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.

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
## Specifications

### Capacities

<table>
<thead>
<tr>
<th>Feature</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max. Signaling/Media Sessions</td>
<td>250</td>
</tr>
<tr>
<td>Max. SRTP/RTP Sessions</td>
<td>180</td>
</tr>
<tr>
<td>Max. Registered Users</td>
<td>800</td>
</tr>
</tbody>
</table>

### 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
- **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 Failback for external connectivity (including ED1)
- **Transparent Media**: Low latency, unprocessed payload transfer
- **Media De-anchoring**: Hairpinning 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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dimensions</td>
<td>43.7 (1U) x 310 x 210 mm (HxWxD)</td>
</tr>
<tr>
<td>Weight</td>
<td>4.4 lb (2.0kg)</td>
</tr>
<tr>
<td>Mounting</td>
<td>Desktop or 19&quot; rack mount</td>
</tr>
<tr>
<td>Power</td>
<td>Single universal AC power supply 100-240V, 0.8A, 50-60 Hz</td>
</tr>
<tr>
<td>Environmental</td>
<td>Operational: 5 to 40°C (41 to 104°F) Storage: -25 to 85°C (-13 to 185°F) Humidity: 10 to 90% non-condensing</td>
</tr>
</tbody>
</table>

### Regulatory Compliance

- **Safety and EMC**: IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 388
- **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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