Get Connected with

Index For Synway's SBC Series Session Border Controllers

SBC30

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

SBC30

Capacities			
Max Signaling	30(from 5 to 30)	**Max. Transcoding Sessions**	30(from 5 to 30)
Max. RTP/SRTP Sessions	30(from 5 to 30)	**Max. Registered Users**	250(upgradeable to 500)

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:	19" rack mount or Desktop
Power:	100-240V AC redundant dual feed
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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SBC60

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

Synway®

SBC60

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:	19" rack mount or Desktop
Power:	100-240V AC redundant dual feed
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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SBC120

Session Border Controllers (SBCs)

- 60~120 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 SBC120 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 120 concurrent sessions, the SBC120 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC120 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.

60~120 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 SBC120 architecture can scale up from 60 to 120 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

SBC120

Capacities			
Max Signaling	120(from 60 to 120)	**Max. Transcoding Sessions**	120(from 60 to 120)
Max. RTP/SRTP Sessions	120(from 60 to 120)	**Max. Registered Users**	1000(upgradeable to 2000)

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
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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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

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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SBC500

Session Border Controllers (SBCs)

- 250~500 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 SBC500 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 500 concurrent sessions, the SBC500 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC500 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.

250~500 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 SBC500 architecture can scale up from 250 to 500 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

SBC500

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)

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
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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SBC1000

Session Border Controllers (SBCs)

- 500~1,000 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 SBC1000 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 1,000 concurrent sessions, the SBC1000 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC1000 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.

500~ 1,000 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 SBC1000 architecture can scale up from 500 to 1000 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®

SBC1000

Capacities			
Max Signaling	1000(from 500 to 1000)	**Max. Transcoding Sessions**	1000(from 500 to 1000)
Max. RTP/SRTP Sessions	1000(from 500 to 1000)	**Max. Registered Users**	8000(upgradeable to 16000)

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*690mm
Weight:	About 12Kg
Mounting:	19" rack mount
Power:	100-240V AC redundant dual feed
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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SBC2000

Session Border Controllers (SBCs)

- 1000~2,000 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 SBC2000 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 2,000 concurrent sessions, the SBC2000 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC2000 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.

1000~2,000 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 SBC2000 architecture can scale up from 1000 to 2000 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®

SBC2000

Capacities			
Max Signaling	2000(from 1000 to 2000)	**Max. Transcoding Sessions**	2000(from 1000 to 2000)
Max. RTP/SRTP Sessions	2000(from 1000 to 2000)	**Max. Registered Users**	8000(upgradeable to 16000)

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*690mm
Weight:	About 12Kg
Mounting:	19” rack mount
Power:	100-240V AC redundant dual feed
Environmental:	Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃ Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

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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 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 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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SBC120H

Hybrid SBC and Media Gateway

- 60~120 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 SBC120H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for SMB enterprises and service providers.

Scaling up to 120 concurrent sessions, the SBC120H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC120H 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.

60~120 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 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup

SBC120H

Capacities			
Max Signaling	120(from 60 to 120)	**Max. Transcoding Sessions**	120(from 60 to 120)
Max. RTP/SRTP Sessions	120(from 60 to 120)	**Max. Registered Users**	1000(upgradeable to 2000)

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

Synway ®

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SBC250H

Hybrid SBC and Media Gateway

- 120~250 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 SBC250H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for large enterprises and service provider.

Scaling up to 250 concurrent sessions, the SBC250H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC250H 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 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup

SBC250H

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 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

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
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SBC500H

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 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup

SBC500H

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

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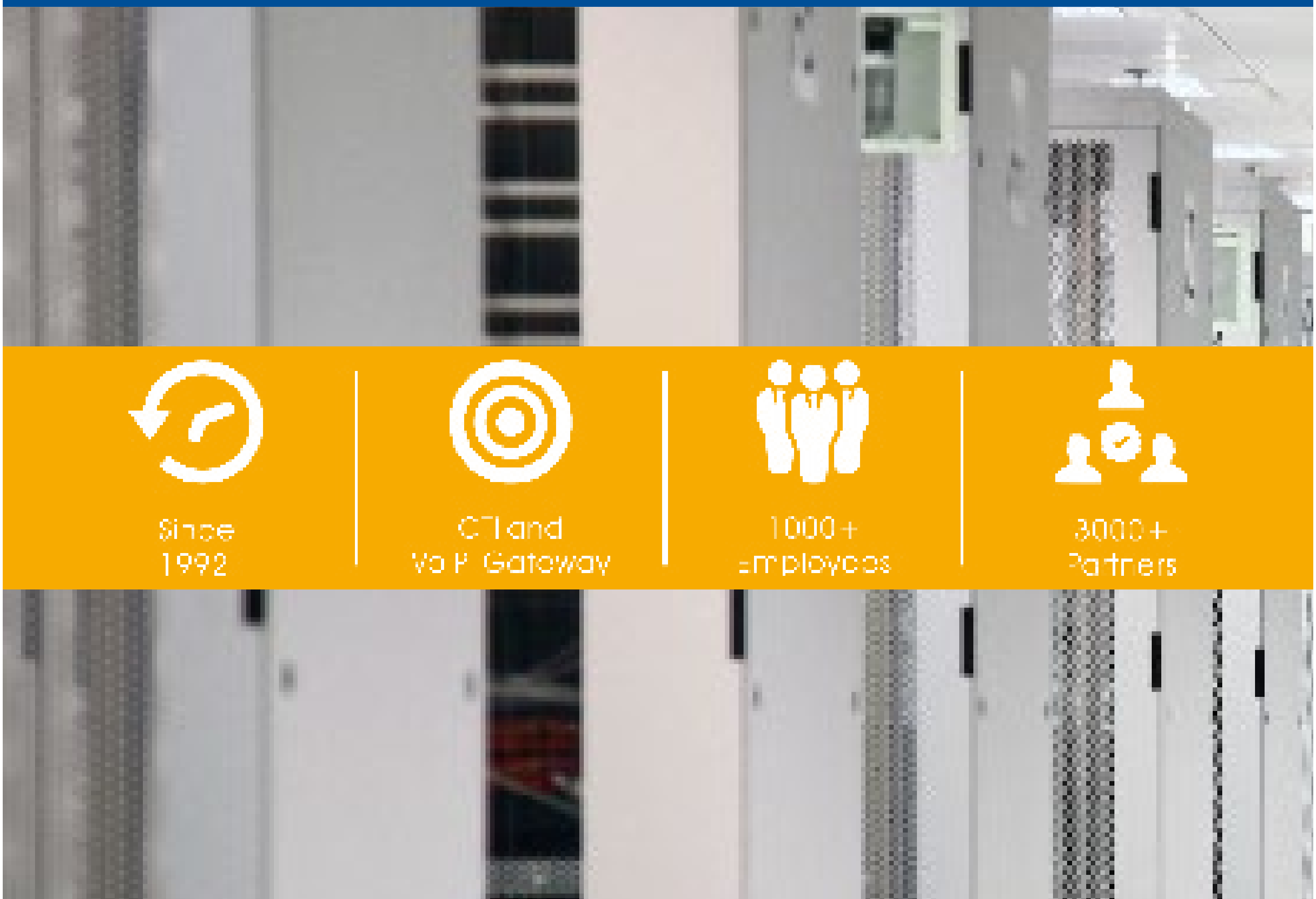
Synway®



As a leading VoIP enabling-technologies provider in China, Synway has been partnered with applications & solution providers worldwide to deliver turkey solutions for enterprises and telecom carriers. Based on long-standing business network, Synway's appliances and equipments, with third-party compliant software platforms from mainstream application providers, have served 5,000 plus customers, including contact centers, financial institutes, public security, government agencies, service providers, hospitality and operators.

In ever-changeable environments, Synway's long-term goal would be of partnership with vendors of cloud-based unified communications, providing enterprises and SPs with a complete range of cloud-based applications, including Video and Audio Conferencing, Contact Center, IP-PBX, Unified Messaging, Social Media Services and more. For in-house IPR and better customer value, Synway provides strategic partners with customized OEM or ODM services to localize more efficiently. To achieve 0-defective rate, Synway has adopted ISO9001, CE, FCC, RoHS, 3C and more since 2001.

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