

## Session Border Controllers (SBCs)

- 5~30 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)



With versatile and robust architecture, The Synway SBC30 Session Border Controller (SBC) offers a complete connectivity solution for SMB enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

Scaling up to 30 concurrent sessions, the SBC30 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC30 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

### 5~30 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK



#### High interoperability

Adopted by over 500 SPs and enterprises, and proven interoperability with SIP trunks, SIP platforms and IP cloud services



#### Enhanced security

Security-oriented, robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft



#### Superior voice quality

Integrate decades of SW/HW technologies to obtain advanced capabilities for optimizing and monitoring voice service quality



#### High resiliency

Telco-grade reliability, with High Availability (HA) using 1+1 active/standby redundancy, local branch survivability and PSTN fallback



#### Flexible scalability

The SBC30 architecture can scale up from 5 to 30 sessions, and the various licensing options assure economical scalability

### Basic Features and Functions For SBC

- Dos/DDos protection
- QOS/ TOS/DSCP setting
- Signal encryption(TLS/IPSec)
- Media encryption (SRTP)
- NAT transverse
- SIP interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup



Capacities			
<b>Max Signaling</b>	30(from 5 to 30)	<b>Max. Transcoding Sessions</b>	30(from 5 to 30)
<b>Max. RTP/SRTP Sessions</b>	30(from 5 to 30)	<b>Max. Registered Users</b>	250(upgradeable to 500)
Network Interfaces			
<b>Ethernet:</b>	2(10/100 BASE-TX(RJ-45)) & Customizable		
Security			
<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)		
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication		
<b>Privacy:</b>	Topology hiding, user privacy		
<b>Traffic Separation:</b>	Self-adjustable automatic load balance		
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access		
<b>VoIP firewall:</b>	Optional		
Interoperability			
<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode		
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold		
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users		
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP		
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions		
<b>Number Manipulations:</b>	Ingress and egress digit manipulation		
<b>Transcoding and Vocoders:</b>	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB		
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion		
<b>NAT:</b>	Hosted NAT, RTP self-adaption		
<b>WebRTC controller:</b>	Optional or customizable		
Voice Quality and SLA			
<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions		
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ		
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).		
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation		
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection		
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption		
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved		
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs		
<b>Echo cancellation:</b>	G.168 128 ms tail length		
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking		
SIP Routing			
<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth		
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI		
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization		
<b>SIPREC:</b>	SynAPI recording interface		
Management			
<b>OAM&amp;P:</b>	Browser-based GUI, SNMP, INI Configuration file		
Physical/Environmental			
<b>Dimensions:</b>	190*30*120mm		
<b>Weight:</b>	About 0.7Kg		
<b>Mounting:</b>	Desktop		
<b>Power:</b>	100-240V AC		
<b>Environmental:</b>	Operating temperature: 0°C —40°C ;Storage temperature: -20°C —85°C Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing		