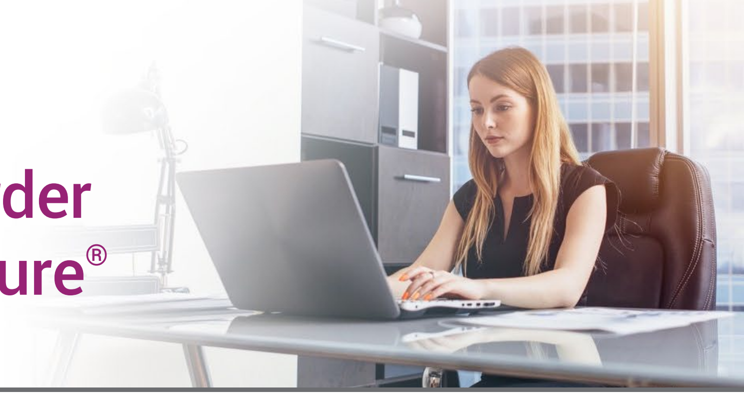
Ribbon Communications SBC SWe Lite™ Session Border Controller for Microsoft® Azure®



The Ribbon Communications Session Border Controller Software Edition Lite (SBC SWe Lite) is a full featured SBC that delivers security, interworking, and survivability to enterprises deploying cloud-based Unified Communications (UC) services such as Microsoft® Teams®. Available from the Azure® Marketplace and offering an easy to use GUI, an enterprise can spin up and deploy the SBC SWe Lite in a cost-efficient Azure virtual machine (VM) quickly and easily. Based on technology deployed with the largest service providers and enterprises worldwide, enterprises can trust the SBC SWe Lite to secure Voice over Internet Protocol (VoIP) from bad actors seeking to deny service, eavesdrop on communications, or make fraudulent toll calls.

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The Ribbon SBC SWe Lite derives its management, signaling and media plane feature set from a common code base with the Tolly® certified and Miercom® performance verified SBC SWe, SBC 1000 and SBC 2000 products. That heritage protects and interconnects your voice infrastructure as you transition UC and SIP trunking. SBC SWe Lite offers dependable protection from Denial of Service (DoS)/Distributed DoS (DDoS) in addition to encrypting calls and maintaining your privacy. It provides reliable, scalable performance that ensures maximum uptime and service availability while interworking with a wide variety of third-party SIP and legacy voice infrastructure.

By offering the same management and provisioning interface as the Ribbon SBC 1000/SBC 2000, the SBC SWe Lite dramatically speeds deployment through the simple and highly intuitive configuration wizard. 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 cost-optimized Azure VM processing environments. The result is clear: customers significantly reduce the costs and complexities of deploying Azure-based trusted SBC services in support of cloud-based UC/SIP trunking.

1 vCPU Azure VM Instance Capabilities

- 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, SILK): 100
- Maximum call setup rate: 10 cps
- Maximum number of registered users: 1,000
- Encryption:
  - Maximum TLS-encrypted SIP sessions: 300
  - Maximum RTP ↔ SRTP sessions: 300

4 vCPU Azure VM Instance Capabilities

- Maximum SIP ↔ SIP sessions: 1000
- Maximum RTP ↔ RTP sessions (direct media or RTP proxy/media anchoring mode): 1000
- Maximum transcode sessions (G.711 ↔ G.729, SILK): 400
- Maximum call setup rate: 10 cps
- Maximum number of registered users: 5,000
- Encryption:
  - Maximum TLS-encrypted SIP sessions: 1000
  - Maximum RTP ↔ SRTP sessions: 500

Business Continuity

- Site survivability for SIP clients (including Yealink® and Polycom® phones and conference bridges) through a 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
- Microsoft® Office 365® Phone System E911 support; SIP/PIDF-LO pass through and ELIN Gateway

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 Avaya® Aura® Communication Manager and Cisco® Unified Communications Manager
  - Microsoft Teams Direct Routing ↔ SIP trunks, ISDN trunks, or SIP-based PBXs
  - Microsoft Skype for Business ↔ SIP trunks, ISDN trunks, or FXO ports
- REST-based programmatic management option
- SNMP v2c/v3 for comprehensive network management
- 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

Authentication

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

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)
- SIP (RFC 3261) over UDP, TCP, TLS

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), OPUS, SILK-NB/WB, 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 pass through
- RTP/RTCP (RFC 3550, 3551)
- RTP/RTCP multiplexing over single UDP port (RFC 5761)

Other Protocol Support

- DNS
- RIPv2, OSPF dynamic routing
- DHCP client
- Asynchronous DNS for SIP
- IPv4, IPv6, and IPv4/IPv6 interworking
- Support for 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
- Azure® and on-premises Active Directory/LDAP-based call routing
- Routing based on quality metrics
- Least-cost routing
- Time of Day routing
- 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, international, and URI-based routing
- Digit manipulation (name/number manipulation using regular expression and Active Directory lookup)
- SIP routing, based on source and destination IP address and fully Qualified Domain Name (FQDN)

Security

- TLS (Transaction Layer Security) for signaling encryption
  - TLS 1.2 (RFC 5246)
  - DTLS version 1.2 (RFC 6347)
- 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)
  - DTLS extension for SRTP/SRTCP (RFC 5764)
- Multiple unique X.509 public key certificates/PKCS #12 files (up to 11)
- 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

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Microsoft® Teams®

- Certified SBC for Phone System Direct Routing
  - Enhanced 911 (E911), ELIN support
  - SILK-NB, SILK-WB codec support for improved Teams user experience
  - Media Bypass enhancements to improve user experience, including support for deployments behind a public router (option to configure SBC with private IP address)
  - Multiple encryption certificates to ease migration from on-premises Skype for Business Server to Office 365 Phone System
  - Video media stream forwarding
- Supports multiple tenant Direct Routing deployments with Microsoft partners and/or PSTN carriers

Microsoft Skype® for Business

- Certified SBC for Skype for Business deployments
- 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
- Microsoft SCOM support

Recommended Azure Virtual Machines

Recommendations assume 1 active session per 10 employees.

- Small/Medium Business, less than 100 employees: B1MS
- Medium Business, less than 300 employees: B2S
- Medium Business, less than 1000 employees: DS1 v2
- Enterprise, less than 4000 employees: DS3 v2

Please consult azure.microsoft.com for information on VM availability and pricing.

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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