SBC, IP-PBX & VoIP Gateway

Safe Bridge Between IP & TDM & Mobile Networks

Session Border Controllers (SBCs)

- 5~2,000 Sessions in 1U footprint
- Complete Security Protocols
- Dedicated DSP Transcoding
- High-Availability and Redundancy
- Survivability in IP & PSTN Networks

With versatile and robust architecture, The Synway's 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 2,000 concurrent sessions, the Synway's SBC connects IP-PBXs to any SIP trunking and cloud-based services, and offers superior performance in connecting any SIP to SIP environment. It 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.

IP-PBX (Embedded SBC Features)

- 5~100 Concurrent Calls in 1U footprint
- 10~1,000 Registered Users
- Comprehensive UC Features
- Survivability Via PSTN Network
- Integrated Security(SRTP/RTP/HTTPS)

With an advanced hardware platform and software functionalities, the UC series IP-PBX can support up to 1,000 registered users and offer effortless setup and deployment via the web-browser user interface. Besides auto-discovery of diverse endpoints and auto-provisioning, the UC series offers a set of comprehensive features, including customizable call-routing, multi-level IVRs, call queues, auto-attendant, call detail records (CDR), multi-site peering, voicemail/fax forwarding to email and more.

VoIP Gateway (TDM-IP/IP-IP)

- GSM/WCDMA/VoLTE Gateway: 4~32Chs
- PRI/SS7/SIGTRAN/Transcoding Gateway: 30~1,920Chs
- Hybrid Gateway: T1/E1/FXO/FXS/Wireless
- FXO/FXS Gateway: 4~32Chs
- Dedicated DSP-Powered Transcoding

Thanks to the past two years' continuous efforts, Synway finally made a huge breakthrough and brought out a whole new package of VoIP solutions. Our VoIP gateway, different than rivals, not just for its high cost advantages, but relies more on the easy-to-use, customizable and Plug-And-Play capabilities. We are proud of our world-class voice optimization technologies, which ensures seamless interoperability under any complex network environments and also makes the VoIP gateway an ideal option for system integrators and software developers.

Session Border Controllers (SBCs)

Model	SBC30	SBC60	SBC250	SBC500	SBC1000	SBC2000
Capacities						
Signaling	5-30	30-60	60-250	150-500	500-1000	1000-2000
Transcoding Sessions	5-30	30-60	60-250	150-500	500-1000	1000-2000
RTP/SRTP Sessions	5-30	30-60	60-250	150-500	500-1000	1000-2000
Max Registered Users	128	128	1024	2048	4000	8000
Interfaces						
PSNT	-	Up to 1 E1	Up to 6 E1	-	-	-
Ethernet	2*(10/100 BASE-TX (RJ-45))		2*(10/100/1000 BASE-TX (RJ-45))		2*(10/100/1000 BASE-TX (RJ-45))	
Physical						
Dimensions (mm)	190*30*120	190*30*120	440*44*267	440*44*267	440*44*690	440*44*690
Weight (kg)	≈0.65	≈0.65	≈3.1	≈3.1	≈12	≈12
Mounting	Desktop	Desktop	19'' rack mount, 1U		19'' rack mount, 1U	
Power supply	12V DC	12V DC	110-240 AC redundant dual feed		110-240 AC redundant dual feed	
Telephony Interfaces (SBC60 & SBC250)						
Digital	Up to 6E1/T1 Interfaces					
Clock Source	50 ppm High Precision					
Digital PSTN Protocols:	ISDN: ISDN User Side, ISDN Network Side, SS1: SS1/MFCR2 Signaling; SIP signaling: SIP V1.0/2.0, RFC3261; SS7 MTP1~3,SS7 TCAP, SS7 ISUP, SIGTRAN, SS7 1+1 active/standby redundancy					
Security						
Access Control:	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)					
Encryption/Authentication:	TLS, SRTP, 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.					
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, emergency call detection and prioritization					
SIPREC:	SynAPI recording interface					
Management						
OAM&P:	Browser-based GUI, SNMP, INI Configuration file					
Environmental						
Environmental:	Operating temperature: 0℃ -40℃ ; Storage temperature: -20℃ -85℃ Humidity: 8%- 90% non-condensing; Storage humidity: 8%-90% non-condensing					

IP-PBX (Embedded SBC Features)

Model	UC200	UC500	UC500H
Capacities			
Registered Users	60/120/220/300	150/300/400/500	150/300/400/500
Concurrent calls	15/30/45/60	30/60/80/100	30/60/80/100
Conference attendees	15/30/45/60	30/60/80/100	30/60/80/100
Interfaces			
FXS/FXO	2FXS+4FXO	2FXS+2FXO	16 to 96 FXS/FXO
GSM/UMTS/VoLTE	-	-	8 to 48 ports
Digital Trunking	-	-	1/2/4/8/16 E1 (PRI ISDN/SS7)
Network	2*(10/100 BASE-TX (RJ-45))	2*(10/100 BASE-TX (RJ-45))	2*(10/100 BASE-TX (RJ-45))
Physical			
Power Supply	12V DC, >3A	100-220V DC	100-220V DC
Dimensions(mm)	186*30*108	440*44*202	440*88*372
Weight(kg)	0.83	2.52	5.4
Mounting	Desktop	19'' rack mount, 1U	19'' rack mount, 2U
Interfaces			
NAT Router	Yes		
Peripheral Ports	USB, TF		
LED Indicators	Power/Ready, Network, PSTN Line, USB, TF		
Reset Switch	Yes		
Voice/Video Capabilities			
Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 32~128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer		
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, GSM, AAL2-G.726-32; T.38		
Video Codecs	H.264, H.263, H263+		
QoS	Multiple Layers		
Signaling & Control			
DTMF Methods	In Audio, RFC2833, and SIP INFO		
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of various IP endpoints with no Configuration		
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP(RFC3261), STUN, SRTP, TLS, LADP		
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone		
Security			
Media Encryption	SRTP, TLS, HTTPS, TELNET with Fail2ban, Whitelist, Blacklist, alerts and more to protect against attacks		
Environmental			
Environmental	Operating: 32 ~ 113ºF / 0 ~ 45ºC, 8 ~ 90% (non-condensing); Storage: -4 ~ 185ºF / -20 ~ 85ºC		
Additional Features			
Multi-Language Support	English/Chinese/Traditional Chinese for Web UI; Customizable IVR/voice prompts for English, Chinese, British English;Customizable language pack to support any other languages		
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF		
Polarity Reversal/Wink	Yes, with enable/disable option upon call establishment and termination		
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/availability/ busy level, in-queue announcement		
Customizable Auto Attendant	Unlimited layers of IVR (Interactive Voice Response)		
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom etc.		

VoIP Gateway (TDM-IP/IP-IP)

Models	SMG1000	SMG2000L	SMG2000	SMG3000	SMG4000	UNIWAY	SIM Bank
Products models	4-32 Ports FXO/FXS	1/2 Ports E1/T1	1/2/4 Ports E1/T1	8/16/64 Ports E1/T1	4-32 Ports	Up to 6/8 slots for modules	64/128 Sim Slots
Segmentation	SMB	SMB Enterprise	Enterprise Carrier	Carrier	Enterprise Carrier	Enterprise Carrier	Enterprise Carrier
Sessions	4-32 Chs	30-60 Chs	30-120 Chs	240-1920 Chs	4-32 Chs	4-600 Chs	128 Sim Slots
Network	2 (10/100 BASE-TX (RJ-45))	2 (10/100 BASE-TX (RJ-45))	2 (10/100/1000 BASE-TX (RJ-45))	2 (10/100/1000 BASE-TX (RJ-45))	2 (10/100 BASE-TX (RJ-45))	2 (10/100 BASE-TX (RJ-45))	2 (10/100 BASE-TX (RJ-45))
LAN Port	1-2	2	2	2	1-2	2	2
TDM interface	FXO/FXS	E1/T1/J1	E1/T1/J1	STM-1	GSM/CDMA/ WCDMA/3G/ VoLTE	Hybrid	GSM/CDMA/ WCDMA/3G/ VoLTE
Signaling	SIP/TDM	SIP/ ISDN R2/CAS	SIP/MGCP/SS7/SIGTRAN ISDN/R2/CAS		SIP/GSM CDMA/ WCDMA	Hybrid	SIP/GSM CDMA/ WCDMA
Media capability	DSP-based						
Codecs	G.711/729/723/722/iLBC/AMR						
Power Supply	Single	Single	Dual	Dual	Single	Single/dual	Single
Power Requirements	12V DC or 100-240V AC	12V DC	100-240V AC	100 – 240V AC	12V DC+10% ≥3A	100-240V AC	100-240V AC
Media Resources	Echo Cancellation/Fax/Conferencing						
Routing	Call routing (IP ↔PSTN)	Call routing and translation(PCM↔IP)			Call routing IP↔Wireless	Call routing IP↔PSTN	Call routing IP↔Wireless
Environment	Operating temperature range :0 to +55 °C, 8-90% relative humidity non-condensing Storage temperature range:-20 to +85 °C, 8-90% relative humidity non-condensing						
Safety	Compliant with most international standards, please ask Synway or its sales representatives worldwide. Synway would comply all new safety standard to for different regions around the world while needed.						
EMC/EMI	Compliant with most international standards. For compliance documents, please contact Synway's sales representatives.						
OAM&P	Network Time Protocol(NTP) Web User interface (WebUI) supports configuration via browser SNMP MIBs						

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