Q VLAN tagging, DiffServ, TOS
ï¿½ Standalone Survivability: Maintains local calls in the event of WAN failure
ï¿½ 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
ï¿½ Image Enhancement BLC/3D DNR
ï¿½ Direct Media: Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
ï¿½ Test Agent: Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs

SIP Routing

ï¿½ Routing Criteria: Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
ï¿½ Querying External Databases: Routing based on customized queries of ENUM, LDAP, HTTP server (REST API)
ï¿½ Route To: Configured SIP peers, registered users, IP address, request URI
ï¿½ Advanced Routing Features: Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
ï¿½ SIPREC: IETF standard SIP recording interface

Management

ï¿½ OAM&P: Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS

Physical/ Environmental

AudioCodes M500L MSBR DSL POTs (ADSL / VDSL) and EFM;4 port FE switch

ï¿½ Dimensions: 51 x 296 x 160 mm (2 x 11.65 x 6.3 in.) (HxWxD)
ï¿½ Weight: 670g
ï¿½ Mounting: Desktop
ï¿½ Power: Single universal AC power supply 100-240V, 50-60 Hz, 12V/3A or 12V/5A
ï¿½ 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

Please Enquire