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 | Certified SBC for Teams Direct Routing supporting media optimization

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

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Mediant™ 1000

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.
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	Automatic 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.729A/B, GSM-FR, AMR-NB, G.722, 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, 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
SIP Call Handling	
Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
Querying External Databases	Destinations based on customized queries of ENUM, LDAP, HTTP server (REST API)
Advanced Features	Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization
Available Destinations	Configured SIP peers, registered users, IP address, request URI
SBC Media Types	Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC)
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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