# 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 | Supports OPUS and SILK**

### 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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Mediant™ 2600

## 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, 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, 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,<br>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 |

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