# The Ribbon Communications SBC SWe Lite™ Session Border Controller

**The Ribbon Communications Session Border Controller Software Edition Lite (SBC SWe Lite) is an SBC optimized for small and mid-sized business (SMB) virtualized networking environments. The SBC SWe Lite delivers Customer Premises Equipment (CPE) interworking, security and survivability for Unified Communications (UC) and SIP trunking. With support for popular virtualization environments such as Microsoft® Hyper-V®, VMware® vSphere® Hypervisor and Linux® KVM, the SBC SWe Lite is installed quickly and easily. The SBC SWe Lite is certified for Direct Routing with Microsoft Teams.**

The Ribbon SBC SWe Lite derives its feature set from a common code base with the Tolly® certified and Miercom® performance verified SBC 1000 and SBC 2000 products. As such, customers can expect the same advantages when transitioning to UC or SIP trunking — an SBC that protects your voice infrastructure from Denial of Service (DoS)/Distributed DOS (DDOS) attacks, maintains privacy, encrypts your calls, and interworks with a wide variety of third-party SIP and legacy voice infrastructure devices/services, all while providing reliable, scalable performance that ensures maximum uptime and service availability.

By offering the same management and provisioning interface as the Ribbon SBC 1000/SBC 2000, the SBC SWe Lite dramatically speeds configuration through the simple and highly intuitive configuration wizard. In addition, the common code base affords the SBC SWe Lite a benefit unmatched by many competitors: an extremely compact CPU, RAM and program data store footprint uniquely positioned for highly constrained uCPE (universal CPE) processing environments. The result is clear: customers significantly reduce the costs and complexities of deploying uCPE devices in support of UC/SIP trunking.

## Single Instance Capabilities, Minimum Configuration (1 vCPU, 1 GB RAM Virtual Machine)

- Maximum number of SIP ↔ SIP sessions: 100
- Maximum number of RTP ↔ RTP sessions (direct media or RTP proxy mode): 100
- Maximum number of transcode sessions (G.711 ↔ G.729): 50
- Call Setup Maximum call setup rate: 10 cps
- Registrations Maximum number of registered users: 1,000
- Encryption
  - Maximum number of TLS-encrypted SIP sessions: 100
  - Maximum number of RTP ↔ SRTP sessions: 100
  - Maximum number of SRTP ↔ SRTP sessions (direct media or RTP proxy mode): 100

## Single Instance Capabilities, Maximum Configuration (2 vCPUs, 1.5 GB RAM Virtual Machine)

- Maximum number of SIP ↔ SIP sessions: 1000
- Maximum number of RTP ↔ RTP sessions (direct media or RTP proxy mode): 1000
- Maximum number of transcode sessions (G.711 ↔ G.729): 100
- Maximum call setup rate: 10 cps
- Registrations
  - Maximum number of registered users: 1,000
- Encryption
  - Maximum number of TLS-encrypted SIP sessions: 1000
  - Maximum number of RTP ↔ SRTP sessions: 1000
  - Maximum number of SRTP ↔ SRTP sessions (direct media or RTP proxy mode): 1000

## Single Instance Capabilities, Maximum Configuration (4 vCPUs, 2.5 GB RAM Virtual Machine)

- Maximum number of SIP ↔ SIP sessions: 1000
- Maximum number of RTP ↔ RTP sessions (direct media or RTP proxy mode): 1000
- Maximum number of transcode sessions (G.711 ↔ G.729): 300
- Maximum call setup rate: 10 cps
- Registrations
  - Maximum number of registered users: 5,000
- Encryption
  - Maximum number of TLS-encrypted SIP sessions: 1000
  - Maximum number of RTP ↔ SRTP sessions: 1000
  - Maximum number of SRTP ↔ SRTP sessions (direct media or RTP proxy mode): 1000

## Business Continuity

- Site survivability for SIP clients (including Polycom® VVX® phones) through built-in SIP registrar
- BroadSoft® BroadWorks® local survivability
- Multiple SIP trunking service provider support for redundancy
- ITSP E911 Support
- 911 Call Preemption

## Management Capabilities

**Operations, Administration and Management**

- Single, secure, web-based GUI with real-time monitoring
- 3 step Configuration Wizard, for quick provisioning between:
  - SIP trunks ↔ SIP phones, ISDN-based PBXs, and SIP based PBXs such as the Avaya® Aura® Communication Manager and the Cisco® Unified Communications Manager
  - Microsoft Skype for Business ↔ SIP trunks, ISDN trunks, or FXO ports
- REST-based programmatic interface to remotely manage multiple SBC SWe Lites
- SNMP for comprehensive network management using third-party management systems
- Configuration backup and restore; configuration upload from one site to another; partial configuration import/ export through REST
- CDR reporting
- Syslog for troubleshooting, with support for complimentary Ribbon LX syslog server and log parser tool
- Historical stats and TCAs

**Authentication**

- Local user (User name/password)
- Active Directory®
- RADIUS

## Media Services

- Supported codecs (including for transcode operations): G.711, G.722, G.722.2 (AMR-WB), G.723.1 (5.3 kbps, 6.3 kbps), G.726 (32kbps), G.729A/B (8 kbps)
- DTMF/RFC4733; Inband DTMF; SIP INFO/RFC-2833
- Voice Activity Detection (VAD)
- G.168 Echo Cancellation with standard 128 ms tail length
- Comfort video noise generation and packet loss concealment
- Automatic call type detection — voice, fax or modem
- Music on hold
- Call progress tones — ring back, busy, re-order
- RTP inactivity monitoring (dead call detection)
- RTP pass-through and media bypass
- RTCP
- Multiple media streams per session
- Caller ID support
- Video

## Signaling

- Maximum number of signaling groups: 100
- Back-to-Back User Agent (B2BUA)
- SIP (UDP/TCP/TLS) ↔ SIP (UDP/TCP/TLS)
- SIP Message Manipulation (SMM)

## Protocol Support

- SNMPv2c, SNMPv3
- NTP
- HTTPS
- SIP (RFC 3261) over UDP, TCP, TLS
- RTP/RTCP (RFC 3550, 3551)
- DNS
- RIPv2, OSPF as dynamic IP routing protocols
- DHCP client
- Asynchronous DNS for SIP
- NAT
- IPv4, IPv6, and IPv4/IPv6 interworking
- Support for Reason Header interworking

## Routing/Policy

- Maximum number of call route entries: 1000
- Active Directory/LDAP-based call routing
- Routing based on quality metrics
- Least-cost routing
  - Event-based action set
- On-board call forking (up to eight end points)
- Supplementary services
  - Call hold
  - Call transfer (blind & assisted)
  - Call forward
- Embedded policy/routing engine
- Optional centralized policy/routing via Ribbon Centralized Policy Server (PSX Server) using SIP
- Screening, blocking, routing, presentation, call type filters
- Route prioritization
- Leading digit routing; international routing; URI-based routing
- Digit manipulation (name/number manipulation using regular expression and Active Directory lookup)
- SIP routing
  - Based on source and destination IP address
  - Fully Qualified Domain Name (FQDN)
- Detect proxy failure and route to alternate paths
- Re-route on failure based on based on cause code
- Lync E911 support; SIP/PIDF-LO pass-through

## Security

- TLS (Transaction Layer Security) for signaling encryption (RFC 5246)
- Built-in VoIP firewall
- Secure Real-time Transport Protocol (SRTP) & Control Protocol (SRTCP) for media and media control encryption
  - SDES (Session Description Protocol Security Descriptions) key negotiation (RFC 4568)
- Wildcard certificate support
- Topology hiding; User privacy
- Prevention of Denial-of-Service (DoS) and Distributed DoS (DDoS) attacks
- Dialed Number Identification Service (DNIS), Calling Line Identification (CLID), Call type pre-authentication
- Malformed packet protection
- Access Control Lists (ACLs)
- NAT/NAPT and port forwarding, NAT traversal
- Traffic separation (VLAN interface separation)

## Quality of Service (QoS)
- Bandwidth management
- Call Admission Control (CAC) (deny excessive calls based on static configuration for bandwidth management)
- P-time mediation for rate limiting
- Per-call statistics
- DiffServ/DSCP marking

## Packet Network Time Source
- Network Time Protocol (NTP) per RFC 1708

## Microsoft® Teams®
- Certified SBC for Direct Routing
- Supports multiple tenant Direct Routing deployments with Microsoft partners and/or PSTN carriers

## Microsoft Skype® for Business
- Certified SBC for Skype for Business deployments
- Lync Server 2013 and Lync Server 2010 qualified for SBC
- Qualified for Microsoft Office 365® Exchange Unified Messaging
- Non-Skype for Business/Lync SIP client user state reporting (e.g., presence, user busy, etc.) to Skype for Business/ Lync Server
- Microsoft SCOM support

## Virtual Machine System Requirements
**CPU**
- 1, 2 or 4 virtual CPUs (vCPU) depending upon desired capacity; processing recommended on a second generation Intel® Core™ or Intel® Xeon® processor

**Memory**
- 1, 1.5, or 2.5 GiB RAM depending upon desired capacity

**Hard Disk Drive (HDD)**
- 5 GiB

**Virtual Network Interface Cards (vNICs)**
- Minimum 2 vNICs in operation

## Supported Virtual Machine Environments
- Microsoft Hyper-V®
- VMware® vSphere® Hypervisor (ESXi) Version 5.5 or above
- Linux® KVM (Kernel-based Virtual Machine)

## About Ribbon Communications
Ribbon is a company with two decades of leadership in real-time communications. Built on world class technology and intellectual property, Ribbon delivers intelligent, secure, embedded real-time communications for today's world. The company transforms fixed, mobile and enterprise networks from legacy environments to secure IP and cloud-based architectures, enabling highly productive communications for consumers and businesses. With locations in 28 countries around the globe, Ribbon's innovative, market-leading portfolio empowers service providers and enterprises with rapid service creation in a fully virtualized environment. The company's Kandy Communications Platform as a Service (CPaaS) delivers a comprehensive set of advanced embedded communications capabilities that enables this transformation.

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