Mediant™ 4000

Session Border Controller

Benefits
- Pure-IP SBC for medium-large enterprise deployments
- Offers comprehensive security, interoperability and reliability
- Delivers high service performance and voice quality
- Flexible licensing options for cost-effective scalability

Key Features
- Scalable to 5,000 SBC sessions
- Extensive SIP mediation capabilities
- Supports remote workers and mobile SIP clients
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Branch survivability during WAN failure
- Active/Standby High Availability
- Advanced media handling including transcoding and wideband speech

The AudioCodes **Mediant 4000 Session Border Controller (SBC)** is a mid-to-high scale capacity member of AudioCodes' field-proven hardware-based SBC product family, designed to offer enterprises and service providers a reliable and scalable SBC solution. The Mediant 4000 SBC supports wide-ranging SIP interoperability, delivering service assurance and enabling scalable, reliable and secured connectivity between different VoIP networks.

The Mediant 4000 SBC provides 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.

Extensive Mediation Capabilities and Proven Interoperability
The Mediant 4000 SBC includes comprehensive media security and SIP normalization capabilities. It offers full interoperability with an extensive list of IP-PBXs, unified communications solutions and SIP trunking provider networks.

Security
The Mediant 4000 SBC provides robust protection for the IP communications infrastructure, preventing fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability
The Mediant 4000 SBC offers active/standby high availability and maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities result in minimum communications downtime.

Applications
- SIP trunking
- Hosted PBX & UC as a Service
- IP contact centers
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems
- Residential VoIP

Mediant™ 4000

SPECIFICATIONS

Capacities

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

Network Interfaces

Ethernet	8 Redundant 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
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
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access

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
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
Transport Mediation	SIP over UDP/TCP/TLS/WebSocket, IPv4 / IPv6, RTP / SRTP (SDES/DTLS)
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
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.729, 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 Controller	Interworking between WebRTC devices 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	Based on bandwidth, session establishment rate, number of connections/registrations
Packet marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintains local calls in the event of WAN failure.
Impairment Mitigation	Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection
Voice Enhancement	Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control
Direct Media (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	RTCP-XR, AudioCodes Session Experience Manager (SEM)
High Availability (Redundancy)	SBC high availability with two-box redundancy, active calls preserved
Quality of Experience	Access control and media quality enhancements based on QoE and bandwidth utilization
Test agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs

SIP Routing

Routing Methods	Request URL, IP address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
Routing Features	Least-cost routing, call forking, load balancing, E911 gateway support, 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: 100-240VAC 50-60Hz 2.5A max	Dual universal power supply: 40-60VDC, 17A max., or 100-240 VAC, 50-60 Hz, 7A max.
Environmental	5°-40° C	

Regulatory Compliance

Safety and EMC	UL60950-1
	FCC Part 15 Class A
	ICES-003 Class A
	CE marking: IEC60950-1, EN55024, EN55032 Class A, EN61000-3-2, EN61000-3-3, ETSI EN300 386

ABOUT AUDIOCODES

AudioCodes Ltd. (NasdaqGS: AUDC) designs, develops and sells advanced Voice-over-IP (VoIP) and converged VoIP and Data networking products and applications to Service Providers and Enterprises. AudioCodes is a VoIP technology market leader, focused on converged VoIP and data communications, and its products are deployed globally in Broadband, Mobile, Enterprise networks and Cable. The Company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Routers, Session Border Controllers (SBC), Residential Gateways, IP Phones, Media Servers, Value Added Applications and Professional Services. AudioCodes' underlying technology, VoIPPerfectHD™, relies on AudioCodes' leadership in DSP, voice coding and voice processing technologies. AudioCodes' High Definition (HD) VoIP technologies and products provide enhanced intelligibility and a better end user communication experience in Voice communications.