# Mediant™ 1000 Hybrid E-SBC and Media Gateway

## Benefits

- Fully integrated device for secured SIP trunking and PSTN access
- Hybrid SBC and Media Gateway platform lowers CAPEX and reduces space and power footprints
- Scalable "pay-as-you-grow" modular architecture
- Extensive interoperability and partnerships that extend across multiple vendor devices and protocol implementations
- Offers comprehensive security, interoperability and reliability
- Delivers high service performance and voice quality
- Branch office survivability in the event of a WAN Outage

## Key Features

- Rich and powerful SIP normalization and routing mechanisms for seamless interoperability
- Hybrid SBC enables seamless migration and PSTN fallback
- Modular support for analog and digital TDM interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- Optional Open Solution Network (OSN) server module for hosting value-added applications

The **AudioCodes Mediant 1000 Enterprise Session Border Controller (E-SBC)** and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 1000 connects IP-PBXs to any SIP trunking service provider, scaling up to 150 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 192 voice channels in a modular 1U platform.

### Vast mediation capabilities and proven interoperability

The Mediant 1000 supports a wide range of voice coders and is capable of transcoding between narrowband and wideband voice coders, providing SIP normalization, fax handling, gain control and numerous additional media processing features. It offers certified interoperability with leading unified communications solutions and SIP trunking providers.

### Security

The Mediant 1000 provides robust protection for IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

### Reliability

The Mediant 1000 maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities (including PSTN fallback with E911) 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

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# Mediant™ 1000

## Specifications

| Capacities | | | |
|---|---|---|---|
| Max. Signaling/Media Sessions | 150 | Max. SRTP/RTP Sessions | 120 |
| Max. Transcoding Sessions | 96 | Max. Registered Users | 600 |
| **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/J1 spans provided on trunk modules. Each module supports 1, 2, or 4 E1/T1/J1 spans, with an option of PSTN Fallback | | |
| Digital PSTN Protocols | Supporting various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. It also supports different variants of CAS protocols, including MFC R2, E&M immediate start, E&M delay dial / start and others. | | |
| BRI Module | Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN Fallback. Providing S/T interfaces; NT or TE termination; 2W per port (power supplied) | | |
| Analog Module | Up to 24 FXS/FXO interfaces, provided on 4 ports FXO / 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 | | |
| **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, advanced RTP latching | | |
| Encryption/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 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 | | |
| Registration and Authentication | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users | | |
| Transport Mediation | SIP over UDP/TCP/TLS, IPv4 / IPv6, RTP / SRTP (SDES) | | |
| Message Manipulation | Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex) | | |
| URI and Number Manipulations | URI user and host name 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.729, GSM-FR, AMR-NB/WB, SILK-NB/WB, 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 | Based on bandwidth, session establishment rate, number of connections/registrations | | |
| 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 for external connectivity (including E911) | | |
| Impairment Mitigation | Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection | | |
| Voice Enhancement | Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control | | |
| Direct Media (No Media Anchoring) | Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption | | |
| Voice Quality Monitoring | RTCP-XR, AudioCodes Session Experience Manager (SEM) | | |
| 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, third-party routing control through REST API | | |
| Advanced Routing Criteria | QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters | | |
| Routing Features | Least-cost routing, call forking, load balancing, E911 gateway support, 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 | | |
| **OSN Server Platform (Optional)** | | | |
| Single Chassis Integration | Embedded, Open Network Solution Platform for third-party services | | |
| Memory | Up to 8GB RAM | | |
| Storage | HDD or SSD | | |
| **Physical / Environmental** | | | |
| Dimensions | 1U x 320mm x 345mm (HxWxD) | Weight | Approx. 9.7lb (4.4kg) |
| Mounting | Desktop or 19” mount | Power | Single power supply 100-240V, 50-60 Hz, 1.5A max. optional redundant power supply |
| Environmental | Operational: 0 to 40° C (32 to 104°F); Storage: -20 to 70°C (-4 to 158°F)<br>Relative Humidity: 10 to 85% non-condensing | | |
| **Regulatory Compliance** | | | |
| Telecommunication Standards | TIA/EIA-IS-968, TBR-4, TBR-13, and TBR-21 | | |
| Safety and EMC Standards | UL60950-1; FCC 47 CFR part 15 Class B<br>CE Mark (EN55022 Class B, EN60950-1, EN55024, EN300 386, EN61000-3-2/3-3) | | |
| Environmental Specifications | ETS 300019-2-1 Storage T1.2, ETS 300019-2-2 Transportation T2.3<br>ETS 300019-2-3 Operating T3.2 | | |

## 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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