Mediant™ 9030/9080

Session Border Controller

AudioCodes **Mediant 9000 session border controllers (SBCs)** are high capacity solutions for service providers and enterprises, delivering service assurance, security and reliable connectivity between different VoIP networks. They connect IP-PBXs to any SIP trunking service provider and offer superior performance in connecting any SIP to SIP environment.

The available models are:

- Mediant 9030
- Mediant 9080

Mediant 9000 SBCs are a perfect solution for service providers and large organizations such as contact centers, large data centers, hosted services and government institutions where security, reliability and high performance are critical.

Up to 70,000 SBC Sessions | Pure IP SBC | 1+1 High Availability | OPUS and SILK Support

Comprehensive interoperability
Proven interoperability with SIP trunks, SIP platforms and IP cloud services

Flexible licensing
Various licensing options for easy and cost-effective scalability regardless of enterprise size

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 using 1+1 redundancy and local branch survivability

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Mediant™ 9030/9080

Specifications

Capacities		
	Mediant 9030	**Mediant 9080**
Max. Signaling Sessions	30,000	70,000
Max. Registered Users	200,000	500,000
Max. Transcoding	1,000	30,000 (Media transcoding cluster)
Max. Media Sessions	30,000	70,000
Max. RTP/SRTP Sessions	30,000/30,000	70,000/40,000

Network Interfaces	
Ethernet	12x1Gb or 8x1Gb and 4x10Gb Ethernet 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	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/WebSocket/SCTP, 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), SILK-NB/WB, Opus-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, lite ICE, DTLS, RTP multiplexing, secure RTCP with feedback
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.
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 with two-box redundancy, active calls preserved
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)
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)
SIPREC	IETF standard SIP recording interface, supporting both audio and video SBC sessions

Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC)
Multi Tenancy	Advanced multi-tenant SBC partitioning

Physical/Environmental			
Dimensions	42.9mm x 434.6mm x 707mm (HxWxD)	**Weight**	Between 13.04 kg (28.74 lb) and 16.27 kg (35.86 lb)
Mounting	19" mount	**Operating Temperature**	10° to 35°C
Power	Dual redundant 100-240V AC power supply/ Dual redundant -48 VDC power supply		

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