

# SMG1000-D32SPR

## FXS VoIP Gateway

- Cost effective gateway with 32 FXS ports
- Fax over IP (T.38 and Pass-Through)
- Support IPv4 and IPv6
- TR069 and SNMP
- Multiple codecs: G.711A/U, G.723.1, G.729A/B, AMR, etc.
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

SMG1000-D32SPR Analog Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks. The device offers 32 FXS ports, fax over IP and flexible dial plans. It is ideally suited for small and medium businesses, call centers and multi-location environments that need VoIP services. SMG1000-D32SPR supports the standard SIP protocol and it's compatible with leading IMS/NGN platforms and SIP- based IP telephony systems.



VoIP



TR-069



FAX



Hotline



NAT



Cloud



VLAN



Recording



VPN



QoS

### 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 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  $\mu$ -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:

### Physical Interfaces

Capacity: 32 FXS with RJ11, RJ21  
Ethernet Interfaces: 4\* LAN, 10/100Mbps, RJ-45  
Console: 1\*RS232, 115200bps  
Dimensions(W/D/H): 440\*267\*44mm  
Unit Weight: about 4kg

### Voice & FAX

G.711A/U law, G.723.1, G.729 A/B, AMR, iLBC  
Silence Suppression  
Comfort Noise Generation(CNG) Voice Activity  
Detection(VAD)  
Echo Cancellation(G.168), with up to 128ms  
Adaptive (Dynamic) Jitter Buffer Hook Flash  
Programmable Gain Control  
T.38/Pass-through  
High speed fax up to 14.4kbps Modem/POS  
DTMF mode: Signal/RFC2833/INBAND  
VLAN 802.1P/802.1Q  
(data/voice/management VLANs) Layer3 QoS and DiffServ

### Supplement Service

Call Waiting Blind Transfer Attend Transfer  
Call Forward on Busy  
Call Forward on No Reply Unconditional Call Forward  
Warm/Immediately Hotline  
Call Hold  
Do-not-disturb  
3-Way Conference Message Waiting Indicator

### FXS

Connector: RJ11, RJ21  
Dial Mode: DTMF and Pulse  
Caller ID: DTMF/FSK CLI Presentation Max Cable Length: 5  
km  
Reversed Polarity  
Programmable Call Progress Tone

### Software Features

Hunting Group Web ACL  
PPPoE/IPv4/IPv6  
Digitmap  
Routing Rules based Prefixes Caller/Called Number  
Manipulation

### Maintenance

SNMP v1/v2/v3  
TR069  
Auto Provisioning Web/Telnet  
Configuration Backup/Restore  
Firmware Upgrade via Web CDR  
Syslog(Emerg,alert, critical,error warning,notice,info, debug)  
Ping/Tracert Test Network Capture  
NTP/Daylight Saving Time  
Cloud-based Management

### VoIP

Protocol:  
SIP v2.0 (UDP/TCP),RFC3261 SDP,RTP(RFC2833), RFC3262,  
3263,3264,3265,3515,2976,3311  
SIP TLS/SRTP  
RTP/RTCP, RFC2198, 1889  
RFC4028 Session Timer  
RFC3581 NAT,rport Primary/backup SIP server Outbound  
Proxy  
DNS SRV/ A Query  
SIP Trunk  
Early Media/Early Answer  
NAT:STUN, Static/Dynamic NAT

### Environmental

Power Supply: 100-240VAC, 50-60 Hz Power  
Consumption:40W(Typical) Operating Temperature:0 °C ~  
45 °C  
Storage Temperature: -20 °C ~85 °C  
Humidity:10%-90% Non-Condensing

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**HANGZHOU SYNWAY INFORMATION ENGINEERING CO., LTD.**

Founded in 1995 in Hangzhou, SYNWAY is a leading global provider of IP Unified Communication products including VoIP Gateways, IP PBXs and SBCs, We have been delivering more agile, efficient and affordable communication solutions and unparalleled communication experiences to our customers with our reliable, innovative and future-proof products for years. Through our value-added distributors and resellers worldwide, now SYNWAY serves telecom operators, service providers, system integrators, enterprises, SMBs and OEM partners in over 100 countries.