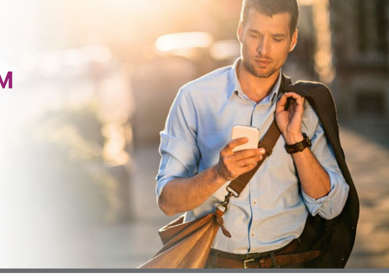
Ribbon Communications SBC SWe Lite™ Session Border Controller



The Ribbon Communications Session Border Controller Software Edition Lite (SBC SWe Lite) protects and secures small and mid-sized business (SMB) Unified Communications (UC) and SIP trunking services. With support for Microsoft® Hyper-V®, VMware® vSphere® Hypervisor and Linux® KVM, the SBC SWe Lite is deployed 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 SWe, 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. To this the SBC SWe Lite adds the media plane features of the SBC SWe, the SBC of choice deployed and trusted by the largest service providers and enterprises worldwide to ensure VoIP security and interoperability. The SBC SWe Lite also offers 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 SBC services in support of UC/SIP trunking.

1 vCPU, 1 GB RAM Virtual Machine Configuration
- Maximum SIP ↔ SIP sessions: 300
- Maximum RTP ↔ RTP sessions (direct media or RTP proxy/media anchoring mode): 300
- Maximum transcode sessions (G.711 ↔ G.729): 100
- Maximum call setup rate: 10 cps
- Maximum registered users: 1,000
- Encryption
  - Maximum TLS-encrypted SIP sessions: 300
  - Maximum RTP ↔ SRTP sessions: 300

2 vCPUs, 1.5 GB RAM Virtual Machine Configuration
- Maximum SIP ↔ SIP sessions: 1000
- Maximum RTP ↔ RTP sessions: 1000
- Maximum transcode sessions (G.711 ↔ G.729): 200
- Maximum call setup rate: 10 cps
- Maximum number of registered users: 1,000
- Encryption
  - Maximum TLS-encrypted SIP sessions: 1000
  - Maximum RTP ↔ SRTP sessions: 1000

4 vCPUs, 2.5 GB RAM Virtual Machine Configuration
- Maximum SIP ↔ SIP sessions: 1000
- Maximum RTP ↔ RTP sessions: 1000
- Maximum transcode sessions (G.711 ↔ G.729): 450 (600 if all SIP ↔ SIP sessions configured to run through the DSP)
- Maximum call setup rate: 10 cps
- Maximum registered users: 5,000
- Encryption
  - Maximum TLS-encrypted SIP sessions: 1000
  - Maximum RTP ↔ SRTP sessions: 1000

10 vCPUs, 2.5 GB RAM Virtual Machine Configuration
- Maximum SIP ↔ SIP sessions: 1200
- Maximum RTP ↔ RTP sessions: 1200
- Maximum transcode sessions (G.711 ↔ G.729): 1200
- Maximum call setup rate: 10 cps
- Maximum registered users: 5,000
- Encryption
  - Maximum TLS-encrypted SIP sessions: 1200
  - Maximum RTP ↔ SRTP sessions: 1200

Business Continuity
- Site survivability for SIP clients (including Yealink® Teams and Polycom® UC phones and conference bridges) through built-in SIP registrar
- BroadSoft® BroadWorks® local survivability
- Multiple SIP trunking service provider support for redundancy
- ITSP E911 Support
- 911 Call Preemption
- 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

Management Capabilities
Operations, Administration and Management
- HTTPS-based GUI with real-time monitoring
- 3 step Configuration Wizard, for quick provisioning between:
  - SIP trunks ↔ SIP phones, or SIP based PBXs such as Avaya® Aura® Communication Manager and Cisco® Unified Communications Manager
  - Microsoft Teams Direct Routing ↔ SIP trunks, SIP phones, or SIP-based PBX
  - Microsoft Skype for Business ↔ SIP trunks
- REST-based programmatic management interface
- SNMP v2c/v3 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
- Cloud-init for automatic provisioning from cloud-provided metadata services or custom configuration drive

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

Signaling
- SIP (RFC 3261) over UDP, TCP, TLS
- Maximum number of signaling groups: 100
- Back-to-Back User Agent (B2BUA)
- SIP (UDP/TCP/TLS) ↔ SIP (UDP/TCP/TLS)
- SIP Message Manipulation (SMM)

Media Services
- RTP/RTCP (RFC 3550, 3551)
- 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), OPUS, T.38
- DTMF/RFC4733; Inband DTMF; SIP INFO/RFC 2833
- Voice Activity Detection (VAD)
- G.168 Echo Cancellation with standard 128 ms tail length
- Comfort 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 (RTP proxy mode) and media bypass
- Multiple media streams per session
- Caller ID support
- Video

Other Protocol Support
- DNS
- RIPv2, OSPF dynamic routing
- DHCP client
- Asynchronous DNS for SIP
- IPv4, IPv6, and IPv4/IPv6 interworking
- Reason Header interworking

Routing/Policy
- Interactive Connectivity Establishment (ICE), RFC 8445
  - Full implementation support, including connectivity check generation
  - Lite support, for public Internet ICE agents
- Maximum number of call route entries: 1000
- Active Directory/LDAP-based call routing
- Routing based on quality metrics
- Least-cost routing
- On-board call forking, up to eight end points
- Supplementary services: call hold, call transfer (blind & assisted) and 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, international, URI-based routing
- Digit manipulation (name/number manipulation using regular expression and Active Directory lookup)
- SIP routing based on source, destination IP address or Fully Qualified Domain Name (FQDN)

Microsoft Partner
Gold Communications

Voice
Unified Communications
Business Productivity Solutions
Midmarket Solution Provider

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 Phone System Direct Routing
  - Non-Media Bypass
  - Media Bypass enhancements to improve user experience, including support for deployments behind a public router (option to configure SBC with private IP address)
- 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 SIP client user state reporting (e.g., presence, user busy, etc.) to Skype for Business Server

Virtual Machine System Requirements
CPU 1, 2, 4, or 10 virtual CPUs (vCPU) processing recommended on a second generation Intel® Core™ or Intel® Xeon® processor

Memory 1, 1.5, or 2.5 GiB RAM

Hard Disk Drive (HDD) 5 GiB

Virtual Network Interface Cards (vNIC)
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)

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