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.

International Headquarters
1 Hayarden Street,
Airport City
Lod 70151, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
27 World's Fair Drive,
Somerset, NJ 08873
Tel:+1-732-469-0880
Fax:+1-732-496-2298

Contact us: www.audiocodes.com/info
Website: www.audiocodes.com

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AudioCodes

Mediant™ 1000 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 interoperability
- Software license scalability up to 150 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

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AudioCodes

AudioCodes' Mediant 1000™ Enterprise Session Border Controller (E-SBC) is a member of AudioCodes family of Enterprise Session Border Controllers, enabling connectivity and security between enterprises and Service Providers' VoIP networks.

The Mediant 1000 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 1000 E-SBC is based on field-proven VoIP and network services with a native host processor with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC functions on the Mediant 1000 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 3000 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 1000 E-SBC is designed as a secured VoIP and data platform. Enhanced Media Gateway security features include encryption schemes, such as SRTP for media, TLS for SIP control, IPSec for management and Denail 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

The Mediant 1000 E-SBC is based on a highly interoperable Media Gateway, which supports a mix of 1-4 E1/T1/J1 Spans, 4-20 BRI lines and 4-24 Analog FXS/FXO interfaces. Enhanced dialing plans and voice routing capabilities along with SIP to SIP mediation, allow Enterprise customers to enjoy the benefits of SIP Trunking services and IP-based Unified Communications, as well as flexible PSTN and legacy PBX connectivity.

Mediant™ 1000 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 1000B E-SBC, with its extensive media processing capabilities, supports a wide range of voice coders with the ability of transcoding between narrowband and wideband voice coders, including 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 gateway, the Mediant 1000B E-SBC provides unparalleled interoperability, enabling mediation between an extensive list of IP and TDM PBXs and SIP Trunking providers.

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

AudioCodes Mediant 1000B E-SBC supports enhanced IP Quality of Service (QoS) enforcement and Quality of Experience (QoE) Monitoring, leveraging AUDC SEM, a QoE monitoring and management system. The Mediant 1000B ESBC 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 1000B 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 of distributed enterprises may face service discontinuities in case of WAN failure. The integrated SAS (Stand Alone Survivability) feature of the Mediant 1000B 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 1000B 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). In addition, an advanced, on-board DSP Resource Farm enables the implementation of narrowband as well as wideband/High Definition VoIP (HDVoIP) media processing services, such as announcements, recording, IVR, conferencing and transcoding, all controlled by standard protocols (e.g., SIP and MSCML). Utilizing AudioCodes dedicated DSP resources enables robust and predictable voice performance, as compared to typical software implementations, based on general purpose CPU's.

Applications

Sip Trunking Solution

Using the Mediant 1000B E-SBC, enterprise customers can seamlessly migrate from legacy PSTN connectivity to cost-effective SIP Trunking Services. The Mediant 1000B-ESBC provides security, session mediation and service level assurance services, connecting the enterprise to multiple SIP Trunking providers, while maintaining interoperability and manageability.

Contact Center Solution

Contact Centers place the SIP Application Server in the LAN, with SIP User Agents deployed remotely across the WAN. The Mediant 1000B E-SBC monitors these User Agents, and resolves any NAT traversal issues they might face. In addition, with its vast media processing features such as Voice Activity Detection, Answering Machine, Call Progress Tone, and DTMF detections, the Mediant 1000B E-SBC provides support for outbound calling campaigns, utilizing the same hardware resources.

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 1000B 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 1000B E-SBC can continue to serve in both internal and external calling capacities.

Mediant 1000B E-SBC in Enterprise Networks

Medium Enterprises are motivated to become more productive, efficient, and responsive to their internal users. The convergence of secured voice services, Stand Alone Survivability, PSTN connectivity, 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

- 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 150
Max. Registered Users	Up to 600
Transcoding Sessions	Up to 60
Interfaces	
PSTN Modularity and Capacity	Voice interface: the Mediant 1000B is equipped with 6 Slots for hosting voice processing and PSTN termination modules (up to 120 TDM-VoIP channels per Gateway)
Digital Module	1, 2 or 4 E1/T1/J1 spans per module using RJ-48c connectors with an option of PSTN Fallback
Analog Module	4 ports FXO or FXS per module using RJ-11 connectors, ground start and loop start
BRI Module	4 BRI ports (8 calls) per module, network S/T interfaces. NT or TE termination
Media Processing Module	Support Media processing options of up to 60 Conferencing legs (3 way or N-way), play, record to IP or PSTN
Networking Interfaces	
Ethernet	3 pairs of Active / Stand-by GE interfaces
OSN Server Platform	
Single Chassis Integration	Embedded, Partner Application Platform for third party services
CPU	Multiple options
Memory	1G RAM or 2G RAM
Storage	Single/Dual hard disk drives
Interfaces	10/100/1000B Base-T, USB, VGA, RS-232 Security
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*, 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: G.711a/mu, G.723.1, G.726, G.727, G.729A/B/E, EG.711, Wireless: GSM-FR, GSM-EFR, MS-GSM, AMR, iLBC, EVRC, QCELP, Wideband: G.722
Signal Conversion	DTMF/RFC2833, Inband/T.38 Fax, Packet-time Conversion
NAT	Local and Far-End NAT Traversal for support of remote workers
Signal Detection	Fixed & Dynamic Gain Control, 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 Prioritization	Multiple queues for granular prioritization of VoIP over other non-real time traffic types, Integrated Queuing and scheduling schemes (Strict Priority, Class based 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
Gain control	Fixed & dynamic voice gain control
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 32 separate LANs
Physical / Environmental	
Enclosure	1U chassis
Dimensions	1U x 320mm x 345mm (HxWxD)
Weight	Approx. 5.95lb (2.7kg) loaded with OSN
Power	100-240 V AC power feeds
Regulatory Compliance	
Telecommunications TBR-3 (BRI interface)	TIA/EIA-IS-968 (FXO, T1) interface, ETSI ES203 021 (FXO interface), TBR-4 (ISDN over E1 interface), TBR13/13 (E1 lines)
Safety and EMC	IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 386
Environmental	Storage: ETS300019-2-1 class T1.2 Transportation: ETS300019-2-2 class T2.3 Operating: ETS300019-2-3

*Roadmap