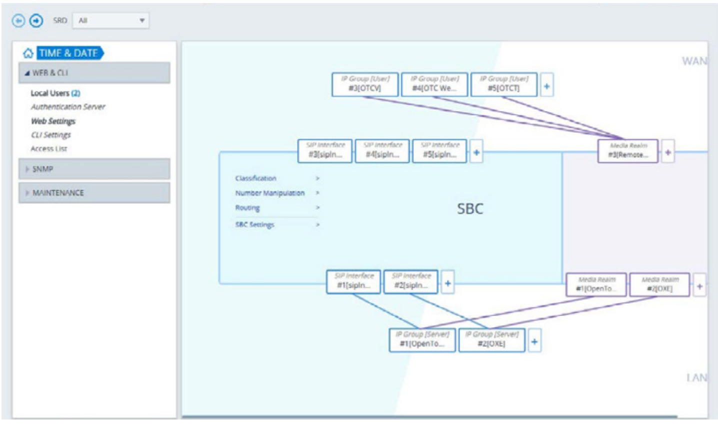
Alcatel·Lucent Enterprise

Alcatel-Lucent OpenTouch Session Border Controller

Protect SIP trunks and enterprise communications with a highly secure SIP perimeter defense solution



The Alcatel-Lucent OpenTouch® Session Border Controller (OpenTouch SBC) addresses the communication security needs of mid-sized and large organizations by protecting them from malicious VoIP attacks, SIP denial of service, and fraud and eavesdropping.

As a highly secure software solution for perimeter defense, OpenTouch SBC acts as the demarcation point between the enterprise and SIP trunking providers. OpenTouch SBC also protects mobile workers and secures their SIP audio and video communications over the internet.

Features	Benefits
Enterprise perimeter defense against SIP denial of service, fraud and eavesdropping	Security: Reinforces firewalls with dedicated protection against SIP-based attacks
Secure and scalable SIP/media connectivity, audio transcoding an network address translation (NAT) traversal for audio and video communications	Cost-saving: Ensures cost-effective, secure conversations over the internet and with SIP service providers
Web-based management with built-in configuration templates: settings and protocol adaptations for certified SIP trunking providers can be configured in a few clicks	Cost-effective interoperability: Offers protocol adaptations for many SIP trunking providers
Redundant servers with SIP and media session preservation	Business continuity: Offers always-on, off-net and mobile communications
VMware vSphere Hypervisor, Microsoft Hyper-V and KVM support	Agile operations: Leverages virtualization infrastructure and skills

Technical specifications

Solutions

- OEM AudioCodes Mediant Virtual Edition
- SIP trunking security solution for:
  - Alcatel-Lucent OmniPCX® Enterprise Communication Server R101 and above
- SIP remote worker security solution for:
  - Alcatel-Lucent OmniPCX Enterprise Communication Server R101 and above
  - ALE SoftPhone
  - ALE DeskPhone (ALE-300, ALE-400, ALE-500)
  - HTTPS Reverse Proxy Light for access to OXE SIP Device Management
- Private SIP trunking with Microsoft Teams Direct Routing:
  - Alcatel-Lucent OmniPCX Enterprise Communication Server R101 and above
- SIPREC trunk recording solution for:
  - OmniPCX Record R2.54 and above

Security

- Miercom certified
- Distributed denial of service (DDOS) prevention: L3/L4 and SIP
- SIP stateful inspection: Prevents DDOS attacks based on fraudulent SIP messages
- SIP topology hiding: SIP headers that disclose internal IP topology are removed or modified
- Secure SIP over Transport Layer Security (TLS) (SIPS): Encryption and authentication of SIP messages, SIP over WSS for WebRTC
- Secure Real-time Transport Protocol (SRTP): Encryption of audio and video streams SDES and DTLS crypto key negotiation (AES 128/256)
- Dynamic audio and video port firewall pinholing
- Signature based SIP Intrusion Detection System (IDS) and dynamic blacklisting
- SIP authentication (http digest) of clients and gateways
- Enhanced media latching
- Integrated NGINX Light Reverse Proxy for HTTPS and LDAP traffic
- Complements NGINX+ standalone for low end
- LDAP authentication

Management

- Manageable by AudioCodes One Voice Operation Center (OVOC) platform
- Secured web-based management
- Zero user management: Provisioning of directory number, SIP user information and security credentials are delegated to the communication server
- Simple Network Management Protocol (SNMP)
- Built-in SBC wizard application for SIP trunking and remote worker scenarios
- Multi-Tenancy for OTEC (SBC only, no Reverse Proxy)

Business continuity

- Alternative routing and load balancing:
  - Detects lost connectivity to the communication server and to the SIP provider's proxy servers, and routes to alternative servers
  - Supports OmniPCX Enterprise geographic redundancy
  - Supports load balancing across a pool of SIP provider proxy servers
  - Least-cost routing (based on date, time and cost)
- High-availability option: Active/standby two-server redundancy
  - Active SIP and media sessions are preserved
  - Virtual IP
- Software upgrade without interruption

Interoperability and protocols

- SIP B2BUA: SIP transparency
- SIP WebRTC gateway
- RFCs supported: RFC 2327, RFC 2617, RFC 2782, RFC 2833, RFC 2976, RFC 3261, RFC 3262, RFC 3263, RFC 3264, RFC 3265, RFC 3311, RFC 3323, RFC 3325, RFC 3362, RFC 3420, RFC 3455, RFC 3489, RFC 3515, RFC 3550, RFC 3581, RFC 3611, RFC 3665, RFC 3666, RFC 3711, RFC 3725, RFC 3824, RFC 3842, RFC 3891, RFC 3892, RFC 3903, RFC 3960, RFC 3966, RFC 4028, RFC 4117, RFC 4168, RFC 4235, RFC 4244, RFC 4320, RFC 4321, RFC 4475, RFC 4566, RFC 4568, RFC 4582, RFC 4730, RFC 4733, RFC 4960, RFC 4961, RFC 4975, RFC 5022, RFC 5079, RFC 5124, RFC 5245, RFC 5389, RFC 5628, RFC 5761, RFC 5763, RFC 5764, RFC 5806, RFC 5853, RFC 6035, RFC 6135, RFC 6140, RFC 6188, RFC 6337, RFC 6341, RFC 6442, RFC 7245, RFC 7261, RFC 7865, RFC 7866, RFC 8068
- Transport mediation: SIP over UDP to SIP over TCP, or SIP over TLS, or SIP over WSS
- SIP call-flow mediation
- Real-time audio mediation option: RTP to SRTP encryption
- Extensive SIP profile configuration with third-party SIP Providers
- Extensive SIP signaling interworking: 3xx forwarding Termination, Refer to Reinvite, Diversion Header to History Info, Prack and Update termination
- Programmable header manipulation: Ability to add, modify and delete headers
- Programmable SDP manipulation: Codec list rewriting
- Programmable routing methods: Request URL, source/destination IP address, fully qualified domain name, ENUM, Lightweight Directory Access Protocol
- Uniform resource identifier (URI) and number manipulations:
  - URI user and host name manipulations
  - Ingress and egress digit manipulations
- NAT traversal: Local and far end NAT traversal for support of remote workers
- Audio and video codec filtering
- Audio software transcoding:
  - inband DTMF
  - G711A/G711Mu law
  - Opus, Silk

Media quality and reporting

- Packet marking: 802.1p/Q VLAN tagging, DiffServ, TOS
- Media Anchoring or Direct Media
- Transparent media: Low latency, unprocessed payload transfer
- Voice quality measurement: Voice quality call detail record generation
- RTP Control Protocol-XR support with SIP Publish
- Call Admission Control on media bandwidth, including audio and video
- Allocation of a minimal number of sessions to dedicated SIP interfaces
- Alternative routing based on quality and bandwidth

Capacity and VM prerequisites (VMWare)	Virtual Edition high end	Virtual Edition medium	Virtual Edition low end
Max. SIP endpoints/TLS sessions	6000/6000	6000/6000	1000/100
Max. SIP sessions	4000	2600/1900/1600	250
Max. RTP/SRTP (*0/1/n vCPUs transcoding)	4000	*2600/1900/1600	250
vCPUs/GB RAM/GB HDD/ HyperThreading (HT)	4vCPUs/16 GB RAM/ 10 GB HDD/HT	1vCPU/8 GB RAM/ 10GB HDD/HT	1vCPU/2 GB RAM/ 10 GB HDD/HT
Transcoding	by adding 4 or 12 vCPUs	by adding 1 or 3 vCPUs	N/A

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