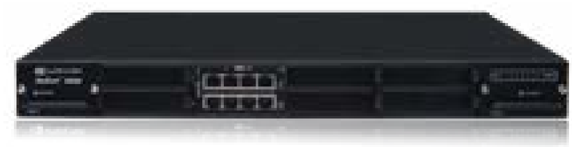
# Mediant™ 4000

## Session Border Controller

AudioCodes **Mediant 4000 Session Border Controller (SBC)**는 기업 및 서비스 공급자를 위한 중대 규모 솔루션으로서 서로 다른 VOIP 네트워크 간에 안정적이며 안전한 연결과 더불어 확장성까지 지원합니다.



최대 5,000개의 동시 세션까지 확장할 수 있는 Mediant 4000은 IP-PBX를 모든 SIP 트렁킹 서비스 공급자에 연결하고 모든SIP를 SIP 환경에 연결할 때 탁월한 성능을 제공합니다.

Mediant 4000은 보안, 안정성 및 고효율성을 요구하는 기업 및 Contact Center, 대규모 데이터 센터, 호스팅 서비스 공급자 및 정부 기관과 같은 대규모 조직을 위한 완벽한 솔루션입니다.

**5,000 SBC 세션 | Pure IP SBC | 1+1 고가용성 | 미디어 최적화를 지원하는 Teams Direct Routing 인증 SBC**

### 포괄적인 상호 운용성
SIP 트렁크, 통합 커뮤니케이션 솔루션, PBX 및 IP 클라우드 서비스와 입증된 상호 운용성

### 유연한 라이선스 제공
기업 규모에 상관없이 쉽고, 효율적인 확장을 위한 다양한 라이선스 옵션 제공

### 강화된 보안
사이버, DoS 및  DDoS 공격  뿐만 아니라 도청, 사기 및 서비스 도난에 대한 강력한 경계 방어.
FIPS(연방 정보 처리 표준) 140-2 레벨 1 준수

### 뛰어난 음성 품질
음성 서비스 품질 최적화 및 모니터링을 위한 고급 기능 지원

### 높은 복원력
1+1 이중화 및 로컬 분기 존속성을 사용한 고가용성

## Specifications

| Capacities | | |
|---|---|---|
| | **Mediant 4000** | **Mediant 4000B** |
| **Max. Signaling** | 5,000 | 5,000 |
| **Max. RTP/SRTP Sessions** | 5,000/3,000 | 5,000 |
| **Max. Transcoding Sessions** | 2,400 | 5,000 |
| **Max. Registered Users** | 20,000 | 20,000 |

| 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 |
| **Certification** | FIPS 140-2 Level 1 |
| **STIR/SHAEKN** | STIR/SHAKEN support. Interworking with STI-AS/VS |

| 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 |
| **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), 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** | Decisions 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 |
| **Multiple LANs** | Support for up to 48 separate LANs |
| **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, CDR,One Voice Operations Center (OVOC) |
| **Multi Tenancy** | Advanced multi-tenant SBC partitioning |

| Physical/Environmental | | |
|---|---|---|
| | **Mediant 4000** | **Mediant 4000B** |
| **Dimensions** | 1U x 19" (444mm) x 14" (355mm) (HxWxD) | 1U x 19" (444mm) x 14.9" (378mm) (HxWxD) |
| **Weight** | Approx. 11.7 lbs (5.3Kg) | Approx. 16.3 lbs (7.4Kg) |
| **Mounting** | Desktop or 19" rack mount | |
| **Power** | Dual power supply (hot- swappable):<br>100-240VAC 50-60Hz 2.5A max | Dual universal power supply (hot- swappable):<br>40-60VDC, 17A max., or 100-240 VAC, 50-60 Hz, 7A max |
| **Environmental** | 5°-40° C | |

* Requires a dedicated software build

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