AudioCodes Session Border Controller (SBC) Products

# Mediant™ 2600

## Hybrid SBC and Media Gateway

### 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

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.

### 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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# Mediant™ 2600

## 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<br>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<br>FCC Part 15 Class A<br>ICES-003 Class A<br>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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