The AudioCodes Mediant 3000 Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The Mediant 3000 connects IP-PBXs to any SIP trunking service provider, scaling up to 1008 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 2016 voice channels in a modular 1U platform.

Vast mediation capabilities and proven interoperability
The Mediant 3000 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 3000 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 3000 E-SBC supports high-availability configurations with reliable, “1+1” redundancy of all system components, ensuring no loss of active sessions during failure time. The Mediant 3000 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

Capacities

Max. Signaling/Media Sessions 1008
Max. Transcoding Sessions 1008
Max. SRTP/RTP Sessions 882
Max. Registered Users Up to 3000 or 5000 (HW config dependent)

Telephony Interfaces

PSTN 1 OC-3 or STM-1 APS optical links, 1 to 3 T3 (DS3) electrical links, up to 63/84 E1/T1 links

Network Interfaces

Ethernet Dual Redundant 100/1000 Base-T Ethernet ports and additional two Dual Redundant 100 Base-T Ethernet ports for OEM and Control (Available on the E1/T1 configuration only)

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/Authentication TLS, SRTP, HTTPS, SSH, IPsec, IKE, SNMPv3, Client/Server SIP Digest authentication, RADIUS Digest
Privacy Topology Hiding, User Privacy
Traffic Separation Physical separation (on E1/T1 configuration only) or VLAN interface separation for multiple Media, Control and OAM interfaces

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 to SIP over TCP or 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

Voice Quality and SLA

Call Admission Control Based on bandwidth, session establishment rate, number of connections/registrations
Standalone Survivability Maintain local calls in the event of WAN failure
Transparent Media Low latency, unprocessed payload transfer
Impairment Mitigation Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/ Comfort Noise Generation, RTP redundancy, broken connection detection
Media De-anchoring Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring AudioCodes Session Experience Manager (SEM)
Test agent Remote verification of connectivity, voice quality and SIP message flow between SIP UAs
Gain control Fixed & dynamic voice gain control
Redundancy Carrier grade “Five 9s” availability with 1+1 hardware redundancy, Active calls preserved

SIP Routing

Routing Methods Request URL, IP Address, FQDN, ENUM, advanced LDAP
Advanced Routing Criteria QoE, bandwidth, SIP message (SIP request, Coder type etc)
Routing Features Least cost routing, call forking, load balancing
SIPRec IETF standard SIP recording interface

Hardware Specifications

Dimensions (HxWxD) 88mm x 482.6mm x 296.8mm, 4-slot, 2U cPCI chassis
Weight Approx. 35.27 lb (16 kg), fully loaded
Power 48 V DC Dual Feed, with up to 2 DC Power modules or 100–240 V AC redundant Dual Feed

Regulatory Compliance

Telecommunication Standards FCC part 68, TBR4 and TBR13
Safety and EMC Standards UL 60950 FCC part 15 Class A CE Mark (EN5022 Class A, EN60950, EN 55024, EN300 386)
Environmental Specifications NEBS level 3: GR-63-Core, GR-1089-Core, Type 1 & 3, ETS300 019

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, VolPerfect 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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