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, VoIPerfectHD™, 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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AudioCodes

Mediant™ 800

Enterprise Session Border Controller (E-SBC)

- A highly integrated device for VoIP, Security & Access, forming a single managed point of demarcation
- Based on three core foundations: Perimeter Defense, Mediation and Service Assurance
- Standards-based solution with proven intero perability
- Software license scalability up to 60 secured SBC VoIP sessions
- Encryption for communication privacy and prevention of eavesdropping
- Transparent communication for mobile users
- Survivability with PSTN Failover
- IP-to-IP protocol normalization and media transcoding
- Proven Voice Quality superiority
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- Extensive filtering and admittance policies

SIPconnect COMPLIANT

AudioCodes

AudioCodes' Mediant 800™ Enterprise Session Border Controller (E-SBC) is a member of AudioCodes family of Enterprise Session Border Controllers, enabling connectivity and security between Small Medium Businesses (SMB) and Service Providers' VoIP networks.

The Mediant 800 E-SBC provides Perimeter Defense as a way of protecting companies from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any Service Provider; and Service Assurance for service quality and manageability. Designed as a cost-effective appliance, the Mediant 800 E-SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multi-service appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP mediation, PSTN access survivability, and third party value-added services applications. This enables enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

Why Enterprise Session Border Controllers?

Session Border Controllers were traditionally deployed at the border of service provider core networks. Both Enterprises and Service Providers have now realized the essential need of enterprise-based session border controllers, located at the customer premises for addressing the security, mediation and SLA requirements of the Enterprise. The Mediant 800 E-SBC provides an open and flexible architecture for all Enterprise deployments, acting as the demarcation point between an Enterprise and a SIP Trunking provider, an enterprise and a hosted Unified Communication service provider or an enterprise and other organizations for direct VoIP calling. Why Enterprise Session Border Controllers? Session Border Controllers were traditionally deployed at the border of service provider core networks. Both Enterprises and Service Providers have now realized the essential need of enterprise-based session border controllers, located at the customer premises for addressing the security, mediation and SLA requirements of the Enterprise. The Mediant 800 E-SBC provides an open and flexible architecture for all Enterprise deployments, acting as the demarcation point between an Enterprise and a SIP Trunking provider, an enterprise and a hosted Unified Communication service provider or an enterprise and other organizations for direct VoIP calling.

Enterprise Session Border Controller and Security Services

AudioCodes' Mediant 800 E-SBC is designed as a secured VoIP platform. Enhanced Media Gateway security features include encryption schemes, such as SRTP for media, TLS for SIP control, IPSec for management, and Denial of service protection. A fully featured Enterprise-class Session Border Controller provides a secured voice network deployment, based on the embedded Back-to-Back User Agent (B2BUA).

Integrated PSTN connectivity

AudioCodes' Mediant 800 E-SBC is built upon a highly interoperable VoIP Media Gateway, which can be delivered in several pre-defined configurations: a single E1/T1/J1 trunk and up to 4 BRI ports (8 calls) or 12 analog (FXS/FXO) ports. In addition, the Mediant 800 E-SBC provides enhanced dialing plans and voice routing capabilities along with its SIP to SIP mediation, allowing business customers to enjoy the benefits of SIP Trunking services, IP Centrex connectivity, Unified Communications, as well as flexible PSTN and legacy PBX connectivity to VoIP.

Mediant™ 800 Enterprise Session Border Controller (E-SBC)

Vast mediation capabilities and proven interoperability

In a world of growing choices of voice coders and SIP flavors, enterprises and services providers alike must ensure interoperability for successful integration and service delivery. The Mediant 800 E-SBC, with its extensive media processing capabilities, supports a wide range of voice coders as well as SIP normalization, fax handling, gain control and numerous additional media processing features.

As a direct evolution of the field-proven and highly interoperable Mediant family of VoIP media gateways, the Mediant 800 E-SBC provides unparalleled interoperability, enabling mediation between an extensive list of IP and TDM-based PBXs and SIP Trunking providers.

Quality of Service (QoS) and Quality of Experience (QoE)

AudioCodes Mediant 800 E-SBC supports enhanced IP Quality of Service (QoS) enforcement and Quality of Experience (QoE) Monitoring. Leveraging AudioCodes SEM, a QoE monitoring and management system, the Mediant 800 E-SBC enables service providers and multi-site enterprises to assess networks, certify VoIP deployments, and measure, monitor, track, and help optimize the QoE of their VoIP services. AudioCodes' Mediant 800 E-SBC also supports enhanced IP Quality of Service (QoS), including Ethernet frame tagging (802.1p), IP packet marking (Diffserv), and traffic shaping.

Survivability Services

Customers served by a centralized, SIP-based IP Centrex server (or branch offices in distributed enterprises) may face service discontinuities in case of WAN failure. The integrated SAS (Stand- Alone Survivability) feature of the Mediant 800 E-SBC enables internal office communication between SIP clients (e.g., IP Phones), along with PSTN fallback, in case of disconnection from the centralized IP Centrex server or IP-PBX.

Value-Added Services by an Open Platform for 3rd-Party Applications

AudioCodes' Mediant 800 E-SBC extends the flexibility of a standalone SBC with the built-in Open Solution Network (OSN) server option (based on an Intel processor). Independent Software Vendors and OEM customers can utilize this integrated, general purpose server to host their own applications (e.g., IP-PBX, IVR, Call Center, Conferencing, and more).

Applications

Sip Trunking Solution

Using the Mediant 800 E-SBC, SMB customers can seamlessly migrate from legacy PSTN connectivity to cost-effective SIP Trunking Services. The Mediant 800-ESBC provides security, session mediation and service level assurance services, connecting the SMB to multiple SIP Trunking providers, while maintaining interoperability and manageability.

Hosted Centrex Solution

IP Centrex solutions rely on VoIP technology, whose implementation may present significant challenges, especially to businesses without prior VoIP experience. One of the challenges is service continuity during WAN outages. The Mediant 800 E-SBC, with its Stand-Alone Survivability feature, is able to monitor registrations to the SIP Proxy, so that if connectivity is lost the Mediant 800 E-SBC can continue to serve in both internal and external calling capacities.

Mediant 800 E-SBC in SMB Networks

Small and medium Enterprises are motivated to become more productive, efficient, and responsive to their internal users. The convergence of secured voice services, Stand-Alone Survivability, Data Routing and Security, WAN Access, Mediation Services and Service Level Assurance, ensures a high level of investment protection, cost-optimization and support for the growing communication needs of the Enterprise.

Benefits for Service Providers

- A highly integrated device for VoIP, Data, Security & Access, forming a single managed point of demarcation
- Extensive interoperability and partnerships that extend across multiple vendor devices and protocol implementations
- Enhanced SIP Mediation capabilities, which enable SIP Trunking in a variety of TDM-PBX and IP-PBX customer environments
- Simplified management & maintenance using a unified management tool
- Assuring standalone survivability at the customer premises during WAN outage
- Quality of Experience (QoE) management solution

Benefits for Business Customers

- A highly integrated device for secured SIP Trunking and PSTN access, forming a single and managed point of demarcation for VoIP networks
- An integrated VoIP Media Gateway and E-SBC, reducing CAPEX and OPEX, eliminating the need to purchase and deploy different devices and simplifying maintenance and management
- Future-proof solution with the ability to support various SIP Trunking and UC applications
- Multiple service provider connectivity to optimize tariff rates
- Local survivability and PSTN Failover upon WAN network connectivity failures

Specifications

Capacities	
Max. Sessions	Up to 60
Max. Registered Users	Up to 200
Interfaces	
Analog	• 4/8/12 FXS ports, using RJ-11 connectors • 4/8/12 FXO ports, using RJ-11 connectors • 2/4/6 E&M ports, using R48S connectors
Digital	• 4/8 BRI ports, network S/T interfaces, NT or TE termination, using RJ-49c connectors • Dual span E1/T1, using RJ-48c connectors
Networking Interfaces	
Ethernet	1 or 6 pairs of GE/FE interfaces for 1+1 port redundancy
OSN Server Platform (Optional)	
Single Chassis Integration	Embedded, open Network Solution Platform for third-party services
CPU	Intel ATOM N270, 1.6 GHz up to 2MB RAM; or Intel Celeron 847E, 2 x 1.1 GHz, 2MB RAM
Memory	Up to 4GB RAM
Storage	SATA storage
Security	
Access Control	White/Black Lists, Denial of Service protection
VoIP Firewall and deep packet inspection	RTP pinhole management according to SIP offer/answer model. Rouge RTP detection and prevention*, SIP message policy
Encryption and Authentication	TLS, SRTP, HTTPS, SSH, IPSec, IKE, SNMPv3, MD5, Client/Server authentication
Privacy	Topology Hiding, User Privacy
Traffic Separation	Physical or VLAN Interface Separation (Media/OAM/Control)
Interoperability	
SIP	Standalone SIP B2BUA, Netann (RFC4240), MSCML (RFC5022) or RFC 4117 transcoding device control. Full SIP transparency, mature & broadly deployed SIP stack
ITSP and PBX support	Interoperable with many SIP trunk Service Providers and PBX vendors
Transport Mediation	SIP over UDP to SIP over TCP or SIP over TLS, IPv4 to IPv6 ,v.34 fax, RTP to SRTP
Header Manipulation	Programmable 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
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 and T1/E1/J1
Transcoding and Vocoders	Coder normalization, including: transcoding, coder enforcement and re-prioritization. Extensive vocoder support: Wireline: SILK, G.711a/mu, G.723.1, G.729A/B/E, Wideband: G.722, AMR-WB, SILK WB and AMR, G.726
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
Signal Detection	DTMF/RFC2833, Packet-time conversion
Voice Quality and SLA	
Call Admission Control	Deny excessive calls based on session establishment rate, number of connections and number of registrations (per SIP trunk or routing domain)
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 Queuing and scheduling schemes (Strict Priority, Class based Prioritization queuing, fairness)
Stand Alone Survivability	Maintain local calls in the event of WAN failure. Outbound calls 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
Media Anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)
SIP Routing	
Routing Methods	Request URL, Source/ Destination IP Address, Fully Qualified Doman Name, ENUM, LDAP
Alternative Routing and load balancing	Detect proxy failures and route to alternative proxies. Load balance across a pool of proxies, least cost routing
Multiple LANs	Support for up to 12 separate LANs
Physical / Environmental	
Enclosure	1U chassis
Dimensions	1U x 320mm x 345mm (HxWxD)
Weight	Approx. 5.95lb (2.7kg) loaded with OSN
Mounting	Desktop or 19" mount
Power	90-254 V AC power feeds
Operating Temperature	5°-40° C
Regulatory Compliance	
Telecommunications	TIA/EIA-IS-968, ETSI ES 203 021 (FXO interface)
Safety and EMC	• UL60950-1 • CB certification including national deviations • EN60950-1, EN55024, EN55022 Class A, EN61000-3-2, EN61000-3-3, EN300 386 • FCC 47 Part 15 Class A

*Roadmap