Mediant™ 4000

Session Border Controller

The AudioCodes **Mediant 4000 session border controller (SBC)** is a mid-to-high scale capacity solution for enterprises and service providers, delivering service assurance and enabling scalable, reliable and secured connectivity between different VoIP networks.

Scaling up to 5,000 concurrent sessions, the Mediant 4000 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

The Mediant 4000 is a perfect solution for enterprises and large organizations such as contact centers, large data centers, hosted service providers and government institutions where security, reliability and high performance are critical.

5,000 SBC Sessions | Pure IP SBC | 1+1 High Availability | Supports OPUS and SILK

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

Specifications

Capacities		
	Mediant 4000	**Mediant 4000B**
Max. Signaling	5,000	5,000
Max. RTP/SRTP Sessions	5,000/3,000	5,000
Max. Transcoding Sessions	2,400	5,000
Max. Registered Users	20,000	20,000

Network Interfaces	
Ethernet	8 100/1000 Base-T Ethernet ports for physical separation between multiple LAN and WAN between Media, Control and OA&M

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
URI and Number Manipulations	URI user and host name 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	Decisions based on customized queries of ENUM, LDAP, HTTP server (REST API)
Available Destinations	Configured SIP peers, registered users, IP address, request URI
Advanced Features	Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization
Multiple LANs	Support for up to 48 separate LANs
SIPREC	IETF standard SIP recording interface

Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS
Multi Tenancy	Advanced multi-tenant SBC partitioning

Physical/Environmental		
	Mediant 4000	**Mediant 4000B**
Dimensions	1U x 19” (444mm) x 14” (355mm) (HxWxD)	1U x 19” (444mm) x 14.9” (378mm) (HxWxD)
Weight	Approx. 11.7 lbs (5.3Kg)	Approx. 16.3 lbs (7.4Kg)
Mounting	Desktop or 19” rack mount	
Power	Dual power supply (hot- swappable): 100-240VAC 50-60Hz 2.5A max	Dual universal power supply (hot- swappable): 40-60VDC, 17A max., or 100-240 VAC, 50-60 Hz, 7A max
Environmental	5°-40° C	

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