AudioCodes Mediant 1000 Enterprise Session Border Controller and Media Gateway (M1KB)

Product Name: AudioCodes Mediant 1000 Enterprise Session Border Controller and Media Gateway (M1KB)
Manufacturer: AudioCodes
Model Number: M1KB

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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 M1KB offers a complete connectivity solution for small-to-medium sized enterprises.
AudioCodes M1KB Key Features

ï¿½ 150 SBC Sessions
ï¿½ 192 TDM Sessions
ï¿½ Modular
ï¿½ Extensive Vocoder Support
ï¿½ Comprehensive interoperability, proven interoperability with SIP trunks, SIP platforms and IP cloud services
ï¿½ Hybrid functionality, true hybrid SBC and gateway platform for gradual migration, low CAPEX and reduced space and power footprints
ï¿½ Enhanced security, robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft
ï¿½ Superior voice quality, advanced capabilities for optimising and monitoring voice service quality
ï¿½ High resiliency, high availability using 1+1 redundancy, local branch survivability and PSTN fallback

Scaling up to 150 concurrent sessions, the Mediant 1000 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.
In addition, the Mediant 1000 supports up to 192 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 M1KB Technical Specifications
Capacities

ï¿½ Max Signalling: 150
ï¿½ Max. RTP/STRP Sessions: 120
ï¿½ Max. Transcoding Sessions: 96
ï¿½ Max. Registered Users: 600

Telephony Interfaces:

ï¿½ Modularity and Capacity: 6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)
ï¿½ Digital Module: Up to 6 E1 or 8 T1/J1 spans provided on trunk modules. Each module supports 1, 2, or 4 E1/T1/J1 spans, with an option of PSTN fallback
ï¿½ Digital PSTN Protocols: 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, E&M immediate start, E&M delay dial/start and others.
ï¿½ BRI Module: Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN fallback. Providing S/T interfaces; NT or TE termination; 2W per port (power

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supplied)
ï¿½ Analog Module: Up to 24 FXS interfaces, provided on 4-port FXS modules, ground/loop start, Up to 24 FXO interfaces, provided on 4 port-FXO modules, ground/loop start
ï¿½ Media Processing Module: Up to 4 Media Processing modules (MPM), providing additional DSP resources

Network Interfaces

ï¿½ Ethernet: Up to 6 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, DTLS, 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 interfaces
ï¿½ Intrusion Detection System: Detection and prevention of VoIP attacks, theft of service and unauthorized access

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 Transcoding and Vocoders: Coder normalisation including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB, G.727, iLBC, QCELP, GSM EFR
ï¿½ Signal Conversion: DTMF/RFC 2833/SIP, T.38 fax, V.34, 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. Outbound calls can use PSTN fallback (including E911).
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

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generation, RTP redundancy, broken connection detection
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 customised 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 prioritisation

Management

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

Physical/Environmental

ï¿½ Dimensions: 1U x 444 x 355 mm (HxWxD)
ï¿½ Weight: Approx. 9.7lb (4.4kg)
ï¿½ Mounting: Desktop or 19” rack mount
ï¿½ Power: Dual power supply 100-240V, 50-60 Hz, 1.5A max
ï¿½ Environmental: Operational: 0 to 40° C (32 to 104°F); Storage: -20 to 70°C (-4 to 158°F), Relative Humidity: 10 to 85% non-condensing

OSN Server Platform (Optional)

ï¿½ Single Chassis Integration: Optional embedded, x86, Intel-based Open Solution Network platform for third-party applications

Price: £1,301.40