Mediant™ CE/VE/SE

Mediant CE/VE/SE Software Session Border Controller (SBC)

AudioCodes **Mediant Software Session Border Controller (SBC)**는 광범위한 SIP 상호 운용성, 훌륭한 미디어 처리 및 강력한 보안을 지원하는 확장성이 뛰어난 SBC 솔루션입니다. AudioCodes Mediant 소프트웨어 SBC는 기업 및 서비스 공급자가 Private 또는 Public Cloud를 통해 SIP 트렁킹 및 통합 커뮤니케이션과 같은 음성 서비스를 제공할 수 있게 합니다.

Mediant 소프트웨어 SBC는 다양한 고객의 배포 요구 사항을 충족시키기 위하여 아래와 같은 세 가지 방식으로 제공됩니다.

Mediant CE | 가상화 된 클라우드 환경에서 높은 확장성과 탄력성을 제공하는 클라우드 네이티브 SBC
Mediant VE | 가상화 된 데이터 센터, Public 클라우드 및 NFV 환경 배포를 위해 구축
Mediant SE | 대규모 통신 환경에서 Commercial off-the-shelf servers(COTS)에 실행하도록 설계

포괄적인 SBC 기능 및 SIP 상호 운용성
현장에서 검증된 AudioCodes의 하드웨어 기반 SBC로 코드 베이스 공유

신속한 Cloud 배포
Microsoft Azure 및 Amazon Web Services, Google Cloud Platform과 같은 private 및 public Cloud 에서 최소한의 리소스 사용

NFV-ready
선도적인 NFV 오케스트레이터와의 입증된 상호 운용성

향상된 확장성
10개에서 수만개의 세션까지 쉽게 확장 가능

High availability
비즈니스 연속성을 위한 1:1 active-standby 구성

고성능 및 강력한 보안
암호화 및 공격 보호를 지원하는 내장형 소프트웨어 기반 미디어 트랜스코딩 지원

선도적인 UC 및 호스트형 텔레포니 플랫폼에 대한 인증
미디어 최적화를 지원하는 Teams Direct Routing 인증 SBC

통합 WebRTC Gateway
신호 및 미디어를 모두 지원하는 간단하고 안전한 WebRTC 배포

Specifications

Capacities	Mediant CE	Mediant VE	Mediant SE
Max. Signaling Sessions	40,000	24,000	70,000
Max. Media Sessions	40,000	24,000	70,000
Max. SRTP-RTP Sessions	40,000	10,000	40,000
Max. Transcoding Sessions	27,000	12,000 [1]	30,000 [1]
Max. Registered Users	100,000	75,000	500,000

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, advanced RTP latching
Encryption and 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
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
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/SCTP, IPv4 / IPv6, RTP / SRTP (SDES/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), SILK-NB/WB, Opus-NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, packet-time conversion
WebRTC Gateway	Interworking between WebRTC devices 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	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	Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control
Direct Media	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption while avoiding media anchoring
Voice Quality Monitoring	RTCP-XR, AudioCodes One Voice Operations Center (OVOC)
High Availability	SBC 1+1 high availability with active calls preservation
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, third-party routing control through REST API
Advanced Routing Criteria	QoE, 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
SBC Media Types	Audio\Video\Fax\Text\Message Session Relay Protocol (MSRP)\Binary Floor Control Protocol (BFCP)
Recording Solutions	Lawful Interception (LI[2]), SIPREC for both audio and video sessions
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, HTTP reverse proxy One Voice Operations Center (OVOC), Session Detail Records (SDRs)
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
Cloud Environments	
Public Cloud	Azure, AWS, GCP
Private Cloud	OpenStack, VMware® vSphere

Mediant VE SBC Minimum Requirements			
Hypervisor	VMware® vSphere ESXi™ 5.5, & 6.5 and above, Linux KVM, Microsoft Hyper-V	**Virtual Resources**	1 vCPU; 2 GB RAM; 10 GB Disk; Virtual NICS - 2/3 (Standalone/HA)

[1] With media transcoding cluster
[2] Requires a dedicated software build

About AudioCodes

AudioCodes Ltd. (NasdaqGS: AUDC)는 디지털 작업 공간을 위한 고급 음성 네트워킹 및 미디어 처리 솔루션의 선두 공급업체입니다. 기업 DNA의 일부인 인간 목소리에 대한 약속으로 AudioCodes는 기업과 서비스 제공업체가 통합 커뮤니케이션, 컨택 센터 및 호스팅된 비즈니스 서비스를 위한 IP 음성 네트워크를 구축하고 운영할 수 있도록 합니다. AudioCodes의 광범위한 혁신적인 제품, 솔루션 및 서비스는 대규모 다국적 기업과 전 세계의 선도적인 1차 사업자들에 의해 사용되고 있습니다.

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