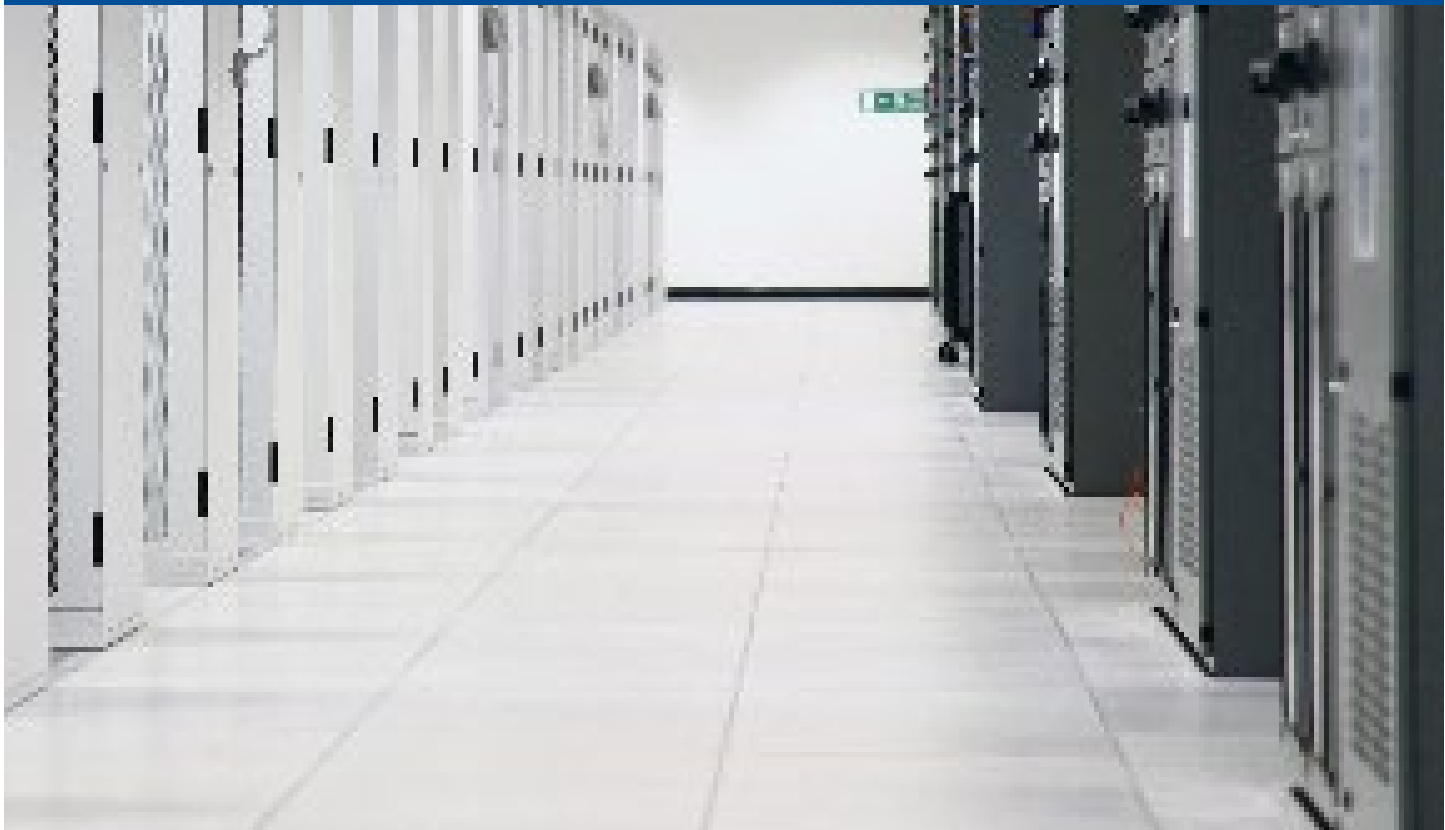
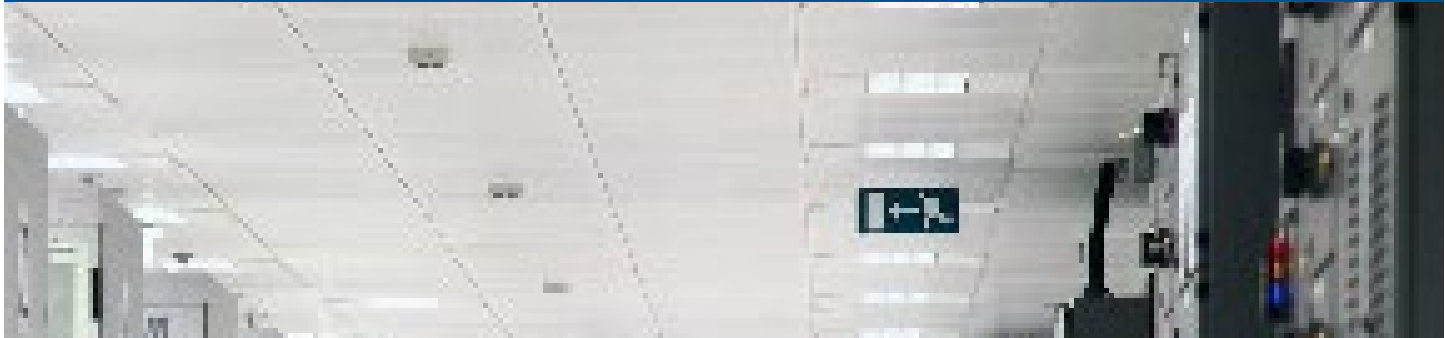


Get Connected with



Session Border Controller

Next-Generation Session Border Controller for SMB,Enterprises,SPs

Synway Information Engineering Co., Ltd.

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
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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
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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		

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

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

Telephony Interfaces

Analog	Optional
Digital	Up to 2E1/T1 Interfaces
Clock Source	50 ppm High Precision
Digital PSTN Protocols:	ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy

Network Interfaces

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

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



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		

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

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
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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
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Physical/Environmental

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



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

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
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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
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Physical/Environmental

Dimensions:	44*440*267mm
Weight:	About 3.1Kg
Mounting:	19" rack mount
Power:	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

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
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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
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Physical/Environmental

Dimensions:	44*440*267mm
Weight:	About 3.1Kg
Mounting:	19" rack mount
Power:	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		

Hybrid SBC and Media Gateway

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



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

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

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

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



High interoperability

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



Hybrid functionality

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



Enhanced security

Security-oriented, robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft



Superior voice quality

Integrate decades of SW/HW technologies to obtain advanced capabilities for optimizing and monitoring voice service quality



High resiliency

Telco-grade reliability, with High Availability (HA) using 1+1 active/standby redundancy, local branch survivability and PSTN fallback

Basic Features and Functions For SBC

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



Capacities

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

Telephony Interfaces

Analog	Optional
Digital	Up to 8E1/T1 Interfaces
Clock Source	50 ppm High Precision
Digital PSTN Protocols:	ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy

Network Interfaces

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

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



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			
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, 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*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

- 500~1,000 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 SBC1000H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for large enterprises and service provider.

Scaling up to 1,000 concurrent sessions, the SBC1000H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC1000H 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.

500~ 1,000 SBC Sessions | 1+1 High Availability | High Survivability | 30+ TDM Sessions



High interoperability

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



Hybrid functionality

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



Enhanced security

Security-oriented, robust perimeter defense against cyber, DoS and DDoS attacks, as well as eavesdropping, fraud and service theft



Superior voice quality

Integrate decades of SW/HW technologies to obtain advanced capabilities for optimizing and monitoring voice service quality



High resiliency

Telco-grade reliability, with High Availability (HA) using 1+1 active/standby redundancy, local branch survivability and PSTN fallback

Basic Features and Functions For SBC

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



Capacities

Max Signaling	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)

Telephony Interfaces

Analog	Optional
Digital	Up to 63E1/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
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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
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Physical/Environmental

Dimensions:	44*440*690mm
Weight:	About 12Kg
Mounting:	19" rack mount
Power:	100-240V AC redundant dual feed



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		
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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

- Max. 2000 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 SBC2000H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for large enterprises and service provider.

Scaling up to 2,000 concurrent sessions, the SBC2000H connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP/TDM to SIP environment.

The SBC2000H 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.

2,000 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



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- QOS/ TOS/DSCP setting
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- 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	1000(upgradeable to 2000)	Max. Transcoding Sessions	1000(upgradeable to 2000)
Max. RTP/SRTP Sessions	1000(upgradeable to 2000)	Max. Registered Users	8000(upgradeable to 16000)

Telephony Interfaces

Analog	Optional
Digital	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
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Security

Access Control:	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
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