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 leader focused on VoIP communications, applications and networking elements, and its products are deployed globally in Broadband, Mobile, Cable, and Enterprise networks. The company provides a range of innovative, cost-effective products including Media Gateways, Multi-Service Business Gateways, Residential Gateways, IP Phones, Media Servers, Session Border Controllers (SBC), Security Gateways and Value Added Applications. AudioCodes underlying technology, VoIPerfectHD™, relies primarily 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 emerging Voice networks.

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AudioCodes

Mediant™ 3000

Enterprise Session Border Controller (E-SBC)

- Based on three core foundations: Perimeter Defense, Mediation and Service Assurance
- Standards-based solution with proven interoperability
- Software license scalability from 250 up to 1000 SBC 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
- Carrier-Grade Simplex and High-Availability configurations
- Proven Voice Quality superiority
- Media Processing for Transcoding, Gain Control, DTMF/FAX, etc.
- Extensive filtering and admittance policies

AudioCodes

AudioCodes' Mediant™ 3000 Enterprise Session Border Controller (E-SBC) is a member of AudioCodes family of Enterprise Session Border Controllers, enabling connectivity and security between Enterprise and Service Providers' VoIP networks. The Mediant 3000 E-SBC provides Perimeter Defense as a way of protecting enterprises 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 for high capacity and high performance, the Mediant 3000 E-SBC is based on AudioCodes' VoIPerfect best-of-breed Media Gateway technology, scaling up to 1000 secured SBC VoIP sessions. The native implementation of SBC functions on the Mediant 3000™ VoIP Media gateway platform provides a host of additional capabilities that are not possible with standalone SBC appliances, such as VoIP gateway and PSTN routing functionality. 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.

Integrated PSTN connectivity

Customers can safely and transparently migrate from traditional PSTN to SIP Trunking with the Mediant 3000 E-SBC, a cost-effective method of increasing the value of their data network, while also protecting their investment in legacy PBX equipment. In addition to E1/T1 interfaces, the Mediant 3000 E-SBC supports high-density PSTN interfaces, such as T3, STM-1 and OC3.

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 3000 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 3000 VoIP media gateway, the Mediant 3000 E-SBC provides unparalleled interoperability, enabling mediation between an extensive list of IP and TDM PBXs and SIP Trunking providers.

High Availability and survivability

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 E-SBC is equipped with PSTN interfaces that can also provide local survivability via PSTN fallback connectivity (including E911), when the WAN is unavailable.

MediantTM 3000 Enterprise Session Border Controller (E-SBC)

Applications

SIP Trunking Solution

Using the Mediant 3000 E-SBC, enterprise customers can seamlessly migrate from legacy PSTN connectivity to cost-effective SIP Trunking Services. The Mediant 3000 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

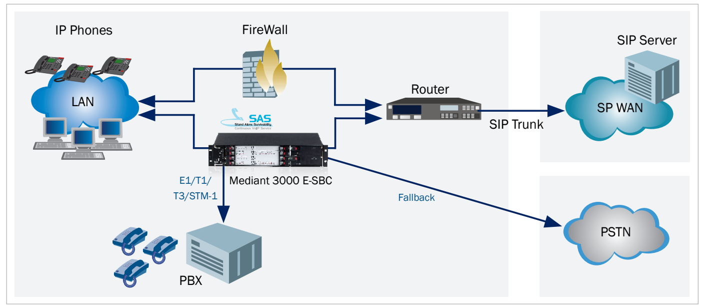
Contact Centers place the SIP Application Server in the LAN, with SIP User Agents deployed remotely across the WAN. The Mediant 3000 E-SBC monitors these User Agents, and resolves any NAT traversal issues they might face. In addition, with its vast media processing features such as Voice Activity Detection, Answering Machine, Call Progress Tone, and DTMF detections, the Mediant 3000 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 3000 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 3000 E-SBC can continue to serve in both internal and external calling capacities.

Mediant 3000 E-SBC in Service Provider Networks

As Enterprises strive to control their communication operating and equipment costs, outsourcing Voice and Data infrastructure to a Service Provider is becoming an attractive option. The Mediant 3000 E-SBC offers Service Providers, who are delivering hosted and managed communication services, a clear and easy-to-manage demarcation point, combining Security, Mediation Services, and Service Level Assurance.



Mediant 3000 E-SBC in Enterprise Networks

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

The high-density Mediant 3000 E-SBC is a well-suited platform for converging VoIP Gateways and Session Border Controllers, thereby improving the enterprise headquarters' service level for local, branch and mobile users.

Benefits for Service Providers

- A highly integrated device for providing SIP Services to Enterprises
- 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

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 1000 SBC, IP to IP transcoding, or SRTP to RTP sessions
Security	
Access Control	Denial & Distributed Denial of Service protection through line rate filtering using White/Black Lists, including bandwidth throttling
VoIP firewall	RTP pinhole management according to SIP offer/answer model. Service theft protection by avoiding rogue RTP
Encryption and Authentication	TLS, SRTP, HTTPS, SSH, IPSec, IKE, SNMPv3, Client-side Authentication, RADIUS
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
ITSP and PBX support	Interoperable with many SIP trunk Service Providers and PBX vendors, such as Verizon, Skype and Microsoft OCS
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
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 T1/E1/J1, T3, OC-3, STM-1 physical interfaces
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, EVRC-B; Wideband: AMR-WB, 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	Voice Activity, Call Progress Tone, and Answering Machine
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
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
Transparent Media	Low latency, unprocessed payload transfer
Gain control	Fixed & dynamic voice gain control
Media Anchoring*	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Redundancy*	"Five 9s" availability with 1+1 hardware redundancy, Active calls preserved
SIP Routing	
Routing methods	Request URL, Source/Destination IP Address, Fully Qualified Doman Name, ENUM*
Alternative Routing and load balancing	Detect proxy failures and route to alternative proxies. Load balance across a pool of proxies
Multiple LANs	Support for up to 16 separate LANs
Hardware Specifications	
IP Networking:	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)
PSTN	1 OC-3 or STM-1 APS optical links, 1 to 3 T3 (DS3) electrical links, up to 63/84 E1/T1 links
Enclosure	4-slot, 2U cPCI chassis
Dimensions	(HxWxD) 88mm x 482.6mm x 296.8mm
Weight	Approx. 35.27 lb (16 kg), fully loaded
Power	48 V DC Dual Feed, with up to 2 DC Power modules, 100–240 V AC redundant Dual Feed
Regulatory Compliance	
Telecommunications	FCC part 68, TBR4 and TBR13
Safety and EMC	• UL 60950 • FCC part 15 Class A • CE Mark (EN55022 Class A, EN60950, EN 55024, EN300 386)
Environmental	NEBS level 3 (on OC3/STM-1/T3 configurations): GR-63-Core, GR-1089-Core, Type 1 & 3, ETS300 019

**to be available on v6.2*