

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



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

| Capacities | | | | |
|----------------------------------|--|------------------------|--|-----------------------|
| | Max. Signaling | Max. RTP/SRTP Sessions | Max. Transcoding Sessions | Max. Registered Users |
| Mediant 800B | 250 | 250/250 | 57 | 1500 |
| Mediant 800C | 400 | 400/300 | 114 | 2000 |
| Telephony Interfaces | | | | |
| Analog | 4/8/12 FXS ports; 4/8/12 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 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 | | | |
| Intrusion Detection System | Detection and prevention of VoIP attacks, theft of service and unauthorized access | | | |
| 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/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 Vocoders | 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, iLBC | | | |
| Signal Conversion | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V34, packet-time conversion, V.150.1 | | | |
| WebRTC Gateway | Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, 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, RTPC-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, QoE, 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 forking, E911 emergency call detection and prioritization | | | |
| SBC Media Types | Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP) | | | |
| SIPREC | IETF 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 | | | | |
| Dimensions | 1U x 345mm x 320mm (HxWxD) | Weight | Approx. 5.95lb (2.7kg) loaded with OSN | |
| Mounting | Desktop or 19" rack mount | Operating Temperature | 5°-40° C | |
| 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 adaptor | | | |



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