Hybrid SBC and Media Gateway

The AudioCodes Mediant 800 enterprise session border controller (E-SBC) and media gateway offers a complete connectivity solution for small-to-medium sized enterprises.

Scaling up to 400 concurrent sessions, the Mediant 800 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 800 supports up to 124 voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN.

### 400 SBC Sessions | 124 TDM Sessions | 1+1 High Availability | Supports OPUS and SILK

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
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<tbody>
<tr>
<td>Comprehensive interoperability</td>
<td>Proven interoperability with SIP trunks, SIP platforms and IP cloud services</td>
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<tr>
<td>Hybrid functionality</td>
<td>True hybrid SBC and gateway platform for gradual migration, low CAPEX and reduced space and power footprints</td>
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<tr>
<td>Enhanced security</td>
<td>Robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft</td>
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<tr>
<td>Superior voice quality</td>
<td>Advanced capabilities for optimizing and monitoring voice service quality</td>
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<tr>
<td>High resiliency</td>
<td>High availability using 1+1 redundancy, local branch survivability and PSTN fallback</td>
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![AudioCodes Session Border Controllers](image-url)
## Specifications

### Capacities

<table>
<thead>
<tr>
<th></th>
<th>Max. Signaling</th>
<th>Max. RTP/SRTP Sessions</th>
<th>Max. Transcoding Sessions</th>
<th>Max. Registered Users</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mediant 800B</td>
<td>250</td>
<td>250/250</td>
<td>57</td>
<td>1500</td>
</tr>
<tr>
<td>Mediant 800C</td>
<td>400</td>
<td>400/300</td>
<td>114</td>
<td>2000</td>
</tr>
</tbody>
</table>

### Telephony Interfaces

- **Analog**: 4/8/12 FXS ports; 4/8/16 FXO ports
- **Digital**: Up to 4 E1/T1 interfaces; 4/8 BRI Ports
- **Clock Source**: 5 ppm High Precision
- **Digital PSTN Protocols**: Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. Different CAS protocols, including MFC R2, E&M immediate start, E&M delay start and others.

### Network Interfaces

- **Ethernet**: 4 GE or 4 GE + 8 FE interfaces configured in 1+1 redundancy or as individual ports

### 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 times, early media, call hold, delayed offer and more
- **Registration and Authorization**: 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
- **Number Manipulations**: Ingress and egress digit manipulation
- **Transcoding and Vcoders**: 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/SP T.38 fax, T.38 V3, V.34, packet-time conversion, V150.1
- **WebRTC Gateway**: Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video codec, lite ICE, DTLS, RTP multiplexing,
- **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. Outbound calls can use PSTN fallback (including E911)
- **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, QoS, bandwidth
- **Querying External Databases**: Destinations 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 parking, E911 emergency call detection and prioritization
- **SIPREC**: IETF standard SIP recording interface

### Management

- **OAM&P**: Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS

### Physical/Environmental

<table>
<thead>
<tr>
<th>Dimensions</th>
<th>Weight</th>
<th>Operating Temperature</th>
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<tr>
<td>1U x 320mm x 345mm</td>
<td>Approx. 5.95lb (2.7kg) loaded with OSN</td>
<td>5°-40°C</td>
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### Power

- **Internal AC power supply rated**: 100-240 VAC ~ 50-60Hz 1.5A maximum
- **Optional**: Additional 12V XIA DC power, via an AudioCodes external AC/DC power adaptor

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