



## Index For Synway's SBC Series Session Border Controllers

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## 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/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	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(10/100/1000 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, QoS, 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: 0C —40C ;Storage temperature: -20C —85C Humidity: 8%—90% non-condensing;Storage humidity: 8%—90% non-condensing		

## 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/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	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(10/100/1000 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 or Desktop		
Power:	100-240V AC redundant dual feed		
Environmental:	Operating temperature: 0°C —40°C ;Storage temperature: -20°C —85°C Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing		

### 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/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	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, QoS, 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°C —40°C ;Storage temperature: -20°C —85°C Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing		

## 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/H.323/H.248 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup





### Capacities

<b>Max Signaling</b>	250(from 120 to 250)	<b>Max. Transcoding Sessions</b>	250(from 120 to 250)
<b>Max. RTP/SRTP Sessions</b>	250(from 120 to 250)	<b>Max. Registered Users</b>	2000(upgradeable to 4000)

### Telephony Interfaces

<b>Analog</b>	Optional
<b>Digital</b>	Up to 4E1/T1 Interfaces
<b>Clock Source</b>	50 ppm High Precision
<b>Digital PSTN Protocols:</b>	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

<b>Ethernet:</b>	2(10/100/1000 BASE-TX(RJ-45)) & Customizable
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### Security

<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication
<b>Privacy:</b>	Topology hiding, user privacy
<b>Traffic Separation:</b>	Self-adjustable automatic load balance
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access
<b>VoIP firewall:</b>	Optional

### Interoperability

<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
<b>Number Manipulations:</b>	Ingress and egress digit manipulation
<b>Transcoding and Vocoders:</b>	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
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
<b>NAT:</b>	Hosted NAT, RTP self-adaption
<b>WebRTC controller:</b>	Optional or customizable

### Voice Quality and SLA

<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs
<b>Echo cancellation:</b>	G.168 128 ms tail length
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking

### SIP Routing

<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
<b>SIPREC:</b>	SynAPI recording interface

### Management

<b>OAM&amp;P:</b>	Browser-based GUI, SNMP, INI Configuration file
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### Physical/Environmental

<b>Dimensions:</b>	44*440*267mm
<b>Weight:</b>	About 3.1Kg
<b>Mounting:</b>	19" rack mount
<b>Power:</b>	100-240V AC redundant dual feed



## 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/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)
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: 0C —40C ;Storage temperature: -20C —85C Humidity: 8%—90% non-condensing;Storage humidity: 8%—90% non-condensing		

### 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/H.323/H.248 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup



## Capacities

<b>Max Signaling</b>	1000(from 500 to 1000)	<b>Max. Transcoding Sessions</b>	1000(from 500 to 1000)
<b>Max. RTP/SRTP Sessions</b>	1000(from 500 to 1000)	<b>Max. Registered Users</b>	8000(upgradeable to 16000)

## Network Interfaces

**Ethernet:** 2(10/100/1000 BASE-TX(RJ-45)) & Customizable

## Security

<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication
<b>Privacy:</b>	Topology hiding, user privacy
<b>Traffic Separation:</b>	Self-adjustable automatic load balance
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access
<b>VoIP firewall:</b>	Optional

## Interoperability

<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
<b>Number Manipulations:</b>	Ingress and egress digit manipulation
<b>Transcoding and Vocoders:</b>	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
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
<b>NAT:</b>	Hosted NAT, RTP self-adaption
<b>WebRTC controller:</b>	Optional or customizable

## Voice Quality and SLA

<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs
<b>Echo cancellation:</b>	G.168 128 ms tail length
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking

## SIP Routing

<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoS, bandwidth
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
<b>SIPREC:</b>	SynAPI recording interface

## Management

**OAM&P:** Browser-based GUI, SNMP, INI Configuration file

## Physical/Environmental

<b>Dimensions:</b>	44*440*690mm
<b>Weight:</b>	About 12Kg
<b>Mounting:</b>	19" rack mount
<b>Power:</b>	100-240V AC redundant dual feed
<b>Environmental:</b>	Operating temperature: 0C — 40C ;Storage temperature: -20C — 85C Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing

### 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/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	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: 0C — 40C ;Storage temperature: -20C — 85C Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing		

### 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





### Capacities

<b>Max Signaling</b>	60(from 30 to 60)	<b>Max. Transcoding Sessions</b>	120(from 30 to 60)
<b>Max. RTP/SRTP Sessions</b>	120(from 30 to 60)	<b>Max. Registered Users</b>	500(upgradeable to 1000)

### Telephony Interfaces

<b>Analog</b>	Optional
<b>Digital</b>	Up to 2E1/T1 Interfaces
<b>Clock Source</b>	50 ppm High Precision
<b>Digital PSTN Protocols:</b>	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

<b>Ethernet:</b>	2(10/100/1000 BASE-TX(RJ-45)) & Customizable
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### Security

<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication
<b>Privacy:</b>	Topology hiding, user privacy
<b>Traffic Separation:</b>	Self-adjustable automatic load balance
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access
<b>VoIP firewall:</b>	Optional

### Interoperability

<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
<b>Number Manipulations:</b>	Ingress and egress digit manipulation
<b>Transcoding and Vocoders:</b>	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
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
<b>NAT:</b>	Hosted NAT, RTP self-adaption
<b>WebRTC controller:</b>	Optional or customizable

### Voice Quality and SLA

<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs
<b>Echo cancellation:</b>	G.168 128 ms tail length
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking

### SIP Routing

<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
<b>SIPREC:</b>	SynAPI recording interface

### Management

<b>OAM&amp;P:</b>	Browser-based GUI, SNMP, INI Configuration file
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### Physical/Environmental

<b>Dimensions:</b>	190*30*120mm
<b>Weight:</b>	About 0.7Kg
<b>Mounting:</b>	19" rack mount
<b>Power:</b>	100-240V AC redundant dual feed



### 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/H.323/H.248 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup



### Capacities

<b>Max Signaling</b>	120(from 60 to 120)	<b>Max. Transcoding Sessions</b>	120(from 60 to 120)
<b>Max. RTP/SRTP Sessions</b>	120(from 60 to 120)	<b>Max. Registered Users</b>	1000(upgradeable to 2000)

### Telephony Interfaces

<b>Analog</b>	Optional
<b>Digital</b>	Up to 4E1/T1 Interfaces
<b>Clock Source</b>	50 ppm High Precision
<b>Digital PSTN Protocols:</b>	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

<b>Ethernet:</b>	2(10/100/1000 BASE-TX(RJ-45)) & Customizable
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### Security

<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication
<b>Privacy:</b>	Topology hiding, user privacy
<b>Traffic Separation:</b>	Self-adjustable automatic load balance
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access
<b>VoIP firewall:</b>	Optional

### Interoperability

<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
<b>Number Manipulations:</b>	Ingress and egress digit manipulation
<b>Transcoding and Vocoders:</b>	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
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
<b>NAT:</b>	Hosted NAT, RTP self-adaption
<b>WebRTC controller:</b>	Optional or customizable

### Voice Quality and SLA

<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs
<b>Echo cancellation:</b>	G.168 128 ms tail length
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking

### SIP Routing

<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
<b>SIPREC:</b>	SynAPI recording interface

### Management

<b>OAM&amp;P:</b>	Browser-based GUI, SNMP, INI Configuration file
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### Physical/Environmental

<b>Dimensions:</b>	44*440*267mm
<b>Weight:</b>	About 3.1Kg
<b>Mounting:</b>	19" rack mount
<b>Power:</b>	100-240V AC redundant dual feed



### 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/H.323/H.248 interworking
- Support IPV4 , IPV6 and VPN
- Load balancing
- Transmission speed limit
- RTP encoding/decoding
- Anti-phreaking
- Redundancy and Backup



### Capacities

<b>Max Signaling</b>	250(from 120 to 250)	<b>Max. Transcoding Sessions</b>	250(from 120 to 250)
<b>Max. RTP/SRTP Sessions</b>	250(from 120 to 250)	<b>Max. Registered Users</b>	2000(upgradeable to 4000)

### Telephony Interfaces

<b>Analog</b>	Optional
<b>Digital</b>	Up to 8E1/T1 Interfaces
<b>Clock Source</b>	50 ppm High Precision
<b>Digital PSTN Protocols:</b>	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

<b>Ethernet:</b>	2(10/100/1000 BASE-TX(RJ-45)) & Customizable
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### Security

<b>Access Control:</b>	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
<b>Encryption/Authentication:</b>	TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication
<b>Privacy:</b>	Topology hiding, user privacy
<b>Traffic Separation:</b>	Self-adjustable automatic load balance
<b>Intrusion Detection System:</b>	Detection and prevention of VoIP attacks, theft of service and unauthorized access
<b>VoIP firewall:</b>	Optional

### Interoperability

<b>SIP B2BUA:</b>	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
<b>SIP Interworking:</b>	3xx redirect, REFER, PRACK, early media, call hold
<b>Registration and Authentication:</b>	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
<b>Transport Mediation:</b>	Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP
<b>Header Manipulation:</b>	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
<b>Number Manipulations:</b>	Ingress and egress digit manipulation
<b>Transcoding and Vocoders:</b>	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
<b>Signal Conversion:</b>	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion
<b>NAT:</b>	Hosted NAT, RTP self-adaption
<b>WebRTC controller:</b>	Optional or customizable

### Voice Quality and SLA

<b>Call Admission Control:</b>	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
<b>Packet Marking:</b>	802.1p/Q VLAN tagging, DiffServ
<b>Standalone Survivability:</b>	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911).
<b>Impairment Mitigation:</b>	Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation
<b>Voice Monitoring and Enhancement:</b>	acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection
<b>Direct Media:</b>	Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
<b>High Availability:</b>	SBC high availability with 1+1 redundancy, active calls preserved
<b>Test Agent:</b>	Ability to remotely verify SIP message flow between SIP UAs
<b>Echo cancellation:</b>	G.168 128 ms tail length
<b>Advanced Media Processing:</b>	T.38 real-time fax, T.38 – G.711 interworking

### SIP Routing

<b>Routing Criteria:</b>	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
<b>Route To:</b>	Configured SIP peers, registered users, IP address, request URI
<b>Advanced Routing Features:</b>	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
<b>SIPREC:</b>	SynAPI recording interface

### Management

<b>OAM&amp;P:</b>	Browser-based GUI, SNMP, INI Configuration file
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### Physical/Environmental

<b>Dimensions:</b>	44*440*267mm
<b>Weight:</b>	About 3.1Kg
<b>Mounting:</b>	19" rack mount
<b>Power:</b>	100-240V AC redundant dual feed



### 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

<b>Max Signaling</b>	500(from 250 to 500)	<b>Max. Transcoding Sessions</b>	500(from 250 to 500)
<b>Max. RTP/SRTP Sessions</b>	500(from 250 to 500)	<b>Max. Registered Users</b>	4000(upgradeable to 8000)

### Telephony Interfaces

<b>Analog</b>	Optional
<b>Digital</b>	Up to 8E1/T1 Interfaces
<b>Clock Source</b>	50 ppm High Precision
<b>Digital PSTN Protocols:</b>	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

<b>Ethernet:</b>	2(10/100/1000 BASE-TX(RJ-45)) & Customizable
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Since  
1992



CTI and  
Vo P Gateway



1000+  
employees



3000+  
Partners

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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