Hybrid SBC and Media Gateway
The AudioCodes Mediant 1000 enterprise session border controller (E-SBC) and media gateway offers a complete connectivity solution for small-to-medium sized enterprises.

Scaling up to 150 concurrent sessions, the Mediant 1000 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

In addition, the Mediant 1000 supports up to 192 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.

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
Local branch survivability and PSTN fallback with E911
### AudioCodes Session Border Controllers

#### Mediant™ 1000

**Specifications**

**Capacities**

<table>
<thead>
<tr>
<th></th>
<th>Max. Signaling</th>
<th>150</th>
<th>Max. RTP/SRTP Sessions</th>
<th>120</th>
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<tbody>
<tr>
<td>Max. Transcoding Sessions</td>
<td>96</td>
<td>Max. Registered Users</td>
<td>600</td>
<td></td>
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</tbody>
</table>

**Telephony Interfaces**

- **Modularity and Capacity**: 6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)
- **Digital Module**: Up to 6 E1 or 8 T1/UT1 spans provided on trunk modules. Each module supports 1, 2, or 4 E1/T1/UT1 spans, with an option of PSTN fallback
- **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.
- **BRI Module**: Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN fallback. Providing 5/1 interfaces; NT or TE termination, 2W per port (power supplied)
- **Analog Module**: Up to 24 FXS interfaces, provided on 4-port FXS modules, ground/loop start
- **Media Processing Module**: Up to 4 Media Processing modules (MPM), providing additional DSP resources

**Network Interfaces**

- **Ethernet**: Up to 6 GE interfaces configured in 1+1 redundancy or as individual ports
- **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, 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**: 3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer and more
- **Registration and Authentication**: SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
- **Transport Mediation**: Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/RTSP (SDES)
- **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 Vocoder**: 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, G.722, G.727, iLBC, QCELP, GSM EFR
- **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**: 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
- **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)
- **Advanced Features**: Alternative destinations, load balancing, LCR, call rerouting, E911 emergency call detection and prioritization
- **Available Destinations**: Configured SIP peers, registered users, IP address, request URI
- **SBC Media Types**: Audio/Video/Fax/Text/Message Session Relay Protocol (MSRP)/Binary Floor Control Protocol (BFCP)

**Management**

- **OAM&P**: Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC)
- **OSN Server Platform (Optional)**: Optional embedded, x86, Intel-based Open Solution Network platform for third-party applications

**Single Chassis Integration**

- **Optional embedded, x86, Intel-based Open Solution Network platform for third-party applications**

**Physical/Environmental**

- **Dimensions**: 4U x 444 x 355 mm (HxWxD)
- **Mounting**: Desktop or 19” mount
- **Environmental**: Operational: 0 to 43° C (32 to 104°F), Storage: -20 to 70°C (-4 to 158°F)

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