Mediant™ 3000

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 3000 enterprise session border controller (E-SBC)** and media gateway is a complete connectivity solution for small-to-medium sized enterprises.

Scaling up to 1,008 concurrent sessions, the Mediant 3000 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 3000 supports up to 2,016 voice channels in a modular 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.

1,008 SBC Sessions | 2,016 TDM Sessions | High Availability | 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

High availability, local branch survivability and PSTN fallback

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Specifications

Capacities			
Max. Signaling	1,008	**Max. RTP/SRTP Sessions**	882
Max. Transcoding Sessions	Up to 3,000 or 5,000 (HW config dependent)	**Max. Transcoding Sessions**	1,008

Telephony Interfaces	
PSTN	1 OC-3 or STM-1 APS optical links, 1 to 3 T3 (DS3) electrical links, up to 64/84 E1/T1 links
Network Interfaces	
Ethernet	Dual Redundant 100/1000 Base-T Ethernet ports and additional two Dual Redundant 100 Base-T Ethernet ports for OEM and Control (Available on the E1/T1 configuration only)
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, DTLS, 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/WebSocket, IPv4/IPv6, RTP/SRTP (SDES/DTLS)
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, AMR-WB (G.722.2), G.727, iLBC, QCELP, GSM EFR, EVRC, MS-RTA NB/WB, SPEEX NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion, V.150.1
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
High Availability	SBC high availability, active calls preserved
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)
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 prioritization
SIPREC	IETF standard SIP recording interface
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS
Physical/Environmental	
Dimensions	(HxWxD) 88mm x 482.6mm x 296.8mm, 4-slot, 2U cPCI chassis
Weight	Approx. 35.27 lb (16 kg), fully loaded
Power	48 V DC Dual Feed, with up to 2 DC Power modules or 100–240 V AC redundant Dual Feed

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