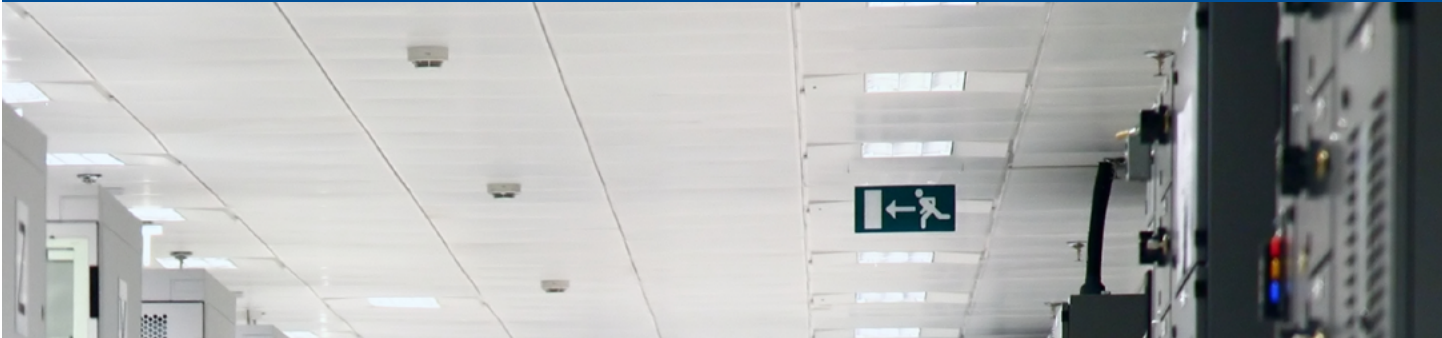


Get Connected with



 **SYNWAY**®



SMG Series VoIP Gateway

Seamless bridge between VoIP & TDM Networks

Synway Information Engineering Co., Ltd.

VoIP Gateway

Seamless bridge between VoIP & TDM Networks

With over two decades of expertise in signaling and voice technology, Synway VoIP gateway family helps customers access to IP networks from legacy telephony applications more reliably and efficiently. Our field-proven PSTN/IP signaling technology possesses unparalleled and seamless interoperability with any complex network environments, which exponentially reduces your investments and time to markets under any uncertain situations.

Thanks to long standing IP expertise, Synway has brought out a whole new package of VoIP solutions. Our VoIP gateway, different than rivals, not only delivers its high cost advantages, but also features easy-to-use, customizability, Plug-And-Play capabilities, and world-class voice optimization technologies. It ensures seamless interoperability with any complex network environments and also makes the VoIP gateway an ideal option for system integrators, SPs and enterprises.

Key Features and Benefits

- Widest product range

Widest option for enterprises and carriers, portfolios include cost-effective and versatile FXO/FXS/GSM/WDCMA/VOLTE/E1/T1 VoIP gateway, high-performance hybrid VoIP gateway and carrier-grade SS7/SIGTRAN VoIP Gateway, as well as transcoding, call classification and SBC gateway for specific requirements.

- Flexible configuration

Compliant with diverse networks (FXO, FXS, T1, E1, J1, GSM, CDMA, 3G, VoLTE, IP): support various multimedia processing capability (conferencing, fax, compression and SuPerForm echo cancellation for voice enhancement) and an array of Protocols (SS7, SIGTRAN, ISDN PRI, CAS, R1, R2, Wireless).

- Compliance with any applications

With optional inbuilt industrial server, Synway's VoIP gateways are compliant with any IP-based applications; it also supports any category of third party software, including UC, IP-PBX, Contact Center and more. In legacy PSTN network, Synway VoIP gateway could converge applications based on legacy networks.

- Low to high scalability

Modular architecture and wide product range ensure flexibility and expandability from low density to high density. In a single chassis, 2~1,920 channels are available for diverse applications. For all product categories, a unified management software allows for easy configurations, system upgrading or general maintenance.

- Diverse media resources

Support high-capacity voice playback and Codecs, conferencing, faxing; Support T.38/T.30; optimized for IP-PBX, IVR and ACD applications, with EXT IVR server or GUI management.

- Carrier-Grade reliability

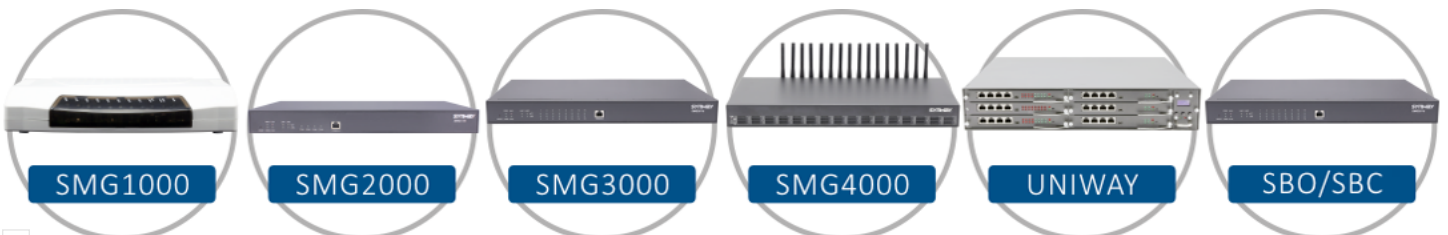
Special power system with standby redundancy; advanced cooling system to reassure long-standing robustness ; special air cleaner to protect against dust accumulation inside chassis; RoHS compliant; Inside temperature control and alert system.

- Cloud-Based management system (DCMS)

Real-Time Monitoring of VoIP Gateway Status; Management Capability for More Than 10,000 Devices; DCMS Features Cloud-based Management Capability; Superior Network Access Traversal (NAT) Capability. For Detail, Please refer to www.synway.net.

Technical Specification:

Models	SMG1000	SMG2000S	SMG2000	SMG3000	SMG4000	UNIWAY	SBO	SBC
Products models	2~32 Ports FXO/FXS configurable	1/2/4 Ports E1/T1	1/2/4 Ports E1/T1	8/16/64 Ports E1/T1	4~32 Ports	8 slots for modules	500~2,000 ports	SBC
Segmentation	SMB	SMB Enterprise	Enterprise Carrier	Carrier	Enterprise Carrier	Enterprise Carrier	Carrier	Enterprise Carrier
Sessions	4-32 Chs	30-240 Chs	30-120 Chs	240-1920 Chs	4-32 Chs	4-600 Chs	500-2000 Chs	1-500 Chs
Network	100M Ethernet	1000M Ethernet						
LAN Port	1-2	2	2	2	1-2	2	2	2
TDM interface	FXO/FXS	E1/T1/J1	E1/T1/J1	STM-1	GSM/CDMA/WCDMA/3G/VoLTE	Hybrid	N/A	N/A
Signaling	SIP/TDM	SIP/ ISDN R2/CAS	SIP/MGCP/SS7/SIGTRAN ISDN/R2/CAS		SIP/GSM CDMA/WCDMA	Hybrid	SIP	SIP
Media capability	DSP-based							
Codecs	G.711/729/723/722/iLBC/AMR							
Power Supply	Single	Single/dual			Single	Single/dual		
Power Requirements	AC Power Supply Range 100 – 240 VAC The power supply will operate at frequencies between 47 Hz and 63 Hz				Input voltage: DC12V+10% Input current:≥3A	AC: 90-120V or 200~265V Power consumption: less than 350W	Input voltage:100-240V AC Maximum power consumption:≤22W	
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)	N/A	N/A
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							



SMG1000

FXO/FXS VoIP Gateway

- Support 2~32FXS/FXO Ports, Field Approved Globally
- Superior Voice Quality by Designated DSP Chipsets
- User-Friendliness and Web-based Administration

With a simple and economical way to help legacy telephone, fax machine and PBXs interconnect with IP network. Synway SMG1000 analog gateway enables call center and multi-branch enterprises to process powerful, versatile and efficient VoIP solutions with unparalleled cost advantages.

SMG1000 Analog Gateway allows for a well-planned, phased migration to an IP network, making the gateways an easy solution for enterprises looking to enhance their legacy PBX equipment with new VoIP access and applications. Connected between a PBX or a LAN, the SMG1000 Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks.



Key Features and Benefits

- Fit to Diverse Applications

Used to connect IP telephones to a legacy PBX, integrate network-hosted applications with the PBX, extend the PBX to branch offices, and integrate various voice and call processing capabilities in an enterprise LAN or WAN environment;

- Flexible Call Routing

Route calls from the switched network to a VoIP destination on the IP network or conversely, and support the following call routing options: TDM to IP or IP to TDM, with IP load balancing, IP fault tolerance;

- Controlled Migration to IP

Compatible with general FXO/FXS lines, and a variety of popular PBX manufacturers (Digital PSTN lines compliance would be available), and protects investment in legacy telecommunications equipment;

- User-Friendly Management

Supports configuration via serial, Telnet, and a web browser including context-sensitive help
Easy to install, configure, debug, and maintain;

- Enhanced Voice/Fax Processing

Supports a variety of compression algorithms, including G.711 A-law and μ -law, G.729A/B; Transcode fax from T.30 fax protocol (supporting V.17) to T.38 for transmission over a packet network;

- Web Server Interface & Hot Swap

Each gateway unit is delivered with a web server interface, allowing configuration and software upgrades via a web browser; Allows gateway units to be added or removed without affecting other gateway;

Technical Specification:					
Features	Model	SMG1004	SMG1008	SMG1016	SMG1032
Model		SMG1004-2S20 SMG1004-4S SMG1004-4O	SMG1008-4S4O SMG1008-8S SMG1008-8O	SMG1016-8S8O SMG1016-16S SMG1016-16O	SMG1032-16S16O SMG1032-32S SMG1032-32O
Sessions		4	8	16	32
Telephony ports		4	8	16	32
Interface		RJ11	RJ11	RJ11, RJ21, RJ45	RJ11, RJ21, RJ45
LAN		1*10/100Mbps		2*10/100Mbps	
Console Port		N/A		1	
Signaling		SIP/TDM			
Power Supply		Single			
Power Requirements		12V DC		AC Frequency 47 Hz to 63 Hz	
Codecs		G.711/729/723/722/iLBC/AMR			
DTMF Mode		RFC2833, SIP Info, In-band			
Fax		T.38 and Pass-through			
QoS		DiffServ, ToS, 802.1 P/Q VLAN tagging			
Network		DHCP, PPPoE, Static Route, VLAN			
Network Protocol		TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN			
Management Protocol		TR-069*			
Signaling		FXS Loop Start, FXS Kewl Start			
Caller ID		BELL202, ETSI (V23), NTT (V23-Japan), and DTMF-based CID			
FXO Connectivity		Programmable AC Impedance, Hangup Detection, Answer Detection, Caller ID Detection			
Dimensions H*W*D(mm)		220*148*40		440*267*44	
Routing		Call routing(IP ↔ PSTN)			
Mounting		Desktop		Wall-mount	
Compatibility		Interoperable with 3CX, Asterisk, Lync Server, FreePBX and certified with Elastix and BroadSoft			
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			

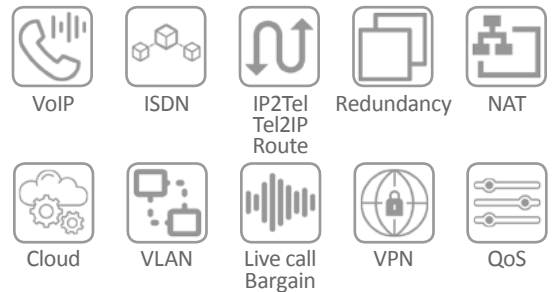
SMG2000L

Mini Size E1/T1 VoIP Gateway

- Support 1/2 E1(T1 Optional) in Mini Size
- Support SIP and SS7/ISDN/R2/CAS and More
- Superior Voice Quality and Reliability in Full Load
- Only 30mm*190mm *120mm(High*Wide*Deep)

The Synway SMG2030L/2060L, with mini size for better space and shipment efficiency, are new members of Synway's VoIP gateway family that enables service providers and enterprises to maximize value of their networks and services. It converts digital E1/T1 PSTN message into IP formats and secures sessions across IP and mixed network boundaries to support the seamless delivery of services.

SMG2030L/2060L are unparalleledly cost effective and compliant with PRI ISDN, R2 and SS7 packets, and adopt the equivalent hardware architecture like Synway's Telco' grade SMG3000 series, with dedicated DSP chipsets for processing IP/TDM signaling and optimizing voice quality. Compared to with rival products, SMG2000L features high reliability and unmatched cost, and delivers a perfect alternative option for enterprises, operators and system integrators.



Key Features and Benefits

- Flexible Scalability
From 30 to 120 simultaneous SIP sessions with multimedia transcoding provides high performance in a small footprint to help lower ownership cost and operational cost; a broad range of gateway toolkits help gateway's maintenance and software upgrading for Web UI, gateway services and firmware;
- Flexible and efficient Gateway Solution
Provide a clear migration path to an all-IP network. It can scale up to 60 simultaneous IP sessions and at the same time provide media transcoding and impressive sessions per second performance; Support SS7 signaling, call routing, call automated failover from IP to TDM for outbound routing;
- Any-to-Any Signaling and Transcoding
Provide any-to-any network connectivity through its ability to interwork multiple protocols to deliver services. It can provide interworking between SS1, R2, ISDN, SS7, SIP formats and any-to-any media transcoding for popular voice codecs, T.38 and G.711 fax interworking, RTP, INBAND and SIPINFO;
- User-Friendly Management & Toolkits
The Web graphical user interface (WebUI) is a real-time web toolkits to configure and monitor and perform real-time monitoring and maintenance. Flexible SIP and Protocols configuration help to configure SIP, SIP trunking, SIP Mediation, PCM, SS7 and ISDN, Routing and more;
- Combined All Features on A Single Platform
Integrated multimedia gateway features facilitate TDM and IP interworking to provide service delivery flexibility and automated failover between domains;
- Integrated Transcoding for Voice, Tone and Faxing
Eliminates the need to add separate hardware to support transcoding requirements helping to reduce CAPEX and number of platforms deployed, and support a range of Codecs, including G.723, G.729, G.711, iLBC, SIPINFO, RFC2833, RF3261, INBOUND and more;

Technical Specification:			
Features	Model	SMG2030L	SMG2060L
Sessions		30	60
Routing Features		Call routing and translation (from PCM to IP or reversely)	
IP Interfaces		Dual redundant 2 *100 Base-T Ethernet for VoIP payload and signaling	
IP protocols		TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN, etc.	
Coder support		G.711A,G.711U, G.729 A/B,G723,G722, GSM, iLBC, RFC 2833,RF 3261,SIPINFO,INBOUND	
Power Supply		Single	
TDM Signaling Protocols		ISDN PRI/MF R2/SS7 ISUP/SS7 MTP1~3/SS7 SIGTRAN/SS7 TCAP	
Power Requirements		12V DC	
Codecs		G.711/729/723/722/iLBC/AMR	
Mounting		Desktop	
Compatibility		Interoperable with 3CX, Asterisk, Lync Server, FreePBX and certified with Elastix and BroadSoft	
Dimensions H*W*D(mm)		30*190*120	
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	
Routing		Call Routing and translation (PCM↔IP)	
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	

Perfect interoperability with major IP PBX brands



SMG2000/3000

E1/T1 VoIP Gateway

- Support 1/2/4/8/16/63 E1(T1 Optional)in 1U Space
- Support SIP and SS7/ISDN/R2 and More
- Telco-grade Redundancy & Dual Power Supply

With a more efficient manner to interconnect and deliver large-scalability services, The Synway SMG2000/3000 enables service providers and carriers to maximize value of their networks and services. The SMG2000/3000 converts digital PSTN message into IP formats and connects and secures sessions across IP and mixed network boundaries to support the seamless delivery of services.

The MG2000/3000 connects IP and hybrid networks via telephony and E1(T1 or STM-1)/Ethernet links in a compact 1U form factor appliance. It also transforms media and signaling to support efficient and reliable voice, fax and multimedia sessions for mobile, fixed and cloud-based applications. The convergence of IP multimedia and TDM gateway functionality in a single chassis in the SMG offers significant reductions in investment and operational cost when compared to less integrated alternatives.



VoIP



SS7



IP2Tel
Tel2IP
Route



Redundancy



NAT



Cloud



VLAN



Radius



VPN



QoS

Key Features and Benefits

- Flexible Scalability

From 30 to 1,920 simultaneous SIP sessions with multimedia transcoding provide high performance in a small footprint to help lower ownership cost and operational cost;

- Flexible and Efficient Gateway Solution

Provide a clear migration path to an all-IP network. It can scale up to 1,920 simultaneous IP sessions and at the same time provide media transcoding and impressive sessions per second performance; Support SS7 signaling, call routing, call automated failover from IP to TDM for outbound routing;

- Any-to-Any Signaling and Transcoding

Provide any-to-any network connectivity through its ability to interwork multiple protocols to deliver services. It can provide interworking between SS1, R2, ISDN, SS7, SIP formats and any-to-any media transcoding for popular voice codecs, T.38 and G.711 fax interworking, RTP, INBAND and SIPINFO;

- Combined All Features on A Single Platform

Integrated multimedia gateway features facilitate TDM and IP interworking to provide service delivery flexibility and automated failover between domains;

- Integrated Transcoding for Voice, Tone and Faxing

Eliminates the need to add separate hardware to support transcoding requirements helping to reduce CAPEX and number of platforms deployed, and support a range of Codecs, including G.723, G.729, G.711, iLBC, SIPINFO, RFC2833, RFC3261, INBOUND and more;

Technical Specification:						
Models	SMG2030	SMG2060	SMG2120	SMG3008	SMG3016	SMG3064 SMG3064R
Sessions	30	60	120	240	480	1890
Segmentation	Enterprise			Carrier-grade		
Telephony ports	1	2	4	8	16	63
Interface	RJ45	RJ45	RJ45	RJ45	RJ45	Fiber Optical
Power Supply	Dual					
Power Requirements	AC Power Supply Range 100 – 240 VAC The power supply will operate at frequencies between 47Hz and 63Hz					
Codecs	G.711/729/723/722/iLBC/AMR					
Dimensions H*W*D(mm)	44*440*267					44*440*690
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					
Routing	Call Routing and translation (PCM↔IP)					
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					

Compatible ITSP



SMG4000

GSM/WCDMA/UMTS/VoLTE VoIP Gateway

- Support 4/8/16/32 GSM/WCDMA/CDMA/UMTS/LTE Protocols
- Superior Voice Quality by Designated DSP Chipsets
- Support SMS and A Range of Mobile Applications

Synway SMG4000 Wireless Gateway could be compliant with a variety of wireless/mobile protocols (2G/3G/4G), enabling interconnection between GSM/CDMA/WCDMA/VoLTE network and VoIP network smoothly and safely. It is able to bridge wireless network with IP networks efficiently, regarding to the high-demanding user requirements.

SMG4000 adapts self-propelled SIM card slots, advanced built-in VoIP processors and wireless modules, and helps enterprises and SPs launch diverse cost-efficient and flexible Wireless-to-IP communication systems. SMG4000 could also be applied into a range of systems, including remote billing and charge, Mobility PBX, PSTN backup lines, VoIP localization, SMS platform and more.



VoIP



GSM/UMTS
VoLTE



IP2Tel
Tel2IP
Route



Carrier
Selection



NAT



Cloud



VLAN



SMS



VPN



QoS

Key Features and Benefits

- DSP-based Algorithm

DSP-enabled voice optimization to assure crystal voice quality and maximize bandwidth efficiency; High-speed response and connectivity in the extreme network environments, with better run efficiency;

- High security

Support a range of SIM Block manners. In specific environments, SMG4016/SMG4032 (16/32 Ports) could use and activate multiple SIM cards circularly, improve system security, make full use of bandwidth, and increase ROI.

- Complete Protocols Range

Support standard SIP protocols, and could be used worldwide; Support both 2G/3G/4G wireless network (in different versions), including both GSM/WCDMA/VoLTE and More;

- High Voice Optimization Capability

Adopt Synway's homegrown voice optimization technologies to ensure crystal clear communication, including DSP-based 128mc echo cancellation; Telco-grade reliability and continuous high performance in fully loaded capacity and in the long run;

- High Flexibility and Scalability

Could be configured from 4/8/16/32 Ports of Wireless-to-IP transmission, and support a diversity of wireless networks in a single system

- User-friendly GUI

Easy-to-use service Web based UI configuration and management tools could help accelerate service deployment and simplify platform management

- High Interoperability with Terminals

Compliant with all brands of terminal mobile devices and a complete range of SIP trunking worldwide, support auto-provision in complex network environment

- Telco-Grade Reliability

Adopts telco-grade standards and components to design and manufacture, and certified and approved by most telecom operators worldwide

Technical Specification:					
Features	Model	SMG4004	SMG4008	SMG4016	SMG4032
Models		SMG4004G SMG4004C SMG4004W SMG4004WA SMG4004WT	SMG4008G SMG4008C SMG4008W SMG4008WA SMG4008WT	SMG4016G SMG4016C SMG4016WA SMG4016WT	SMG4032G SMG4032C SMG4032WA
Sessions		4	8	16	32
Telephony ports		1	2	4	8
TDM interface		GSM/CDMA/WCDMA/3G/VoLTE			
Interface		RJ45	RJ45	RJ45	RJ45
SIM card slots		4	8	64	128
Power Supply		Single/dual			
Power Requirements		12V DC			
Protocol		SIP, IAX2			
Transport Protocol		UDP, TCP, TLS, SRTP			
Codecs		G.711 (alaw/ulaw), G.722, G.726, G.729A, GSM, ADPCM, Speex			
Voice Capability		ITU-T G.168 LEC			
DTMF Mode		RFC2833, SIP Info, In-band			
Calling Type		Termination (VoIP to GSM/WCDMA), Origination (GSM/WCDMA to VoIP)			
Fax		T.38 and Pass-through			
QoS		DiffServ, ToS, 802.1 P/Q VLAN tagging			
Network		DHCP, DDNS, Firewall, OpenVPN, Static IP, QoS, Static Route, VLAN			
Network Protocol		FTP, TFTP, HTTP, SSH			
NAT Traversal		Static NAT, STUN			
Management Protocol		RADIUS, TR-069*			
PRI switch types		Euro ISDN, nation, Q.SIG			
CAS		MFC R2 (Argentina, Brazil, China, Czech Republic, Colombia, Ecuador, Indonesia, ITU, Mexico, Philippines, Venezuela)			
SS7		ITU, ANSI, China			
Dimensions H*W*D(mm)		260*153*30		440*200*44	
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			

Operator & Frequency	
Model	Frequency
2G	
SMG4000G	GSM:850/900/1800/1900MHz
SMG4000C	CDMA:800MHz
SMG4000GZ	GSM:850/900/1800/1900MHz
3G	
SMG4000W	GSM:900/1800MHz UMTS:900/2100MHz
SMG4000WA	GSM:850/900/1800/1900MHz UMTS:850/1900MHz
SMG4000WT	GSM:850/900/1800/1900MHz, UMTS:850/2100MHz
SMG4000WZ	GSM:8850/900/1800/1900MHz UMTS:50/900/1900/2100MHz
4G	
SMG4000LC	FDD LTE:B1/B3, TDD LTE:B38/B39/B40/B41 TDSCDMA:B34/B39, WCDMA:B1, CDMA2000:BC0 GSM:1X/EVD, 900/1800MHz
SMG4000LE	FDD LTE:B1/B3/B5/B7/B8/B20, TDD LTE:B38/B40/B41 WCDMA:B1/B5/B8, GSM:B3/B8
SMG4000LV	FDD LTE: B4/B13
SMG4000LT	LTE FDD: B1/B2/B3/B4/B5/B7/B8/B28; LTE TDD: B40 WCDMA: B1/B2/B5/B8; GSM: B2/B3/B5/B8

Compatible ITSP



UNIWAY

Hybrid VoIP Gateway

- Interface with FXO, FXS, T1, E1, GSM, CDMA, 3G, VoLTE, IP
- Optional Inbuilt Server, Standalone for any applications
- Telco Grade Standards in Both Hardware and Software

In hybrid VoIP & TDM era with diverse and rapidly-growing needs of unified communications and services, Synway UNIWAY VoIP Gateway adopts the latest modular architecture with built-in server, and opens a new milestone to maximize VOIP and TDM network's value for SP and application developers. One of major advantage of UNIWAY is to reduce time to market and introduce innovative applications more efficiently and effectively. Tailored to satisfy diverse customers' needs, UNIWAY, with its open and standardized format, enables users to develop a range of applications.

The hybrid architecture of UNIWAY allows for standard protocols between different network components and ensures high independence and interoperability, which better caters to sophisticated communications. Also, in the Mobile Internet Era, UNIWAY brings about efficiency and unparalleled cost advantages for developers by optimizing R&D and integrating an array of data, voice, video and other applications.



VoIP



Hybrid



IP2Tel
Tel2IP
Route



Carrier
Selection



USSD



Cloud



VLAN



SMS



VPN



QoS

Key Features and Benefits

- Flexible configuration for any network

Compliant with diverse networks (FXO, FXS, T1, E1, J1, GSM, CDMA, 3G, VoLTE, IP): support various multimedia processing capability (conferencing, fax, compression and SuPerFormTM echo cancellation for voice enhancement and an array of Protocols (SS7, SIGTRAN, ISDN PRI, CAS, R1, R2, Wireless)

- Compliant with any IP-Based applications

With optional inbuilt industrial server, UNIWAY series are compliant with any IP-based applications; it also even supports any category of third party software, including UC, IP-PBX, Contact Center and more. In legacy PSTN network, UNIWAY could converge applications via internal modules.

- Low to High Scalability

Modular architecture ensures flexibility and expandability from low density and high density. Modular design allows for easy configurations, system upgrading or general maintenance

- Multimedia Convergence

Adopt 1000M-Ethernet switching chipset, UNIWAY's media stream exchanges in IP packets, and access to soft switching system via Media Gateway Controller, ensuring high-level applications are streamlined

- Diverse Media Resources

Support high-capacity voice playback and Codecs, conferencing, faxing; Support T.38/T.30; optimized for IP-PBX, IVR and ACD applications, with EXT IVR server or GUI management.

- Carrier-Grade Reliability

Special power system with standby redundancy; advanced cooling system to reassure long-standing robustness ; special air cleaner to protect against dust accumulation inside chassis; Inside temperature control and alert system; No need to change wiring when changing functional modules.

Technical Specification:		UNIWAY2000/UNIWAY2100		
Features	Model			
Functional module available:	UMG-1016: 16*FXO, 16*FXS, or hybrid 8*FXS+8*FXO	UMG-2012: 1, 2, or 4E1(T1)	UMG-4008: 4 or 8 Wireless Ports (GSM/CDMA/WCDMA/3G/4G)	
Voice Processing Codecs	A-law, μ -law, PCM8, PCM16, IAM-ADPCM, VOX, MP3, GSM, G.729A/B, G.722, G.723, iLBC etc;			
Signaling interface	SS7 supports MTP, TUP and ISUP, SCCP, TCAP, MAP			
SIP Protocol	Supported SIP standards: IETF RFC 3261 (SIP: Session Initiation Protocol); IETF RFC 2327 (SDP-Session Description Protocol); IETF RFC 3550 and 3551 (RTP/RTCP); IETF RFC 2833 (DTMF)			
Network Interface	E1/T1/J1/FXS/FXO 2 *TCP/IP 1,000M Ethernet (RJ-45); 2 *LAN Ethernet (RJ-45); 4 USB ports;			
Development Environment	Windows OS: Windows2000/XP/2003/Vista/NT; Linux OS: Including RH7.2/RH9.0/AS4/FC4/SUSE10; Programming language: ANSI C/C++, Microsoft Visual C++, C#, Delphi			
Security and Certifications	Lighting-proof grade: Level 4; Certification: FC/CE/China CCC For RoHS compliance, please contact Synway's sales representatives;			
Physical Characteristics	Dimensions: 2U form factor: 88.1mm (H) x 482.6mm (W) x 430mm (L) Net Weight: about 8Kg (different for the number of optional modules)			
Power Requirement	AC: 90-120V or 200~265V (SELECT RIGHT RANGE ON SHELF), Frequency: 50~60Hz; Power consumption: different for configuration, less than 350Watt;			
Environment Requirement	Ventilation: normal; Operating temperature: 0°C-40°C; Relative humidity: 10%-85%; Avoid dust accumulation; Anti-electrostatic: please Grounded; Installation recommendation: mounted on standard 19-inches rack;			
Routing	Call Routing and translation (PCM \leftrightarrow IP)			
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			

SBC500

Next-Generation Session Border Controller for Enterprises, SPs & Carriers

- Dedicated DSP and Processor Chipsets for 0~500Chs Transcoding
- Decades of Field-Proven Technologies for Highest Level of Security
- Standardize Diverse Proprietary SIP protocols in Global Networks
- Support Intelligent Network Monitoring, Management & Diagnosis

With the development of VoIP technology, VoIP communication applications for enterprise, service providers and carriers become ubiquitous. To empower them to harness and exploit IP-based communication networks efficiently and effortlessly, Synway's new generation session border controller, SBC500, redefines and safeguards smooth interoperability among diverse multimedia applications, complete standardization among different IP protocols and the highest level of security in any networks..

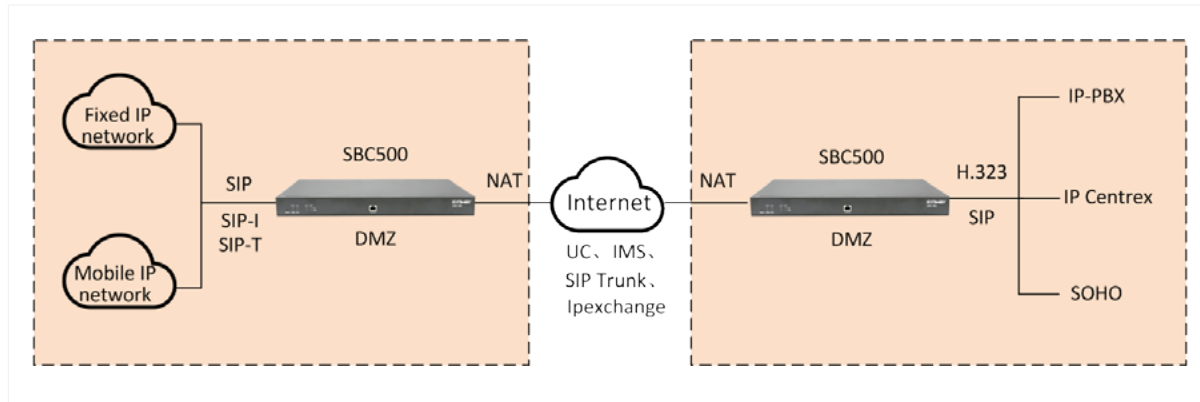
SBC(Session Border Controller), an organic component of communication solution, often is implemented as connection point between internal and external networks, being used for bridging IP-based multimedia transmission among different IP network and ensuring communication security in networks of enterprises, services providers and carriers.



Key Features and Benefits

- Basic Features and Functions
 - 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
- Flexible Scalability for Enterprises, SPs, Carriers
1-500 simultaneous SIP sessions with dedicated DSP-powered multimedia transcoding provide high performance in a small footprint to help lower ownership cost and reassure high security at borders.
- Ensure Highest Level of Operational Security
SBC500 offers real-time communication surveillance solution for enterprises/SPs/carriers communication systems, and helps them deliver any business-driven applications effortlessly and safely.
- Deliver Hi-Flexibility for Session Management
Secure connectivity and controlling can be offered by SBC500, as it efficiently supports bilateral and multilateral interconnection among different networks; besides, Dialing and IP/PSTN routing available.
- Improve Communication Integration Capability
SBC bridges operators with enterprise communication networks seamlessly, rationalizes traffic, streamlines business procedures, and optimizes SIP trunking, PBX/H.323/SIP and VoIP applications.
- Balance Resources and Optimize Loading
Load protection at the border of network provides highly differentiated voice and multimedia services through multi-level resource-balancing policies, registration limit, ID authentication and authorization.
- Integrated Transcoding for Voice & Video
Eliminate need to add separate hardware to support transcoding requirements, and support a range of Codecs, including G.723, G.729, G.711, iLBC, SIPINFO, RFC2833, RF3261, INBOUND and more.

Typical Applications:



- Improve communication security at the border between internal and external networks

Technical Specification:

Features	Model	SBC500				
Sessions		30	60	120	240	480
Input/output Interface		Amount: 2(10/100/1000 BASE-TX(RJ-45)) Self-adaptive bandwidth: Supported Auto MDI/MDIX: Supported				
Console Port		Amount:1(RS-232) Baud rate:115,200bps Connector:RJ45/Customizable Data bits:8 bits Stop bit:1 bit Parity:Unsupported Flow control:Unsupported				
Signaling Protocols		SIP:SIP V1.0/2.0, RFC3261				
Audio Encoding & Decoding		G.711A 64 kbps G.711U 64 kbps G.729 8 kbps G723 5.3/6.3 kbps G722 64 kbps AMR 4.75/5.15/5.90/6.70/7.40/7.95/10.20/12.20 kbps iLBC 15.2 kbps				
Dimensions		L×H×W=440mm*44mm*267 mm				
Net Weight		3.1kg				
Environment		Operating temperature:0°C-45°C Storage temperature:-20°C-85°C Operating Humidity:8%-90% non-condensing Storage humidity:8%-90% non-condensing				
Power Requirements		Input voltage:100-240V AC Maximum power consumption:≤22W				
Sampling Rate		8 kHz				
Safety		Lightning resistance:Level 4				

SBO500 Transcoding Gateway

- Support Complete DSP-enabled Transcoding Range
- Highly Efficient Coding & Decoding IP/TDM Network
- Support 500-Port of Transcoding Capability per Unit

Synway's new family SR500 series call classification equipment is vital for all high density OBD applications, including predictive dialing, dialer, telemarketing and more. It helps minimize dialing cost and maximize value of dialing (human and network) resources. With call classification, OBD applications could improve dialing efficiency by up to 60% and deliver high satisfaction for both agents and subscribers.



Key Features and Benefits

- For Best Dialing Efficiency

With high-accuracy call classification features, SR500 (500Chs) enables solution providers to maximize efficiency and minimize OPEX in a variety of outbound dialing systems, including Predictive Dialing, OBD, Call Center, Telemarketing, etc.

- Detect Any Phone Status for Best Efficiency

To maximize dialing efficiency and deliver intelligent system, call classification functionality helps dialing customers detect and lock on valid calling numbers for hi-accuracy and hi-density dialing, via avoiding “invalid” numbers in “shutdown, suspended, unused number or any unavailable” status.

- Improve Dialing Efficiency by Up to 60%

In general database of OBD systems, unused numbers accounts for 10~ 30%, suspended numbers for 5~ 20%, shutdown number for 5~10%. It means around 20~ 60% dialing numbers are invalid. With call classification function, dialing systems (predictive dialer) could filter out all invalid numbers for best dialing results.

- Reduce Operational Expenditure (OPEX) by 20%

Unsuccessful dialing incurs higher operational costs while labor cost for telemarketing is specifically considered. According to analytical statistics from surveys, OPEX could be reduced by 20% or more if successful dialing ratio increases by 60%.

- 99.999% Accuracy of Call Classification

With call classification functionality, over 99.999% of called number states could be accurately detected by analyzing telephone alert tones. Only in much noise conditions over the network, call classification may default.

- High-efficiency Decoding & Encoding Function

With the popularity of IP communications, increasing need for transcoding between various protocols and coding formats becomes more important. Large-capacity VoIP applications call for high efficient transcoding functionality. High-efficiency decoding & encoding function can be offered by SR500.

WEB Management Interface

User-friendly Configuration Toolkit

- Simplest Way to Resolve Most Sophisticated Requirements

With complete understanding about enterprise and carrier's headaches, Synway's SMG VoIP Gateway adopts the simplest and straight-forward configuration to achieve users' most sophisticated objectives. With this configuration toolkit, IT managers from enterprises and carriers could control VoIP gateways effortlessly and efficiently, despite of their limited knowledge about VoIP gateways. With minimum time and expenditure, enterprises and carriers gain maximum values.

- User-friendly WEB Management

User-friendly WEB management tool has been built into Synway SMG series gateway, and users could log in their account to manage all synway gateways via web and effortlessly monitor real-time operation status, system information, PSTN line status and other data and information, which helps users to save the cost on network management and maintenance.



Figure 1: Basic data display

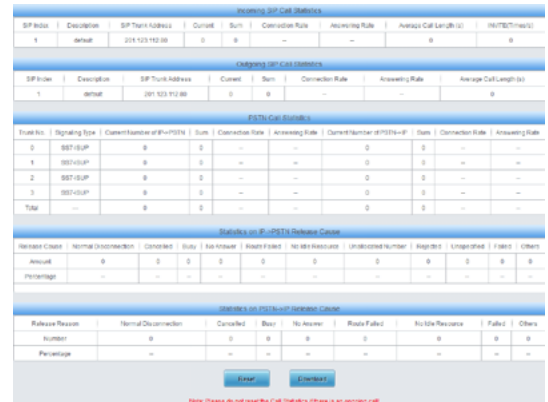


Figure 2: call statistics

- Flexible VoIP Configuration

Five types of VoIP setting are available, including SIP Setting, SIP Trunking setting, SIP account setting, SIP trunk group setting and media setting.

- Compliance with a complete range of protocols

Synway SMG series gateway seamlessly complies with various protocols. Fitting to a variety of communication environments, SMG family could configure PCM parameters via web-based management software. The PCM Settings includes five categories: PSTN, PCM, PCM trunk group, calling-reception rule, and reception timeout.

Worthy of notice, users could set up SS7 parameter in the menu only when the configuration item "Signaling Protocol" in the PCM settings is set to SS7-TUP or SS7-ISUP. SS7 Settings includes eight ones: SS7, TUP, TUP Number Param, ISUP, Number Param, Original CalleeID Pool, Redirecting Number Pool (Hidden item) and SS7 Server.

ISDN setting is available in the menu only when the configuration item "Signaling Protocol" on the PCM settings interface is set to ISDN User Side or ISDN Network Side. Moreover, SMG web management software also could set other parameters, such as fax, routing, number filtering, and number manipulation and so on.

- Simple Configuration for Media Processing

With the popularity of IP communications, increasing need for transcoding and media processing between various protocols and coding formats becomes more significant.

DCMS

Device Cloud-Based Management System

- Real-Time Monitoring of VoIP Gateway Status
- Management Capability for More Than 10,000 Devices
- DCMS Features Cloud-based Management Capability
- Superior Network Access Traversal (NAT) Capability

Synway Device Cloud-Based Management System (DCMS), based on the Internet, is used for monitoring, maintenance and debugging for Synway's VoIP gateways on the field. It helps users leverage Synway's products efficiently and unfailingly, as well as improves controllability and manageability over entire communication systems.

Device ID	Serial No. #	Device Name #	Device Status #	Line #
000002872		SA/02016	OK	0%
000001607		SA/02120	OK	0%
000002330		SA/01032	OK	0%
000001440		SA/03016	OK	2%
000001842		SA/01032	OK	0%
000022087		SA/03016	OK	0%
000000062		SA/02120	OK	0%
000001300		SA/01032	OK	0%
000022087		SA/03016	OK	0%
3034		SA/01032-BO	OK	0%

Serial No. #	Election Name #	Election Name #	Status #	Location Line #	Set Name #	Software Version #	Election #	Election
000001000	000001000	000001000	OK	000001000	000001000	000001000	000001000	000001000
000001001	000001001	000001001	OK	000001001	000001001	000001001	000001001	000001001
000001002	000001002	000001002	OK	000001002	000001002	000001002	000001002	000001002
000001003	000001003	000001003	OK	000001003	000001003	000001003	000001003	000001003
000001004	000001004	000001004	OK	000001004	000001004	000001004	000001004	000001004
000001005	000001005	000001005	OK	000001005	000001005	000001005	000001005	000001005
000001006	000001006	000001006	OK	000001006	000001006	000001006	000001006	000001006
000001007	000001007	000001007	OK	000001007	000001007	000001007	000001007	000001007
000001008	000001008	000001008	OK	000001008	000001008	000001008	000001008	000001008
000001009	000001009	000001009	OK	000001009	000001009	000001009	000001009	000001009

Key Features and Benefits

- Real-Time Monitoring of VoIP Gateway Status
DCMS is a unified management platform developed by Synway to manage all Synway's VoIP gateways remotely, providing centralized monitoring and management over Synway VoIP gateways in the networks.
- Management Capability for More Than 10,000 Devices
DCMS could monitor over 10,000 SMG series VoIP gateways simultaneously, but it just consumes less than 10% CPU processing capability for registration and connectivity procedures.
- DCMS Features Cloud-based Management Capability
Two types of cloud-based management modes are delivered by DCMS: public cloud mode managing devices online through public cloud platform, and private cloud mode in which DCMS users could implement Synway's dedicated software for devices management.
- Superior Network Access Traversal (NAT) Capability
DCMS possesses automatic NAT traversal capability, and need not configure port mirroring setting while building up connection between intranet and internet, or between any networks.
- Relevant Allocation of Communication Resources
DCMS could improve operational and maintenance efficiency and optimize communication resources for users. In large-scalability communications systems, DCMS could reasonably and effectively allocate communication resources to better fit to specific application scenarios, helping users build VoIP communication system with minimum CAPEX and OPEX.

Typical Applications

SMG1000

- The Best Cost and Reduce OPEX by 50%

More HW/SW technologies are adopted to solve core technological problems, and which assures system ran efficiently in the enduring complex communication environments.

- Excellent voice quality and telco-grade reliability

DSP-enabled voice optimization to assure crystal voice quality and system agility as well as maximize bandwidth efficiency;

- Eliminate headaches in rival products

The performance and function can be renewed and updated by advanced synway-own technology, which including ensuring voice detection of FXO calling and DTMT accuracy, Comply with most platforms, Plug and Play.

- High-speed response with minimum bandwidth

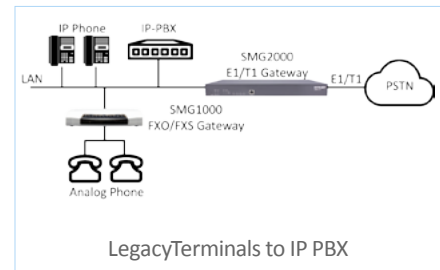
High-speed response and connectivity in the extreme network environments, with better run efficiency;

- User-friendly GUI and central administration

Easy-to-use service Web based UI configuration and management tools could help accelerate service deployment and simplify platform management.

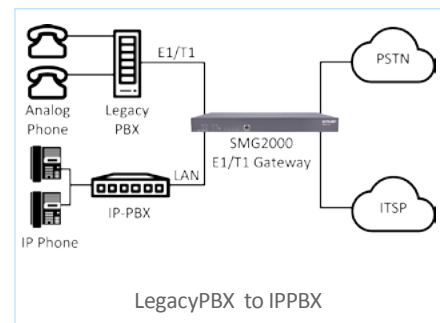
- Customizability Satisfies all Specific Needs

Homegrown inherited soft-hard technology to renew and update performance and functionalities. More compact hardware architecture saves space and assures unparalleled power consumption and cost;



SMG2000

- Support and interface with any applications and platforms based on E1/T1 networks for carrier, SPs, enterprise;
- User-friendly configuration, GUI, centralized administration, customizable features and functionalities;
- DSP-enabled voice optimization to assure crystal voice quality and maximize bandwidth efficiency and reliability;
- High-speed response and connectivity in the extreme network environments, with better run efficiency;
- Telecom grade reliability and continuous high performance in fully loaded capacity and in the long run;
- Homegrown core technologies, Mini-sized footprint assure unparalleled cost and user convenience for enterprises and SPs;



SMG3000

- Build-in SS7, SIGTRAN, SIP, ISDN, R2

Complimentary standardized SS7 Packets (Up to 64 SS7 links, ISUP, MTP1~3, TCAP), SIGTRAN, and SIP for any carrier networks worldwide; Field-proven by over 50 largest operators from European, China, USA, India, Brazil, South America;

- Maximum CPS (Calling Per Second)

Telecom-style DSP algorithm has been optimized for over decades, assuring seamless compliance with any network environment. Plentiful DSP resources are allocated for maximum calling per second (CPS), signaling, media processing, bandwidth optimization, Telco redundancy;

- 99.9999% Availability in Full Load

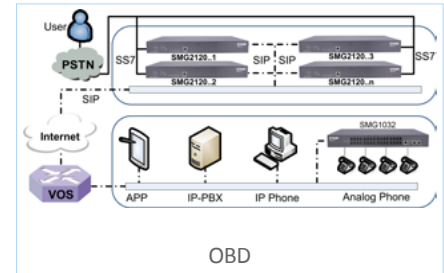
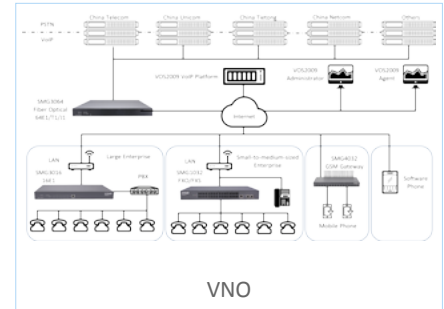
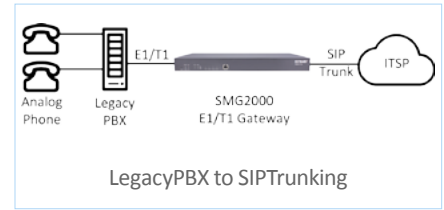
Support 1/2/4/8/16/64E1 (30/60/120/240/480/1,920Chs) per unit, with higher system responsiveness capability in extreme network for better operational results; High performance and stability in cases of unstable (low) bandwidth and high capacity;

- User-friendly Cloud-based Management

With complete understanding about SPs and operator's headaches, SMG3000 adopts the simplest and straight-forward configuration to achieve SPs and Operators' most sophisticated objectives. Over 1,000 units of SMG3000 can run together; User-friendly configuration, GUI, centralized administration, customizable features and functionalities;

- Robust Telco-Grade Architecture

Dual power supply, 1U compact structure, industrial heating sink system, dual 1,000M Ethernet interfaces, and Telco' grade hardware architecture ensure 200,000Hr MTBF and 0-defective rate in long lifecycle;



SMG4000

- Support a Range of System

Could be applied into a range of systems, including, mobility PBX, PSTN backup lines, VoIP localization, SMS notification platform, cloud-based services, contact service center, and GSM/CDMA Trunking.

- Seamless Interoperability

Compliable with a variety of wireless/mobile protocols (2G/3G/4G), enabling interconnection between GSM/CDMA/WCDMA/VoLTE network and VoIP network smoothly and safely.

- Higher Business Values

Integrated "Gateway Plus Built-in SIM Bank" architecture for better communication efficiency. Mobile SMS-based commercial delivers advantages of low cost, dismiss complexity, and also save time and human cost

- Perfectly Combine Voice & Data

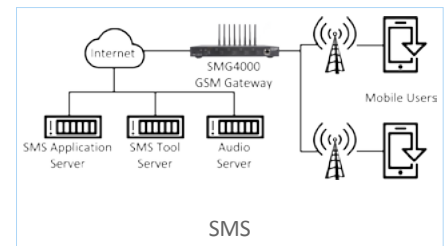
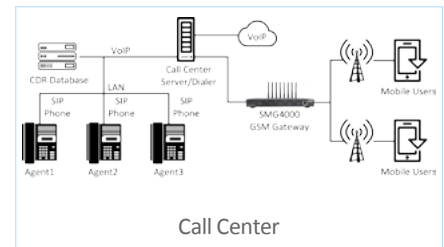
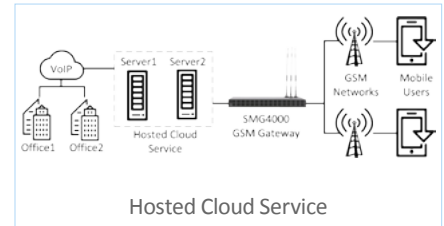
DSP-enabled voice optimization to assure crystal voice quality and maximize bandwidth efficiency; separately process voice or data transmission, or handle simultaneous voice and data transmissions in a single system.

- Hi-Speed & Accuracy Processing

Field-approved in diverse extreme environments, including low bandwidth and unstable wireless signals. High-speed response and connectivity in the extreme network environments, with better run efficiency.

- Boost Operational Efficiency

Significantly improve operation efficiency while applying SMG4000 in the data processing & SMS transmission. Telecom grade reliability and continuous high performance in fully loaded capacity and in the long run; Homegrown hardware and software technologies to guarantee relevant upgrading and customization.





Since
1992



CTI and
VoIP Gateway



1000+
Employees



3000+
Partners

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

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

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