

Session Border Controllers (SBCs)

- 30~60 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 SBC60 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 60 concurrent sessions, the SBC60 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC60 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.

30~60 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 SBC60 architecture can scale up from 30 to 60 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			
Max Signaling	60(from 30 to 60)	Max. Transcoding Sessions	120(from 30 to 60)
Max. RTP/SRTP Sessions	120(from 30 to 60)	Max. Registered Users	500(upgradeable to 1000)
Network Interfaces			
Ethernet:	2(10/100 BASE-TX(RJ-45)) & Customizable		
Security			
Access Control:	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)		
Encryption/Authentication:	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication		
Privacy:	Topology hiding, user privacy		
Traffic Separation:	Self-adjustable automatic load balance		
Intrusion Detection System:	Detection and prevention of VoIP attacks, theft of service and unauthorized access		
VoIP firewall:	Optional		
Interoperability			
SIP B2BUA:	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode		
SIP Interworking:	3xx redirect, REFER, PRACK, early media, call hold		
Registration and Authentication:	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users		
Transport Mediation:	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP		
Header Manipulation:	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions		
Number 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.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB		
Signal Conversion:	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion		
NAT:	Hosted NAT, RTP self-adaption		
WebRTC controller:	Optional or customizable		
Voice Quality and SLA			
Call Admission Control:	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions		
Packet Marking:	802.1p/Q VLAN tagging, DiffServ		
Standalone Survivability:	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).		
Impairment Mitigation:	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation		
Voice Monitoring and Enhancement:	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection		
Direct Media:	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption		
High Availability:	SBC high availability with 1+1 redundancy, active calls preserved		
Test Agent:	Ability to remotely verify SIP message flow between SIP UAs		
Echo cancellation:	G.168 128 ms tail length		
Advanced Media Processing:	T.38 real-time fax, T.38 – G.711 interworking		
SIP Routing			
Routing Criteria:	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth		
Route To:	Configured SIP peers, registered users, IP address, request URI		
Advanced Routing Features:	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization		
SIPREC:	SynAPI recording interface		
Management			
OAM&P:	Browser-based GUI, SNMP, INI Configuration file		
Physical/Environmental			
Dimensions:	190*30*120mm		
Weight:	About 0.7Kg		
Mounting:	Desktop		
Power:	100-240V AC		
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing		