Get Connected with

# Index For Synway's SBC Series Session Border Controllers

Synway's Session Border Controllers

DATASHEET

SBC30

# Session Border Controllers (SBCs)

- 5~30 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)


With versatile and robust architecture, The Synway SBC30 Session Border Controller (SBC) offers a complete connectivity solution for SMB enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

Scaling up to 30 concurrent sessions, the SBC30 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC30 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**5~30 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC30 architecture can scale up from 5 to 30 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

Synway®

# SBC30

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

| Network Interfaces | |
|---|---|
| **Ethernet:** | 2(10/100/1000 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:** | Desktop |
| **Power:** | 100-240V AC |
| **Environmental:** | Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃<br>Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC60

# Session Border Controllers (SBCs)

- 30~60 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)


With versatile and robust architecture, The Synway SBC60 Session Border Controller (SBC) offers a complete connectivity solution for SMB enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

Scaling up to 60 concurrent sessions, the SBC60 connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment.

The SBC60 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**30~60 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC60 architecture can scale up from 30 to 60 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

# SBC60

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

| | |
|---|---|
| Network Interfaces | |
| **Ethernet:** | 2(10/100/1000 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:** | Desktop |
| **Power:** | 100-240V AC |
| **Environmental:** | Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃<br>Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC120

# Session Border Controllers (SBCs)

- 60~120 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC120 Session Border Controller (SBC) offers a complete connectivity solution for SMB enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

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

The SBC120 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**60~120 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC120 architecture can scale up from 60 to 120 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

# SBC120

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

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

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

| Interoperability | |
|---|---|
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |

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

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

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

| Physical/Environmental | |
|---|---|
| **Dimensions:** | 44*440*267mm |
| **Weight:** | About 3.1Kg |
| **Mounting:** | 19" rack mount |
| **Power:** | 100-240V AC redundant dual feed |
| **Environmental:** | Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃<br>Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC250

# Session Border Controllers (SBCs)

- 120~250 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC250 Session Border Controller (SBC) offers a complete connectivity solution for large enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

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

The SBC250 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**120~250 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC250 architecture can scale up from 120 to 250 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

Synway®

# SBC250

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

| Telephony Interfaces | |
|---|---|
| **Analog** | Optional |
| **Digital** | Up to 4E1/T1 Interfaces |
| **Clock Source** | 50 ppm High Precision |
| **Digital PSTN Protocols:** | ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy |
| **Network Interfaces** | |
| **Ethernet:** | 2(10/100/1000 BASE-TX(RJ-45)) & Customizable |
| **Security** | |
| **Access Control:** | DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System) |
| **Encryption/Authentication:** | TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication |
| **Privacy:** | Topology hiding, user privacy |
| **Traffic Separation:** | Self-adjustable automatic load balance |
| **Intrusion Detection System:** | Detection and prevention of VoIP attacks, theft of service and unauthorized access |
| **VoIP firewall:** | Optional |
| **Interoperability** | |
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |
| **Voice Quality and SLA** | |
| **Call Admission Control:** | Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions |
| **Packet Marking:** | 802.1p/Q VLAN tagging, DiffServ |
| **Standalone Survivability:** | Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911). |
| **Impairment Mitigation:** | Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation |
| **Voice Monitoring and Enhancement:** | acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection |
| **Direct Media:** | Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption |
| **High Availability:** | SBC high availability with 1+1 redundancy, active calls preserved |
| **Test Agent:** | Ability to remotely verify SIP message flow between SIP UAs |
| **Echo cancellation:** | G.168 128 ms tail length |
| **Advanced Media Processing:** | T.38 real-time fax, T.38 – G.711 interworking |
| **SIP Routing** | |
| **Routing Criteria:** | Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth |
| **Route To:** | Configured SIP peers, registered users, IP address, request URI |
| **Advanced Routing Features:** | Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization |
| **SIPREC:** | SynAPI recording interface |
| **Management** | |
| **OAM&P:** | Browser-based GUI, SNMP, INI Configuration file |
| **Physical/Environmental** | |
| **Dimensions:** | 44*440*267mm |
| **Weight:** | About 3.1Kg |
| **Mounting:** | 19” rack mount |
| **Power:** | 100-240V AC redundant dual feed |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

# SBC500

## Session Border Controllers (SBCs)

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


With versatile and robust architecture, The Synway SBC500 Session Border Controller (SBC) offers a complete connectivity solution for large enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

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

The SBC500 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**250~500 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

### High interoperability

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

### Enhanced security

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

### Superior voice quality

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

### High resiliency

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

### Flexible scalability

The SBC500 architecture can scale up from 250 to 500 sessions, and the various licensing options assure economical scalability

### Basic Features and Functions For SBC

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

# SBC500

## Capacities

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

## Network Interfaces

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

## Security

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

## Interoperability

| | |
|---|---|
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |

## Voice Quality and SLA

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

## SIP Routing

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

## Management

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

## Physical/Environmental

| | |
|---|---|
| **Dimensions:** | 44*440*267mm |
| **Weight:** | About 3.1Kg |
| **Mounting:** | 19” rack mount |
| **Power:** | 100-240V AC redundant dual feed |
| **Environmental:** | Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃<br>Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC1000

# Session Border Controllers (SBCs)

- 500~1,000 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC1000 Session Border Controller (SBC) offers a complete connectivity solution for large enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

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

The SBC1000 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**500~ 1,000 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC1000 architecture can scale up from 500 to 1000 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

# SBC1000

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

| | |
|---|---|
| **Network Interfaces** | |
| **Ethernet:** | 2(10/100/1000 BASE-TX(RJ-45)) & Customizable |
| **Security** | |
| **Access Control:** | DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System) |
| **Encryption/Authentication:** | TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication |
| **Privacy:** | Topology hiding, user privacy |
| **Traffic Separation:** | Self-adjustable automatic load balance |
| **Intrusion Detection System:** | Detection and prevention of VoIP attacks, theft of service and unauthorized access |
| **VoIP firewall:** | Optional |
| **Interoperability** | |
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |
| **Voice Quality and SLA** | |
| **Call Admission Control:** | Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions |
| **Packet Marking:** | 802.1p/Q VLAN tagging, DiffServ |
| **Standalone Survivability:** | Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911). |
| **Impairment Mitigation:** | Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation |
| **Voice Monitoring and Enhancement:** | acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection |
| **Direct Media:** | Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption |
| **High Availability:** | SBC high availability with 1+1 redundancy, active calls preserved |
| **Test Agent:** | Ability to remotely verify SIP message flow between SIP UAs |
| **Echo cancellation:** | G.168 128 ms tail length |
| **Advanced Media Processing:** | T.38 real-time fax, T.38 – G.711 interworking |
| **SIP Routing** | |
| **Routing Criteria:** | Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth |
| **Route To:** | Configured SIP peers, registered users, IP address, request URI |
| **Advanced Routing Features:** | Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization |
| **SIPREC:** | SynAPI recording interface |
| **Management** | |
| **OAM&P:** | Browser-based GUI, SNMP, INI Configuration file |
| **Physical/Environmental** | |
| **Dimensions:** | 44*440*690mm |
| **Weight:** | About 12Kg |
| **Mounting:** | 19" rack mount |
| **Power:** | 100-240V AC redundant dual feed |
| **Environmental:** | Operating temperature: 0℃ —40℃ ;Storage temperature: -20℃ —85℃<br>Humidity: 8%— 90% non-condensing;Storage humidity: 8%— 90% non-condensing |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC2000

# Session Border Controllers (SBCs)

- 1000~2,000 Pure IP SBC Sessions with Various Licensing
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC2000 Session Border Controller (SBC) offers a complete connectivity solution for large enterprises and service provider and enables scalable, reliable and secured connectivity between diverse VoIP networks.

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

The SBC2000 could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between VoIP networks, such as connecting IP-PBX systems to any IP-based applications.

**1000~2,000 SBC Sessions | 1+1 High Availability | Pure IP SBC | Support OPUS & SILK**

## High interoperability

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Flexible scalability

The SBC2000 architecture can scale up from 1000 to 2000 sessions, and the various licensing options assure economical scalability

## Basic Features and Functions For SBC

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

# SBC2000

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

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

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

| Interoperability | |
|---|---|
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |

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

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

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

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

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC60H

# Hybrid SBC and Media Gateway

- 30~60 Hybrid IP SBC Sessions & TDM Survivability
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)


With versatile and robust architecture, The Synway SBC60H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for SMB enterprises and service provider.

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

The SBC60H could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN networks.

**30~60 SBC Sessions | 1+1 High Availability | High Survivability | 30+ TDM Sessions**

## High interoperability

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

## Hybrid functionality

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Basic Features and Functions For SBC

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

# SBC60H

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

| Telephony Interfaces | |
|---|---|
| **Analog** | Optional |
| **Digital** | Up to 2E1/T1 Interfaces |
| **Clock Source** | 50 ppm High Precision |
| **Digital PSTN Protocols:** | ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy |
| **Network Interfaces** | |
| **Ethernet:** | 2(10/100/1000 BASE-TX(RJ-45)) & Customizable |
| **Security** | |
| **Access Control:** | DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System) |
| **Encryption/Authentication:** | TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication |
| **Privacy:** | Topology hiding, user privacy |
| **Traffic Separation:** | Self-adjustable automatic load balance |
| **Intrusion Detection System:** | Detection and prevention of VoIP attacks, theft of service and unauthorized access |
| **VoIP firewall:** | Optional |
| **Interoperability** | |
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |
| **Voice Quality and SLA** | |
| **Call Admission Control:** | Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions |
| **Packet Marking:** | 802.1p/Q VLAN tagging, DiffServ |
| **Standalone Survivability:** | Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911). |
| **Impairment Mitigation:** | Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation |
| **Voice Monitoring and Enhancement:** | acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection |
| **Direct Media:** | Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption |
| **High Availability:** | SBC high availability with 1+1 redundancy, active calls preserved |
| **Test Agent:** | Ability to remotely verify SIP message flow between SIP UAs |
| **Echo cancellation:** | G.168 128 ms tail length |
| **Advanced Media Processing:** | T.38 real-time fax, T.38 – G.711 interworking |
| **SIP Routing** | |
| **Routing Criteria:** | Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth |
| **Route To:** | Configured SIP peers, registered users, IP address, request URI |
| **Advanced Routing Features:** | Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization |
| **SIPREC:** | SynAPI recording interface |
| **Management** | |
| **OAM&P:** | Browser-based GUI, SNMP, INI Configuration file |
| **Physical/Environmental** | |
| **Dimensions:** | 190*30*120mm |
| **Weight:** | About 0.7Kg |
| **Mounting:** | Desktop |
| **Power:** | 100-240V AC |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC120H

# Hybrid SBC and Media Gateway

- 60~120 Hybrid IP SBC Sessions & TDM Survivability
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC120H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for SMB enterprises and service providers.

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

The SBC120H could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN networks.

**60~120 SBC Sessions | 1+1 High Availability | High Survivability | 30+ TDM Sessions**

## High interoperability

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

## Hybrid functionality

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Basic Features and Functions For SBC

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

# SBC120H

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

| Telephony Interfaces | |
|---|---|
| **Analog** | Optional |
| **Digital** | Up to 4E1/T1 Interfaces |
| **Clock Source** | 50 ppm High Precision |
| **Digital PSTN Protocols:** | ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy |

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

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

| Interoperability | |
|---|---|
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |

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

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

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

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

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC250H

# Hybrid SBC and Media Gateway

- 120~250 Hybrid IP SBC Sessions & TDM Survivability
- High Interoperability with Various SIP Trunks & Platforms
- Enhanced Security and High Resiliency(1+1 Redundancy)

With versatile and robust architecture, The Synway SBC250H Session Border Controller (SBC) and media gateway offers a complete connectivity solution for large enterprises and service provider.

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

The SBC250H could be customized to multiple voice channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN networks.

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

## High interoperability

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

## Hybrid functionality

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Basic Features and Functions For SBC

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

# SBC250H

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

| Telephony Interfaces | |
|---|---|
| **Analog** | Optional |
| **Digital** | Up to 8E1/T1 Interfaces |
| **Clock Source** | 50 ppm High Precision |
| **Digital PSTN Protocols:** | ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy |
| **Network Interfaces** | |
| **Ethernet:** | 2(10/100/1000 BASE-TX(RJ-45)) & Customizable |
| **Security** | |
| **Access Control:** | DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System) |
| **Encryption/Authentication:** | TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication |
| **Privacy:** | Topology hiding, user privacy |
| **Traffic Separation:** | Self-adjustable automatic load balance |
| **Intrusion Detection System:** | Detection and prevention of VoIP attacks, theft of service and unauthorized access |
| **VoIP firewall:** | Optional |
| **Interoperability** | |
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |
| **Voice Quality and SLA** | |
| **Call Admission Control:** | Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions |
| **Packet Marking:** | 802.1p/Q VLAN tagging, DiffServ |
| **Standalone Survivability:** | Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911). |
| **Impairment Mitigation:** | Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation |
| **Voice Monitoring and Enhancement:** | acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection |
| **Direct Media:** | Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption |
| **High Availability:** | SBC high availability with 1+1 redundancy, active calls preserved |
| **Test Agent:** | Ability to remotely verify SIP message flow between SIP UAs |
| **Echo cancellation:** | G.168 128 ms tail length |
| **Advanced Media Processing:** | T.38 real-time fax, T.38 – G.711 interworking |
| **SIP Routing** | |
| **Routing Criteria:** | Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth |
| **Route To:** | Configured SIP peers, registered users, IP address, request URI |
| **Advanced Routing Features:** | Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization |
| **SIPREC:** | SynAPI recording interface |
| **Management** | |
| **OAM&P:** | Browser-based GUI, SNMP, INI Configuration file |
| **Physical/Environmental** | |
| **Dimensions:** | 44*440*267mm |
| **Weight:** | About 3.1Kg |
| **Mounting:** | 19" rack mount |
| **Power:** | 100-240V AC redundant dual feed |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

SBC500H

# Hybrid SBC and Media Gateway

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


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

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

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

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

## High interoperability

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

## Hybrid functionality

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

## Enhanced security

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

## Superior voice quality

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

## High resiliency

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

## Basic Features and Functions For SBC

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

Synway®

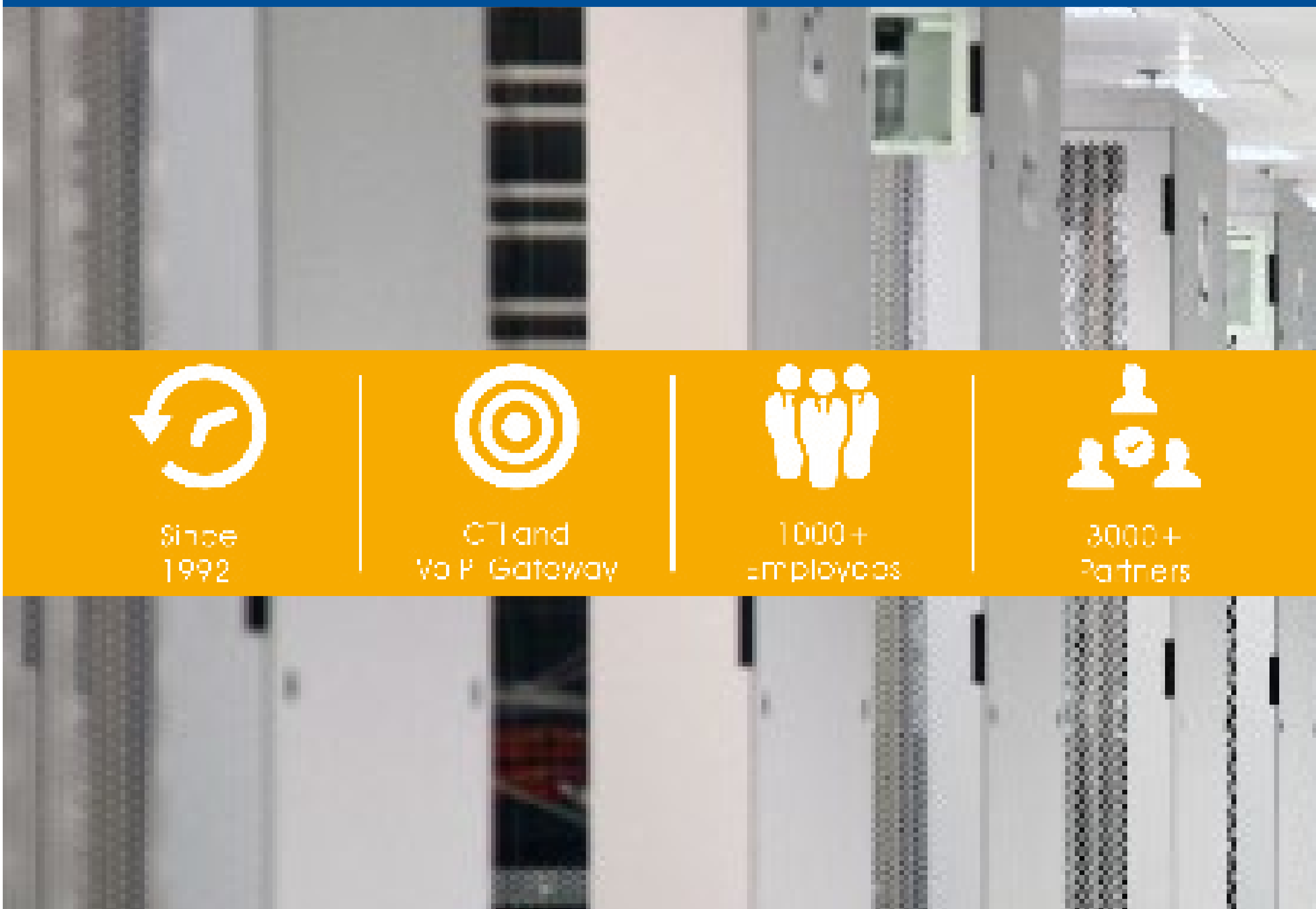
# SBC500H

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

| | |
|---|---|
| **Telephony Interfaces** | |
| **Analog** | Optional |
| **Digital** | Up to 8E1/T1 Interfaces |
| **Clock Source** | 50 ppm High Precision |
| **Digital PSTN Protocols:** | ISDN: ISDN User Side, ISDN Network Side, SS1: SS1 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy |
| **Network Interfaces** | |
| **Ethernet:** | 2(10/100/1000 BASE-TX(RJ-45)) & Customizable |
| **Security** | |
| **Access Control:** | DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System) |
| **Encryption/Authentication:** | TLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication |
| **Privacy:** | Topology hiding, user privacy |
| **Traffic Separation:** | Self-adjustable automatic load balance |
| **Intrusion Detection System:** | Detection and prevention of VoIP attacks, theft of service and unauthorized access |
| **VoIP firewall:** | Optional |
| **Interoperability** | |
| **SIP B2BUA:** | Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode |
| **SIP Interworking:** | 3xx redirect, REFER, PRACK, early media, call hold |
| **Registration and Authentication:** | User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users |
| **Transport Mediation:** | Mediation between SIP over UDP/TCP/TLS, IPv4/IPv6, RTP/SRTP |
| **Header Manipulation:** | Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions |
| **Number Manipulations:** | Ingress and egress digit manipulation |
| **Transcoding and Vocoders:** | Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.729, GSM-FR, AMR-NB, SILK-NB/WB, Opus-NB/WB |
| **Signal Conversion:** | DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion |
| **NAT:** | Hosted NAT, RTP self-adaption |
| **WebRTC controller:** | Optional or customizable |
| **Voice Quality and SLA** | |
| **Call Admission Control:** | Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions |
| **Packet Marking:** | 802.1p/Q VLAN tagging, DiffServ |
| **Standalone Survivability:** | Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback (including E911). |
| **Impairment Mitigation:** | Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort Noise Generation |
| **Voice Monitoring and Enhancement:** | acoustic echo cancellation, fixed and dynamic voice gain control, dynamic programmable jitter buffer, silence suppression, RTP redundancy, broken connection detection |
| **Direct Media:** | Hair-pinning (no media anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption |
| **High Availability:** | SBC high availability with 1+1 redundancy, active calls preserved |
| **Test Agent:** | Ability to remotely verify SIP message flow between SIP UAs |
| **Echo cancellation:** | G.168 128 ms tail length |
| **Advanced Media Processing:** | T.38 real-time fax, T.38 – G.711 interworking |
| **SIP Routing** | |
| **Routing Criteria:** | Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth |
| **Route To:** | Configured SIP peers, registered users, IP address, request URI |
| **Advanced Routing Features:** | Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization |
| **SIPREC:** | SynAPI recording interface |
| **Management** | |
| **OAM&P:** | Browser-based GUI, SNMP, INI Configuration file |
| **Physical/Environmental** | |
| **Dimensions:** | 44*440*267mm |
| **Weight:** | About 3.1Kg |
| **Mounting:** | 19” rack mount |
| **Power:** | 100-240V AC redundant dual feed |

Synway Information Engineering Co., Ltd.
Skirt Building, No. 1 Building, ZhongHeng Century Science Park, No. 4028, NanHuan Rd, BingJiang, Hangzhou, China 310053
Tel: (86) 571 88860561; Fax: (86) 571 88850923; Email: info@synway.net

Synway®




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

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

Synway Information Engineering Co., Ltd. (Headquarter)
Synway R&D Building, No.3756, Rd. Nanhuan.
Binjiang District, Hangzhou, P.R.China 310053
Tel: (86) 571 88860561
Fax: (86) 571 88850923
Email: info@synway.net


Technical Support
Tel: (86) 571 86692545
Tel: (86) 571 88864579
Mobile: (86) 18905817070 (24*7)
Email:techsupport@synway.net



FC CE RoHS ISO 9001-2008