Mediant™ 3100

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 3100 session border controller (SBC) and media gateway** is a complete connectivity solution for medium-to-large sized enterprises, contact centers and service providers.

Scaling up to 5,000 concurrent SBC sessions, the Mediant 3100 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 3100 supports up to up to 64 E1/T1 spans in a 2U 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.

5,000 SBC Sessions | 64 E1/T1 Spans | Extensive Vocoder Support | Certified for Microsoft Teams Direct Routing with local media optimization

Comprehensive interoperability
Proven interoperability with SIP trunks, unified communications solutions, PBXs and IP cloud services

Hybrid functionality
True hybrid SBC and gateway platform for gradual migration to IP communications, 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

Service resilience
Local branch survivability and PSTN fallback

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Specifications

Capacities			
Max. Signaling	5,000	**Max. RTP/SRTP Sessions**	5,000
Max. Registered Users	20,000	**Max. Transcoding Sessions**	3,200

Telephony Interfaces	
Digital PSTN Protocols	Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS- 100 and others.
PSTN	8 to 64 E1/T1 interfaces
Network Interfaces	
Ethernet	8 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, 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/WB, Opus-NB/WB, SILK-NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, DTLS, RTP multiplexing, secure RTCP with feedback.
NAT	Local and far-end NAT traversal for support of remote workers, ICE full and lite support (RFC 8445)
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	2U high 19-inch rack wide (H x W x D) 88 x 438 x 490 mm (3.5 x 17.24 x 19 inches)
Weight	11.5 kg (25.3 lbs.) for fully-populated chassis
Operating Temperature	Operational: 0° - 40° C (41° - 104° F) Storage: -25° - 70° C (-13° - 158° F) Humidity: 5% - 90% non-condensing
Power	Redundant Dual Feed, 100-240 V AC/9-4A, 50-60Hz or 48VDC 18A max

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