## AudioCodes Session Border Controller (SBC) Products

# Mediant 500

### **Session Border Controller**



### **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

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 a single E1/T1 interface with 30 voice channels in a 1U platform. It also ensures secure and reliable communications for

#### Vast mediation capabilities and proven interoperability

branch offices in distributed enterprise communications deployments.

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



# **Mediant 500**

#### **SPECIFICATIONS**

Max. Signaling/Media Sessions	250	Max. SRTP/RTP Sessions	180	
Max. Registered Users	800	Max. Stri / tri Gessions	100	
-	800			
Telephony Interfaces	Cingle E1 /T1 interfe	200		
Digital Course	Single E1/T1 interfa			
Clock Source  Digital PSTN Protocols		ISDN PRI protocols such as EuroISDN, North Also supports different variants of CAS protoco	American NI-2, Lucent™ 4/5ESS, Nortel™ DM ols, including MFC R2, E&M immediate start, E&	
Network Interfaces				
Ethernet	4 GE interfaces con	nfigured in 1+1 redundancy or as individual po	orts	
Security				
Access Control	DoS/DDoS line rate	e protection, bandwidth throttling, dynamic bla	acklisting	
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching			
Encryption/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 OAMP interfaces			
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access			
Interoperability				
SIP B2BUA	Full SIP transparent	cv mature and broadly deployed SIP stack sta	ateful proxy mode	
SIP interworking	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode  3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer			
-	3xx redirect, KEFER, PRACK, session timer, early media, call hold, delayed offer  User registration restriction control, registration and authentication on behalf of users, SIP authentication server			
Registration and Authentication	for SBC users			
Transport Mediation	SIP over UDP/TCP/TLS, IPv4 / IPv6, RTP / SRTP (SDES)			
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			
Vocoders	Coder normalization including coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723. G.726, G.729, GSM-FR, AMR-NB/WB, SILK-NB/WB, Opus-NB/WB			
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, V.34, packet-time conversion			
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. Outbound calls can use PSTN fallback for external connectivity (including E911)			
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			
		Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Direct Media	Hair-pinning of loca	I calls to avoid unnecessary media delays and	d bandwidth consumption	
		I calls to avoid unnecessary media delays and es Session Experience Manager (SEM)	d bandwidth consumption	
Direct Media (No Media Anchoring)	RTCP-XR, AudioCode			
Direct Media (No Media Anchoring) Voice Quality Monitoring	RTCP-XR, AudioCode SBC high availability	es Session Experience Manager (SEM)	ved	
Direct Media (No Media Anchoring) Voice Quality Monitoring High Availability (Redundancy)	RTCP-XR, AudioCode SBC high availability Access control and	es Session Experience Manager (SEM) y with two-box redundancy, active calls preser	rved and bandwidth utilization	
Direct Media (No Media Anchoring) Voice Quality Monitoring High Availability (Redundancy) Quality of Experience Test agent	RTCP-XR, AudioCode SBC high availability Access control and	es Session Experience Manager (SEM) y with two-box redundancy, active calls preser media quality enhancements based on QoE a	rved and bandwidth utilization	
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#### **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, VolPerfect 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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