Mediant™ 800

Hybrid SBC and Media Gateway

AudioCodes **Mediant 800 Enterprise Session Border Controller (E-SBC)**와 Media Gateway는 중소기업을 위한 완벽한 연결 솔루션입니다.

최대 400개의 동시 세션까지 확장할 수 있는 Mediant 800은 IP-PBX를 모든 SIP 트렁킹 서비스 공급자에 연결할 수 있고 모든 SIP를 SIP 환경에 연결할 때 우수한 성능을 제공합니다.

또한 Mediant 800은 1U 플랫폼에서 최대 124개의 음성 채널을 지원하여 전통적인 TDM PBX 시스템과 IP 네트워크 연결 또는 IP-PBX와 PSTN 연결과 같이 TDM과 VoIP 네트워크 간의 다양한 연결을 가능하게 합니다.

400 SBC 세션 | 124 TDM 세션 | 1+1 고가용성| Teams Direct Routing 인증 SBC | OPUS와 SILK 지원

포괄적인 상호 운용성
SIP 트렁크, SIP 플랫폼 및 IP 클라우드 서비스와의 입증된 상호 운용성

하이브리드 기능
시스템 마이그레이션을 위한 진정한 하이브리드 SBC 및 Gateway 플랫폼, 낮은 CAPEX와 공간 및 전력 소모 감소

강화된 보안
사이버, DoS 및 DDoS 공격 뿐만 아니라 도청, 사기 및 서비스 도난에 대한 강력한 경계 방어

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

높은 복원력
1+1 이중화, 로컬 분기 생존 가능성 및 PSTN 폴백을 사용한 고가용성

Mediant™ 800

Specifications

Capacities	Max. Signaling	Max. RTP/SRTP Sessions	Max. Transcoding Sessions	Max. Registered Users
Mediant 800B	250	250/250	57	1500
Mediant 800C	400	400/300	114	2000

Telephony Interfaces	
Analog	4 FXS/FXO ports
Digital	Up to 4 E1/T1 interfaces; 4/8 BRI Ports
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 or 4 GE + 8 FE 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, 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
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 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, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB, iLBC
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion, V.150.1
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks. Supports WebSocket, Opus, VP8 video coder, lite ICE, DTLS, RTP multiplexing.
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
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, One Voice Operations Center (OVOC)

Physical/Environmental			
Dimensions	1U x 345mm x 320mm (HxWxD)	**Weight**	Approx. 5.95lb (2.7kg) loaded with OSN
Mounting	Desktop or 19" rack mount	**Operating Temperature**	5°-40° C
Power	Internal AC power supply rated: 100-240 VAC ~50- 60Hz 1.5A maximum (Optional) Additional 12V 10A DC power, via an AudioCodes external AC/DC power adaptor		

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