AudioCodes Session Border Controller (SBC) Products

Mediant™ 2600

Hybrid SBC and Media Gateway

The AudioCodes **Mediant 2600 Session Border Controller (SBC)** is a mid-range capacity member of AudioCodes' field-proven hardware-based SBC product family, designed to offer enterprises a reliable and scalable SBC solution. The Mediant 2600 SBC supports wide-ranging SIP interoperability, delivering service assurance and enabling scalable, reliable and secured connectivity between different VoIP networks.

The Mediant 2600 SBC provides a perfect solution for enterprises and large organizations such as contact centers, where security, reliability and high performance are critical.

Benefits

- Pure-IP SBC for medium-sized enterprise deployments
- Offers comprehensive security, interoperability and reliability
- Delivers high service performance and voice quality
- Flexible licensing options for cost-effective scalability

Key Features

- Scalable to 600 SBC sessions
- Extensive SIP mediation capabilities
- Supports remote workers and mobile SIP clients
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Branch survivability during WAN failure
- Active/Standby High Availability
- Advanced media handling including transcoding and wideband speech

Extensive Mediation Capabilities and Proven Interoperability

The Mediant 2600 SBC includes comprehensive media security and SIP normalization capabilities. It offers full interoperability with an extensive list of IP-PBXs, unified communications solutions and SIP trunking provider networks.

Security

The Mediant 2600 SBC provides robust protection for the IP communications infrastructure, preventing fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The Mediant 2600 SBC 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 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
- Residential VoIP

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Specifications

Capacities	
Max. Signaling/Media Sessions	600
Max. SRTP/RTP Sessions	600
Max. Transcoding Sessions	600
Max. Registered Users	8,000
Networking Interfaces	
Ethernet	8 Redundant 100/1000 Base-T Ethernet ports for physical separation between multiple LAN and WAN between Media, Control and OA&M
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, SSH, Client/Server SIP Digest authentication, RADIUS Digest
Privacy	Topology Hiding, User Privacy
Traffic Separation	VLAN/physical interface separation for multiple Media, Control and OAM interfaces
Intrusion Detection	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 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
Transcoding and Vocoders	Coder normalization including: transcoding, coder enforcement and re-prioritization Extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB, AMR-WB, SILK-NB, SILK-WB, OPUS[1]
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
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 Queuing and scheduling schemes (Strict Priority, Class based Prioritization queuing, fairness)
Standalone Survivability	Maintain local calls in the event of WAN failure
Transparent Media	Low latency, unprocessed payload transfer (voice and video supported)
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 De-anchoring	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	AudioCodes Session Experience Manager (SEM)
Redundancy	High availability with two box redundancy, active calls preserved
Quality of Experience	Access control and media quality enhancements based on QoE and bandwidth utilization
Test agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Routing	
Routing Methods	Request URL, IP Address, FQDN, ENUM, advanced LDAP
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, Coder type etc.)
Redundancy	Detect proxy failures and route to alternative proxies
Routing Features	Least cost routing, call forking, load balancing
Multiple LANs	Support for up to 48 separate LANs
SIPRec	IETF standard SIP recording interface
Physical / Environmental	
Dimensions	1U x 444mm x 355mm (HxWxD)
Weight	Approx. 11.7 lbs (5.3Kg)
Mounting	Desktop or 19" rack mount
Power	100–240 V AC redundant dual feed
Operating Temperature	5°-40° C
Regulatory Compliance	
Safety and EMC	UL60950-1 FCC Part 15 Class A ICES-003 Class A CE marking: IEC60950-1, EN55024, EN55022 Class A, EN61000-3-2, EN61000-3-3, ETSI EN300 386

[1] Roadmap

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, VoIPerfect 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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