

Hybrid SBC and Media Gateway

- 250~ 500 Hybrid IP SBC Sessions & TDM Survivability
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)



With versatile and robust architecture, The Synway SBC500H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for large enterprises and service provider.

Scaling up to 500 concurrent sessions, the SBC500H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC500H could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN networks.

250~500 SBC Sessions | 1+1 High Availability | High Survivability | 30+ TDM Sessions



High interoperability

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



Hybrid functionality

Fit to complex networks, a sophisticated combo SBC and gateway architecture for gradual migration, low CAPEX and reduced space and power footprints



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

Basic Features and Functions For SBC

- Dos/DDos protection
- QOS/ TOS/DSCP setting
- Signal encryption(TLS/IPSec)
- Media encryption (SRTP)
- NAT transverse
- SIP/H.323/H.248 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup



Capacities			
Max Signaling	500(from 250 to 500)	Max. Transcoding Sessions	500(from 250 to 500)
Max. RTP/SRTP Sessions	500(from 250 to 500)	Max. Registered Users	4000(upgradeable to 8000)
Telephony Interfaces			
Analog	Optional		
Digital	Up to 8E1/T1 Interfaces		
Clock Source	50 ppm High Precision		
Digital PSTN Protocols:	ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy		
Network Interfaces			
Ethernet:	2(10/100/1000 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:	44*440*267mm		
Weight:	About 3.1Kg		
Mounting:	19" rack mount		
Power:	100-240V AC redundant dual feed		

