

Mediant™ 1000

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 1000 enterprise session border controller (E-SBC)** and media gateway offers a complete connectivity solution for small-to-medium sized enterprises.



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.

150 SBC Sessions | 192 TDM Sessions | Modular | Extensive Vocoder Support



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

Specifications

Capacities			
Max. Signaling	150	Max. RTP/SRTP Sessions	120
Max. Transcoding Sessions	96	Max. Registered Users	600
Telephony Interfaces			
Modularity and Capacity	6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)		
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		
Digital PSTN Protocols	Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. Different CAS protocols, including MFC R2, E&M immediate start, E&M delay dial/start and others.		
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 supplied)		
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		
Media Processing Module	Up to 4 Media Processing modules (MPM), providing additional DSP resources		
Network Interfaces			
Ethernet	Up to 6 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		
Transcoding and 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, G.727, iLBC, QCELP, GSM EFR		
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, V.34, 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, RTPC-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)		
Advanced Routing Features	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization		
Route To	Configured SIP peers, registered users, IP address, request URI		
Management			
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS		
OSN Server Platform (Optional)			
Single Chassis Integration	Optional embedded, x86, Intel-based Open Solution Network platform for third-party applications		
Physical/Environmental			
Dimensions	1U x 444 x 355 mm (HxWxD)	Weight	Approx. 9.7lb (4.4kg)
Mounting	Desktop or 19" mount	Power	Dual power supply 100-240V, 50-60 Hz, 1.5A max
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		



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