SBC60H

Hybrid SBC and Media Gateway

- 30~60 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 SBC60H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for SMB enterprises and service provider.

Scaling up to 60 concurrent sessions, the SBC60H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC60H 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.

30~60 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

Synway®

SBC60H

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)

Telephony Interfaces	
Analog	Optional
Digital	Up to 2E1/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:	190*30*120mm
Weight:	About 0.7Kg
Mounting:	19" rack mount
Power:	100-240V AC redundant dual feed

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