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 | Certified SBC for Teams Direct Routing | Supports OPUS and SILK**

- **Comprehensive interoperability**
  Proven interoperability with SIP trunks, SIP platforms and IP cloud services

- **Hybrid functionality**
  True hybrid SBC and gateway platform for gradual migration, low CAPEX and reduced space and power footprints

- **Enhanced security**
  Robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft

- **Superior voice quality**
  Advanced capabilities for optimizing and monitoring voice service quality

- **High resiliency**
  High availability using 1+1 redundancy, local branch survivability and PSTN fallback
Specifications

AudioCodes Session Border Controllers

Mediant™ 800

**Capabilities**

<table>
<thead>
<tr>
<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>
</tr>
<tr>
<td>Mediant 800C</td>
<td>400</td>
<td>400/250</td>
<td>114</td>
</tr>
</tbody>
</table>

**Telephony Interfaces**

| Analog          | 4 FXS/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™ DM5-100 and others. Different CAS protocols, including MFC R2, E&M immediate start, E&M delay dial/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

Automatic 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

SIP redirect, REFER, PRACK, session times; early media, call hold, delayed offer and more

Registration and Authentication

SIP Registrar, registration on behalf of users/clients, SIP Digest access authentication

Transport Mediation

Media between SIP and RTP over UDP/TCP/TLS/WebSocket, IPv4/IPv6, RTP/SRTP, SDES/DTLS

Header Manipulation

Add/delete SIP headers and message body using simple WinS application with powerful capabilities such as variables and utility functions

Number Manipulations

Ingress and egress digit manipulation

Transcoding and Voicodes

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, LBC

Signal Conversion

DTMFRFC 2833/SP T38 fax, T38 V.34, packet-time conversion

WebRTC Gateway

Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video codec, DTLS, RTP multiplexing.

NAT

Local and far-end NAT traversal for support of remote workers, ICE full and lite support (RFC 8445)

**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 fall-back (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 retransmission, 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 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/REST API

Available Destinations

Configured SIP peers, registered users, IP address, request URI

Advanced Features

Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization

SBC Media Types

Audio/Voice/Video/Fax/Text/Message Session Relay Protocol (MSRP), Binary Floor Control Protocol (BFCP)

SIPREC

HoST standard SIP recording interface, supporting both audio and video SBC sessions

**Management**

OAM&P

Browser-based GUI, CLI, SNMP, INI Configuration file, REST API

One Voice Operations Center (OVOC)

**Physical/Environmental**

<table>
<thead>
<tr>
<th>Dimensions</th>
<th>Weight</th>
</tr>
</thead>
<tbody>
<tr>
<td>1U x 345mm x 320mm (HxWxD)</td>
<td>Approx. 5.95lb (2.7kg) loaded with OSN</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Mounting</th>
<th>Operating Temperature</th>
</tr>
</thead>
<tbody>
<tr>
<td>Desktop or 19&quot; rack mount</td>
<td>5°-40° C</td>
</tr>
</tbody>
</table>

| Power | Internal AC power supply rated: 100–240 VAC ~ 50– 60Hz 1.5A maximum |
|-------| (Optional) Additional 12V 10A DC power, via an AudioCodes external AC/DC power adapter |

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