SBC250

Session Border Controllers (SBCs)

- 120~250 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 SBC250 Session Border Controller (SBC) offers a complete connectivity solution for large enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

Scaling up to 250 concurrent sessions, the SBC250 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC250 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.

120~250 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 SBC250 architecture can scale up from 120 to 250 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

Synway®

SBC250

Capacities			
Max Signaling	250(from 120 to 250)	**Max. Transcoding Sessions**	250(from 120 to 250)
Max. RTP/SRTP Sessions	250(from 120 to 250)	**Max. Registered Users**	2000(upgradeable to 4000)

Telephony Interfaces	
Analog	Optional
Digital	Up to 4E1/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

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