SMG 1000-D32S FXS VoIP Gateway

With a simple and economical way to help legacy telephone, fax machine and PBXs interconnect with IP network, Synway's 32 ports analog FXS gateway enables call center and multi-branch enterprises to process powerful, versatile and efficient VoIP solutions with unparalleled cost advantages. Connected between a PBX, LAN or WAN, the 32 ports FXS VoIP Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks.

Designed for voicemail and unified messaging applications, the 32 Ports Analog VoIP Gateway SMG1000-D32S has a 10/100/1000M (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

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.

SMG 1000-D32S
FXS VoIP Gateway

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

Technical Specifications:

- *Physical Interface*

Phone Interface: 32 Ports FXS, RJ-11 available as well

Ethernet Interface: 2* RJ-45 10/100Mbps Base-T Ethernet, Female RJ-45

1000M LAN/WAN available for some product models while required

- *Session Capacity*

32 SIP channels (SMG1000-D32S)

32 FXS channels (SMG1000-D32S)

- *Connectivity*

Dial Mode: DTMF and Pulse

Pulse: 10 and 20PPS

Caller ID:DTMF/FSK

Max Cable Length:5KM

Reversed Polarity

OpenVPN

- *VoIP 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*

Port Groups

IP Trunks

Primary and Secondary SIP Account

32 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,G.726,iLBC,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*

T.38 for transmission over a packet network

T.38/Pass-through, up to 14.4kbps

Synway®

SMG1000-D32S
FXS VoIP Gateway

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*

Call 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

Temperature(Operation):0 °C ~ 45°C

(Storage): -20 ~85°C

Humidity: 10%-90% No condensation.

Operating temperature range: -10 °C ~ 55°C

- *Physical Dimension*

L*W*H 440(mm)*202(mm)*44(mm)

Weight Approximately 5.95ibs(about 2.7kg)

- *Warranty/Certifications*

3 years: The first year exchange for free. On the Second & Third free to repair.

CE, FCC or Any other Certificates Customizable

Broadsoft, Elastix, Asterisk, Teams and other UC platform