AudioCodes Mediant software session border controller (SBC) is a highly scalable SBC solution supporting broad SIP interoperability, advanced media handling and robust security. AudioCodes Mediant software SBC enables enterprises and service providers to deliver voice services, such as SIP trunking and unified communications, via private or public clouds.

The Mediant software SBC is available in three variants to meet different customer deployment needs:

- **Mediant CE** | a cloud-native SBC delivering high scalability and elasticity in virtualized cloud environments
- **Mediant VE** | built for deployment in virtualized data centers, public clouds and NFV environments
- **Mediant SE** | designed to run on commercial off-the-shelf servers (COTS) in high-scale communications environments

All Mediant software SBC variants provide the following:

- **Comprehensive SBC functionality and SIP interoperability**
  - Shared code base with AudioCodes field-proven, hardware-based SBCs

- **Rapid cloud deployment**
  - Optimized resource consumption for private and public clouds such as OpenStack and Amazon Web Services

- **NFV-ready**
  - Proven interoperability with leading NFV orchestrators

- **Enhanced scalability**
  - Easily scale from tens up to tens of thousands of concurrent sessions

- **High availability**
  - 1:1 active-standby configuration for business continuity

- **High performance and robust security**
  - Built-in software-based media transcoding with support for encryption and protection from attacks

- **Qualified for leading UC and hosted telephony platforms**
  - Supported environments include Microsoft Teams, Skype for Business and BroadSoft BroadWorks

- **Integrated WebRTC gateway**
  - Simple and secure WebRTC deployment, supporting both signaling and media
AudioCodes Session Border Controllers

### Specifications

<table>
<thead>
<tr>
<th>Capacities</th>
<th>Mediant CE</th>
<th>Mediant VE</th>
<th>Mediant SE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max. Signaling Sessions</td>
<td>50,000</td>
<td>24,000</td>
<td>55,000</td>
</tr>
<tr>
<td>Max. Transcoding Sessions</td>
<td>50,000</td>
<td>24,000</td>
<td>55,000</td>
</tr>
<tr>
<td>Max. SRTP-RTP Sessions</td>
<td>50,000</td>
<td>10,000</td>
<td>40,000</td>
</tr>
<tr>
<td>Max. Transcoding</td>
<td>27,000</td>
<td>12,000 *</td>
<td>25,000 *</td>
</tr>
<tr>
<td>Max. Registered Users</td>
<td>100,000</td>
<td>75,000</td>
<td>300,000</td>
</tr>
</tbody>
</table>

### Security

- **Access Control**: DoS/DDoS line protection, bandwidth throttling, dynamic blacklisting
- **VoIP Firewall**: RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
- **Encryption and Authentication**: TLS, DTLS, 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
- **Intrusion Detection System**: Detection and prevention of VoIP attacks, theft of service and unauthorized access

### 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
- **Transport Mediation**: SIP over UDP/TCP/TLS/WebSocket, IPv4 / IPv6, RTP / SRTP (SDP/DTLS)
- **Header Manipulation**: Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
- **URI and Number Manipulations**: URI user and host name 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, AMR-WB (G.722.2), SLK-NB/WB, Opus-NB/WB
- **Signal Conversion**: DTMF/RFC 2833/SIP T.38 fax, packet-time conversion
- **WebRTC Controller**: Interworking between WebRTC devices and SIP networks. Supports WebSocket, Opus, VP8 video coder, 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**: Based on bandwidth, session establishment rate, number of connections/registrations
- **Packet Marking**: 802.1p/Q VLAN tagging, DiffServ, TOS
- **Standalone Survivability**: Maintains local calls in the event of WAN failure.
- **Impairment Mitigation**: Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation, RTP redundancy, broken connection detection
- **Voice Enhancement**: Translating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, fixed & dynamic voice gain control
- **Direct Media**: Hairpinning of local calls to avoid unnecessary media delays and bandwidth consumption while avoiding media anchoring
- **Voice Quality Monitoring**: RTCP-XR, AudioCodes Session Experience Manager (SEM)
- **High Availability**: SBC high availability with two-box redundancy, active calls preserved
- **Quality of Experience**: Access control and media quality enhancements based on QoS 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, third-party routing control through REST API
- **Advanced Routing Criteria**: QoS, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
- **Redundancy**: Detection of proxy failures and subsequent routing to alternative proxies
- **Routing Features**: Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization
- **SIPRec**: IETF standard SIP recording interface

### Management

- **OAMP/IM**: Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS, HTTP reverse proxy
- **Multi Tenancy**: Advanced multi-tenant SBC partitioning
- **Deployment tools**: VNFM/Stack manager (Mediant CE), HEAT templates, Cloud Formation
- **Auto-scaling (CE)**: Automatic, REST API, CLI, Web UI

### Mediant VE SBC Minimum Requirements

<table>
<thead>
<tr>
<th>Hypervisor</th>
<th>Virtual Resources</th>
<th>1 vCPU, 2 GB RAM, 10 GB Disk, Virtual NICs - 2 (Standalone)/3 (HA)</th>
</tr>
</thead>
</table>

* With media transcoding cluster