Mediant™ 2600

Session Border Controller

The AudioCodes **Mediant 2600 session border controller (SBC)** is a mid-range capacity solution for enterprises, delivering service assurance and enabling scalable, reliable and secured connectivity between different VoIP networks.

Scaling up to 600 concurrent sessions, the Mediant 2600 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

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

600 SBC Sessions | Pure IP SBC | 1+1 High Availability | Certified SBC for Teams Direct Routing supporting media optimization

Comprehensive interoperability
Proven interoperability with SIP trunks, SIP platforms and IP cloud services

Flexible licensing
Various licensing options for easy and cost-effective scalability regardless of enterprise size

Enhanced security
Robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft

Superior voice quality
Advanced capabilities for optimizing and monitoring voice service quality

High resiliency
High availability using 1+1 redundancy and local branch survivability

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Specifications

Capacities			
Max. Signaling	600	**Max. RTP/SRTP Sessions**	600
Max. Registered Users	8,000	**Max. Transcoding Sessions**	600

Network Interfaces	
Ethernet	8 100/1000 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 (Intrusion Detection System)
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
Encryption/Authentication	TLS, DTLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
Privacy	Automatic topology hiding, user privacy
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces
Interoperability	
SIP B2BUA	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
SIP Interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer and more
Registration and Authentication	SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
Transport Mediation	Mediation between SIP over UDP/TCP/TLS/WebSocket, IPv4/IPv6, RTP/SRTP (SDES/DTLS)
Header Manipulation	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
Number Manipulations	Ingress and 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.729A/B, GSM-FR, AMR-NB/WB, SILK-NB/WB, Opus-NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, DTLS, RTP multiplexing, secure RTCP with feedback.
NAT	Local and far-end NAT traversal for support of remote workers, ICE full and lite support (RFC 8445)
Voice Quality and SLA	
Call Admission Control	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
Packet Marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintains local calls in the event of WAN failure.
Voice Monitoring and Enhancement	Transrating, RTCP-XR, acoustic echo cancellation, replacing voice profile due to impairment detection, fixed and dynamic voice gain control, packet loss concealment, dynamic programmable jitter buffer, silence suppression/comfort noise generation, RTP redundancy, broken connection detection
Direct Media	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
High Availability	SBC high availability with two-box redundancy, active calls preserved
Test Agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Call Handling	
Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
Querying External Databases	Destinations based on customized queries of ENUM, LDAP, HTTP server (REST API)
Available Destinations	Configured SIP peers, registered users, IP address, request URI
Advanced Features	Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization
SBC Media Types	Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)
SIPREC	IETF standard SIP recording interface, supporting both audio and video SBC sessions
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, One Voice Operations Center (OVOC)
Multi Tenancy	Advanced multi-tenant SBC partitioning
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 VAC redundant dual feed (hot-swappable)
Operating Temperature	5°-40° C

1 Hayarden Street, Airport City, Lod, 7019900, Israel
Tel: +972-3-976-4000

80 Kingsbridge Rd - Piscataway, NJ 08854
Tel: +1-732-469-0880

www.audiocodes.com

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