AudioCodes Session Border Controller (SBC) Products

Mediant 500 Session Border Controller

The **AudioCodes Mediant 500 Enterprise Session Border Controller (E-SBC)** is a compact, high performance VoIP connectivity solution for small enterprises and branch office locations. The Mediant 500 connects IP-PBXs and unified communications platforms to any SIP trunking service provider, scaling up to 250 concurrent SBC sessions. It offers superior performance in connecting any SIP to SIP environment, legacy TDM-based PBX systems to IP networks and IP-PBXs to the PSTN, supporting up to 30 voice channels in a 1U platform. It also ensures secure and reliable communications for branch offices in distributed enterprise communications deployments.

Benefits

- A highly integrated device for secured SIP Trunking and PSTN access, forming a single and managed point of demarcation for VoIP networks
- Compact, high performance VoIP connectivity device for small enterprises and branch offices
- Extensive interoperability and partnerships that extend across multiple vendor devices and protocol implementations
- Offers comprehensive security and reliability
- Delivers high service performance and voice quality
- Branch office survivability in the event of a WAN outage

Key Features

- Rich and powerful SIP normalization and routing mechanisms for seamless interoperability
- Support for E1/T1 digital TDM interface
- Supports remote workers and mobile SIP clients
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- High Availability using two box redundancy

Vast mediation capabilities and proven interoperability

The Mediant 500 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 500 provides robust protection for the IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The Mediant 500 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

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Specifications

Capacities	
Max. Signaling/Media Sessions	250
Max. SRTP/RTP Sessions	180
Max. Registered Users	800
Telephony Interfaces	
Digital	Single E1/T1 interface
Clock Source	5 ppm High Precision
Networking Interfaces	
Ethernet	4 GE interfaces configured in 1+1 redundancy or as individual ports
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
Encryption and Authentication	TLS, 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 OAM interfaces
Intrusion Detection System	Detect and mitigate VoIP attacks, prevent Theft of Service and unauthorized access.
Interoperability	
SIP B2BUA	Full SIP transparency, mature & broadly deployed SIP stack
SIP interworking	3xx redirect, REFER, PRACK, Session Timer, Early media, Call hold, Delayed offer
Registration	Registration and authentication on behalf of an IP-PBX
Transport Mediation	SIP over UDP to SIP over TCP or SIP over TLS, IPv4 to IPv6, RTP to SRTP, V.34 Fax
Header Manipulation	Ability to add/modify/delete headers using advanced regular expressions
URI and Number Manipulations	URI User and Host name manipulations. Ingress & Egress Digit Manipulation
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	Maintain local calls in the event of WAN failure. Outbound calls use PSTN Fallback for external connectivity (including E911)
Transparent Media	Low latency, unprocessed payload transfer
Media De-anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)
Redundancy	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
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, Coder type etc)
Redundancy	Detect proxy failures and route to alternative proxies
Routing Features	Least cost routing, call forking, load balancing
SIPRec	IETF standard SIP recording interface
Physical / Environmental	
Dimensions	43.7 (1U) x 310 x 210 mm (HxWxD)
Weight	4.4 lb (2.0kg)
Mounting	Desktop or 19" rack mount
Power	Single universal AC power supply 100-240V, 0.8A, 50-60 Hz
Environmental	Operational: 5 to 40°C (41 to 104°F) Storage: -25 to 85°C (-13 to 185°F) Humidity: 10 to 90% non-condensing
Regulatory Compliance	
Safety and EMC	IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 386
Environmental Storage	ETS300019-2-1 class T1.2
Transportation	ETS300019-2-2 class T2.3
Operating	ETS300019-2-3

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 & 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 and Value Added Applications. AudioCodes' underlying technology, VoIPerfect HDTM, 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.

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Ref. # LTRM-30034 04/14 V.1

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