SMG1000-D40

Cost-Effective and Reliable 4*FXO Analog VoIP Gateway

- **Support SIP and Analog Protocols**
- **Support 4*FXO in Mini-Sized Device**
- **Superior Voice Quality and Rich Voice Resource**
- **High Price-Performance Ratio**

With a simple and economical way to help legacy telephone, fax machine and PBXs interconnect with IP network, Synway's SMG1000-D4O 4-ports analog FXO gateway enables call center and multi-branch enterprises to process powerful, versatile and efficient VoIP outbound dialing solutions with unparalleled cost advantages. Connected between a PBX, LAN or WAN, the 4 ports analog FXO VoIP Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks.

Designed for voicemail and unified messaging applications, the 4 Ports Analog VoIP Gateway SMG1000-D4O has a 10/100 (optional) Base-T Ethernet connection for connecting legacy PBX to a LAN. The analog loop start functionality supports integration via in-band signaling (DTMF or FSK), serial protocols, as well as T.38 for fax transmissions over IP (FoIP).

Highlights

Key Benefits

- High performance VoIP connectivity for SMBs
- Voice optimization to ensure better user experiences
- Enhanced call routing ability with high voice quality
- Easy to install, configure, and maintain
- Support IPv4 and IPv6 international network
- Data/voice/management VLAN and more

- Build-in firewall and access rules
- Support SNMP/TR069/Auto-Provision
- Cloud-based management and bandwidth optimization
- Support SIP, MGCP or other customizable protocols
- Primary/Backup SIP Servers
- Flexible routing and manipulation

Key Features

- Completely non-blocking architecture and Scalable System
- Easy integration with existing telephony interfaces
- Open-standard SIP support and register to multiple SIP proxy servers
- Make and receive IP calls from analog extensions
- Call budgeting based on allocated amount, minutes and call count
- Manageable based call routing TDP-IP/IP-TDM
- Restrict unwanted calls with list of denied numbers
- Real-time call record send to CDR server
- Caller ID presentation and restriction
- Hotline extension setting
- Web-based remote administration
- Consol access via Telnet, SSH.

Core Specifications

Product Model	SMG1000-D4O
Physical Interface	Phone Interface: 4 Ports FXS/FXO, RJ-11 Ethernet Interface: 2* RJ-45 10/1000Mbps Base-T Ethernet, Female RJ-45
Session Capacity	4 SIP channels 4* FXO channels
Connectivity	Supported Dial Mode: DTMF and Pulse Pulse: 10 and 20PPS Caller ID:DTMF/FSK Max Cable Length:5KM Reversed Polarity OpenVPN
IP protocols	TLS / SRTP ;OpenVpn;SIP V2.0 (RFC 3261,3262,3264);IMS/3GPP; SDP ;REFER (RFC 3515);RTP/RTCP;STUN (RFC3489) ;ARP/RARP (RFC 826/903);SNTP (RFC 2030);DHCP/PPPoE； TFTP/HTTP/HTTPS;DNS/DNSSRV (RFC 1706/RFC2782);VLAN802.1P/802.1Q
Call & Routing	T.38/Pass-through, up to 14.4kbpsPort Groups IP Trunks Primary and Secondary SIP Account 4 Inbound/Outbound Routing Number Manipulation Digit maps TDM to IP or IP to TDM IP load balancing IP fault tolerance

Voice Capability	G.711A/U law, G.723.1, G.729A/B, AMR Comfort Noise Generation(CNG) Echo Cancellation(G.168) DTMF mode: Signal/RFC2833/INBAND Silence suppression with comfort noise G.168 automatic echo cancellation Call Progress Analysis (CPA), including Positive Voice Detection, Positive Answering Machine Detection (PAMD), DTMF detection, and fax tone detection Manageable based call routing TDP-IP/IP-TDM. Restrict unwanted calls with list of denied numbers. Voice Activity Detection (VAD) Adaptive (Dynamic) Jitter Buffer Programmable Gain Control Hook Flash
FoIP Protocol & Faxing	Static IP, PPPoE, DHCP ClientT.38 for transmission over a packet network T.38/Pass-through, up to 14.4kbps T.38 FoIP: transcode fax from T.30 fax protocol (supporting V.17) modulation schemes
Network Capability	Static IP, PPPoE, DHCP Client IPv4, IPv6 Static/dynamic ARP DIFFServ, ToS NAT (Rout and Bridge)+ MAC Address Clone Static routing+ Built-in Firewalls QoS, Traffic Shaping Voice/Data/Management Vlan
Maintenance & Upgrading	SNMP/TR069. Auto Provision Action URL Digit map Web/Telnet. ACL Configuration Backup/Restore Bandwidth Optimization Routing Rules based Prefixes Firmware Upgrade via WEB Syslog and CDR. Access Rule list. Network Capture Outward Test(GR909). Automatic Time Synchronization IVR local Maintenance. Cloud-based Management Caller/Called Number Manipulation Open-standard SIP support and register to multiple SIP proxy servers.
Application Capabilities	Supported Dial Mode: DTMF and PulseCall waiting Blind Transfer Attend Transfer Call forward on Busy Call forward on No Reply

	Unconditional Call Forward HotlineCall hold DND Call Pickup 3-way conference Voicemail
Conferencing Resource	Call budgeting based on allocated amount, minutes and call count Complete non-blocking architecture and Scalable System Hotline extension setting Support 3-Way and Multi-Way Conferencing
Environment & Power	Power Supply: 100-240V, 50-60Hz+ Power Consumption: Approximately 50W (Storage): -20 ~85°C Humidity: 10%-90% No condensation. Operating temperature range: -10 ℃ ~ 55 ℃
Physical Dimension	L*W*H 140(mm)*110(mm)*30(mm) Weight Approximately 0.44ibs(about 0.2kg)
Warranty/Certifications	3 years: The first year exchange for free. On the Second & Third free to repair. CE, FCC or Any other Certificates Customizable
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
Dedicated DSP-Empowered Capability	Assuring seamless compliance with any network environment. Plentiful DSP resources are allocated for signaling, media processing, bandwidth optimization, Telco redundancy Telecom-style DSP algorithm has been optimized for over decades
Highly Adjustable For Diverse SoftSwitch	Homegrown core technologies to assure seamless compliance with diverse softswitch platform Including Mitel, Avaya, Broadsoft, Yate,OpenSIP, Asterisk, VECTRA, VSC, SIPPULSE, Tropico, FreeSwitch and more other softswitch

About Synway

As a leading Security and VoIP enabling-technologies provider in China, Synway has been partnered with applications & solution providers worldwide to deliver turkey solutions for enterprises, telecom carriers, intelligence, public security, law enforcement, etc. Based on long-standing business network, Synway's products and service have served 3,000+ customers, including contact centers, financial institutes, public security, national safety agencies and more.

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