The Alcatel-Lucent OpenTouch™ Session Border Controller, Release 2.0, is a flexible perimeter defense solution for SIP conversations.

The OpenTouch Session Border Controller (SBC) is a highly secure software solution for Session Initiation Protocol (SIP) perimeter defense. Located at the customer premises, it is used to protect enterprises from malicious Voice over IP (VoIP) attacks. It simplifies interoperability with SIP service providers.

The OpenTouch SBC provides a flexible architecture for all enterprise deployments, acting as the demarcation point between the enterprise and SIP trunking providers, as well as the enterprise and OpenTouch SIP clients in remote locations to provide direct voice and video conversations over the Internet. The OpenTouch SBC supports up to 4000 SIP audio sessions per server.

**FEATURES**
- Enterprise perimeter defense against SIP denial of service, fraud and eavesdropping
- Certified with SIP service providers
- Addresses the communication security needs of mid-sized and large organizations
- Enables SIP protocol adaptations for interoperability
- Provides secure and scalable SIP/media connectivity and network address translation (NAT) traversal for collaborative OpenTouch voice and video conversations over the Internet
- Acts as a secure softphone proxy for enterprises that need a demarcation point between a segregated voice network and softphones that are in an all-purpose data network
- Provides business continuity over redundant servers with SIP and media session preservation
- Runs on a commercial off-the-shelf (COTS) server and on VMware®
- Provides easy-to-use web-based management
- Provides a configuration wizard application that accelerates interoperability operations

**BENEFITS**
- Provides security between the enterprise and SIP trunking providers
- Complements the enterprise firewall with dedicated protection against SIP-based attacks
- Simplifies the interoperability with various flavors of SIP trunking
- Enables cost-effective and secure conversations with OpenTouch remote workers over the Internet
- Solves SIP and media traversal of NAT devices
- Provides an easy-to-manage central demarcation point between softphones on an untrusted network and the communications network
- Monitors voice quality for service level agreements (SLAs)
- Improves the total cost of ownership with a high-performance solution running on a COTS server and on VMware
TECHNICAL SPECIFICATIONS

Solutions
- SIP trunking security solution for:
  - Alcatel-Lucent OmniPCX™ Enterprise Communication Server 10.1 and above
  - Alcatel-Lucent OpenTouch Business Edition 1.1 and above
- SIP remote worker and secure softphone proxy solution for:
  - Alcatel-Lucent OmniPCX Enterprise Communication Server 10.1 and above
  - Alcatel-Lucent OmniTouch™ 8400 Instant Communication Suite 6.7 and above
- Alcatel-Lucent OpenTouch Business Edition 1.1 and above
- Alcatel-Lucent OpenTouch Multimedia Services 1.1 and above
- My IC Desktop (SIP), OpenTouch Conversation and Connection software clients
- SIP and Real-time Transport Protocol (RTP) encryption (Transport Layer Security (TLS) TLS and Secure RTP (SRTP))
- Alcatel-Lucent OpenTouch Multimedia Services 2.0 and above
- Alcatel-Lucent OmniTouch 8082
  - My IC Phone

Security
- 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 TLS (SIPs): Encryption and authentication of SIP messages
- SRTP: Encryption of audio and video streams
- Support for concurrent SRTP and RTP sessions
- Dynamic audio and video port firewall pinholing
- SIP Intrusion Detection System (IDS) and dynamic blacklisting
- SIP authentication (http digest) of clients and gateways
- SIP authentication through external RADIUS server
- Enhanced media latching

Capacity and recommended hardware
- Server Edition
  - Up to 5000 registered SIP endpoints (5000 TLS sessions)
  - Up to 4000 SIP or SIPS audio sessions, 2000 video sessions per server
  - Up to 4000 RTP sessions, 2000 SRTP sessions
  - Supported server: HP® Proliant™ DL320 G8v2, DL320 G8v1, DL120 G7
  - Software delivery
  - Virtual Edition
  - Up to 1000 registered SIP endpoints (1000 TLS sessions)
  - Up to 250 SIP Sessions
  - VMware ESXi™ version 5.1 or later
  - 2 cores, 2 GB RAM, 10 G0 HDD

Management
- 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)
- SBC wizard application for SIP trunking and remote worker scenarios

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 (from OpenTouch SBC 2.0)

Interoperability and protocols
- SIP B2BUA: SIP transparency
- RFCs partially supported: RFC 4235, RFC 3960
- Transport mediation: SIP over User Datagram Protocol to SIP over Transmission Control Protocol or SIP over TLS
- SIP call-flow mediation
- Real-time audio mediation option: RTP to SRTP encryption
- Extensive SIP profile configuration with third-party vendors
- 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
- Software-based media transrating (frame adaptation)

Media quality and reporting
- Packet marking: 802.1p/Q VLAN tagging, DiffServ, TOS
- 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