

### 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

## 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 OAM&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, lite ICE, DTLS, RTP multiplexing, secure RTCP with feedback.		
NAT	Local and far-end NAT traversal for support of remote workers		
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, RTPC-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		