The AudioCodes Software Enterprise Session Border Controller (E-SBC) Server Edition and Virtual Edition provide a flexible and scalable SBC solution that meets the requirements of today’s datacenter infrastructures. E-SBC supports wide-ranging SIP interoperability, ensures high-quality service, and enables scalable, reliable, and secure connectivity between different VoIP networks.

**Server Edition**
- Runs on dedicated, commercial, off-the-shelf (COTS) servers
- Aimed at highly scalable environments

**Virtual Edition**
- Runs in virtualized datacenter environments
- Supports VMware, Hyper-V, and KVM systems
- Cloud environments: OpenStack, Amazon Web Services, and Cloudband
- Extensive Mediation Capabilities and Proven Interoperability

E-SBC includes comprehensive media security and SIP normalization capabilities. It provides full interoperability with an extensive list of IP PBX systems, unified communications solutions, and SIP-trunking services.

**Security and Reliability**
E-SBC offers robust protection for an IP communications infrastructure. It prevents fraud and service theft and guards against cyber-attacks and other events that could impact service. E-SBC also maintains the high quality needed for enterprise VoIP communications with active/standby high availability. Advanced call routing mechanisms, network voice quality monitoring, and branch survivability capabilities minimize communications downtime.

**Applications**
- SIP trunking
- Hosted PBX and UCaaS
- IP contact centers
- Remote and mobile worker support
- SIP mediation between UC and IP PBX systems
- Residential VoIP

**Benefits**
- Designed for deployment in standardized datacenter environments
- Supports network function virtualization
- Simplifies and accelerates SBC deployments
- Offers comprehensive security, interoperability, and reliability
- Delivers excellent service performance and voice quality
- Flexible licensing options for cost-effective scalability
- Runs on dedicated COTS servers and in virtualized environments
### Key Features
- Scalable to thousands of SBC sessions
- Extensive SIP mediation capabilities
- Supports remote workers and mobile SIP clients
- Perimeter defense against denial of service, fraud, and eavesdropping
- VoIP quality monitoring and enforcement
- Branch survivability during WAN failure
- Active/standby high availability

#### Capacities

<table>
<thead>
<tr>
<th>Feature</th>
<th>Virtual Edition</th>
<th>Server Edition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max Signaling Sessions</td>
<td>6,000</td>
<td>32,000</td>
</tr>
<tr>
<td>Max Media Sessions</td>
<td>6,000</td>
<td>24,000</td>
</tr>
<tr>
<td>Max SRTP–RTP Sessions</td>
<td>4,000</td>
<td>16,000</td>
</tr>
<tr>
<td>Max Registered Users</td>
<td>30,000</td>
<td>120,000</td>
</tr>
</tbody>
</table>

#### Security
- Access Control
  - DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting
- VoIP Firewall
  - RTP protocol management, rogue RTP detection and prevention, SIP message policy, advanced RTP patching
- Encryption and Authentication
  - TLS, DTLS, SRTP, HTTPS, SIM/client, server SIP Digest authentication, RADIUS Digest
- Privacy
  - 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 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 for SBC users
- Transport Mediation
  - SIP over UDP/TCP/TLS/WebSocket, 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 hostname 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.729, GSM-FR, AMR-NB/WB, SILK-NB/WB, Opus-NB/WB
- Signal Conversion
  - RTP RFC 2833, SIP T.38 fax, packet time conversion
- WebRTC Controller
  - Interworking between WebRTC devices and SIP networks Supports WebSocket, Opus, VP8 video coder, ICE-Ups, DTLS, RTP multiplexing, secure RTP 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, Silent 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 (No Media Anchoring)
  - Hairpinning of local calls to avoid unnecessary media delays and bandwidth consumption
- Voice Quality Monitoring
  - RTCP-XR, Audio/Code Session Experience Manager (SEM)
- High Availability (Redundancy)
  - SBC high availability with two-box redundancy, active calls preserved
- 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 UA’s

#### Management
- OAM&P
  - Browser-based GUI, CLI, SNMP (INI configuration file, REST API, EMS)
- Multi Tenancy
  - Advanced multi-tenant SBC partitioning
- SIP Routing
  - Routing Methods: Request URL – IP address, FQDN, ENUM, advanced LDAP third-party routing control through REST API
  - Advanced Routing: 802 1p/q VLAN tagging, DiffServ, TOS
- Criteria
  - QoS, 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
- SPPrec
  - E1T standard SIP recording interface

#### Virtual Edition VE SBC Minimum Requirements
- Hypervisor
  - VMware vSphere ESXi 5.x, Linux KVM, Microsoft Hyper-V
- Virtual NICs
  - 2 (standalone) 3 (high availability)
- Memory
  - 2 GB
- Virtual CPUs
  - 1
- Disk Space
  - 10 GB

### ABOUT GENESYS

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