The AudioCodes Mediant 800 Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 800 connects IP-PBXs to any SIP trunking service provider, scaling up to 250 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 60 voice channels in a 1U platform.

Vast mediation capabilities and proven interoperability
The Mediant 800 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 800 provides robust protection for the IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability
The Mediant 800 offers active/standby high availability and 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
### Specifications

**Capabilities**
- Max. Signaling/Media Sessions: 250
- Max. SRTP/RTP Sessions: 180
- Max. Transcoding Sessions: 45
- Max. Registered Users: 800

**Telephony Interfaces**
- Analog: 4/8/12 FXS ports; 4/8/12 FXO ports
- Digital: 1/2 span E1/T1; 4/8 BRI ports, network S/T interfaces, NT or TE termination
- Clock Source: 5 ppm High Precision

**Networking 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
- VoIP Firewall: RTP pinhole management, Rogue RTP detection and prevention, SIP message policy
- Encryption and Authentication: TLS, SRTP, HTTPS, SDH, Client/Server SIP Digest authentication, RADIUS Digest
- Privacy: Topology Hiding, User Privacy
- Traffic Separation: VLAN/physical interface separation for multiple Media, Control and QAM interfaces
- Intrusion Detection System: Detect and mitigate VoIP attacks, prevent Theft of Service and unauthorized access.

**Interoperability**
- SIP B2BUA: Full SIP transparency, mature & broadly deployed SIP stack
- SIP interworking: 3xx redirect, REFER, PRACK, Session Timer, Early media, Call hold, Delayed offer
- Registration: Registration and authentication on behalf of an IP-PBX
- Transport Mediation: SIP over UDP over TCP over SIP over TLS, IPv4 to IPv6, RTP to SRTP; V.34 Fax
- Header Manipulation: Ability to add/modify/delete headers using advanced regular expressions
- URI and Number Manipulations: URI User and Host name manipulations, Ingress & Egress Digit Manipulation
- Signal Conversion: DTMF/RFC 2833, Inband/T.38 Fax, Packet time Conversion, V.150.1
- NAT: Local and Far End NAT traversal for support of remote workers
- Hybrid PSTN mode: Connect to TDM PBXs or PRI/CAS trunks for least-cost routing or fallback. Also useful for gradual enterprise migration to SIP, Support for analog, BRI and T1/E1/J1

**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
- Intelligent Voice: Multiple queues for granular prioritization of VoIP over other non-real time traffic types, Integrated Queueing and scheduling schemes (Strict Priority, Class based Prioritization queueing, fairness)
- Standalone Survivability: Maintain local calls in the event of WAN failure. Outbound calls use PSTN Failback for external connectivity (including E911)

**Transcoding and Vocoders**

**SIP Routing**
- Routing Methods: Request URL, IP Address, FQDN, ENUM, advanced LDAP
- Advanced Routing Criteria: QoE, bandwidth, SIP message (SIP request, Deter type etc.)
- Redundancy: Detect proxy failures and routing to alternative proxies
- Routing Features: Least cost routing, call forking, load balancing
- Multiple LANs: Support for up to 12 separate LANs
- SIP uc: EIT standard SIP recording interface

**OSN Server Platform (Optional)**
- Single Chassis Integration: Embedded, open Network Solution Platform for third-party services
- Memory: Up to 16 GB RAM
- Storage: HDD or SSD
- Physical / Environmental
  - Dimensions: 1U x 320mm x 345mm (HxWxD)
  - Weight: Approx. 5.95lb (2.7kg) loaded with OSN
  - Mounting: Desktop or 19" rack mount
  - Operating Temperature: 5°-40° C
  - Regulatory Compliance: TBR13/13 (E1 lines), TBR-3 (BRI interface)
  - Safety and EMC: IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 388
  - Environmental Storage: ETS3000019-2-1 class T2.3
  - Transportation: ETS3000019-2-2 class T2.3

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**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, VoPerfect 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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