# Mediant™ 500

## Hybrid SBC and Media Gateway

The AudioCodes **Mediant 500 enterprise session border controller (E-SBC)** and media gateway is a compact, high performance VoIP connectivity solution for small enterprises and branch office locations.

Scaling up to 250 concurrent sessions, the Mediant 500 connects IP-PBXs to any SIP trunking service provider and offers superior performance in connecting any SIP to SIP environment.

The Mediant 500 also supports up to 30 voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN.

**250 SBC Sessions | 30 TDM Sessions | 1+1 High Availability | WebRTC Gateway**

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

### Hybrid functionality
True hybrid SBC and gateway platform for gradual migration and reduced space and power footprints

### 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, local branch survivability and Ethernet redundancy

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Mediant™ 500

## Specifications

| Capacities | | | |
|---|---|---|---|
| **Max. Signaling** | 250 | **Max. RTP/SRTP Sessions** | 200 |
| **Max. Registered Users** | 1,500 | | |

| Telephony Interfaces | |
|---|---|
| **Digital** | Single E1/T1 interface |
| **Clock Source** | 5 ppm High Precision |
| **Digital PSTN Protocols** | Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS- 100 and others. Different CAS protocols, including MFC R2, E&M immediate start, E&M delay dial/start and others. |
| **Network Interfaces** | |
| **Ethernet** | 4 GE interfaces configured in 1+1 redundancy or as individual ports |
| **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, 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 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 |
| **Registration and Authentication** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP (SDES) |
| **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 |
| **SIP Interworking** | 3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer |
| **Signal Conversion** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **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. Outbound calls can use PSTN fallback (including E911). |
| **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 |
| **SIPREC** | IETF standard SIP recording interface |
| **Management** | |
| **OAM&P** | Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, OVOC |
| **Physical/Environmental** | |
| **Dimensions** | 43.7 (1U) x 310 x 210 mm (HxWxD) |
| **Weight** | 4.4 lb (2.0kg) |
| **Mounting** | Desktop or 19" rack mount |
| **Power** | 100-240V, 50-60 Hz, 0.8A |
| **Environmental** | Operational: 0 to 40°C (41 to 104°F); Storage: -25 to 70°C (-13 to 185°F)<br>Relative Humidity: 10 to 90% non-condensing |

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