VG-1404SP

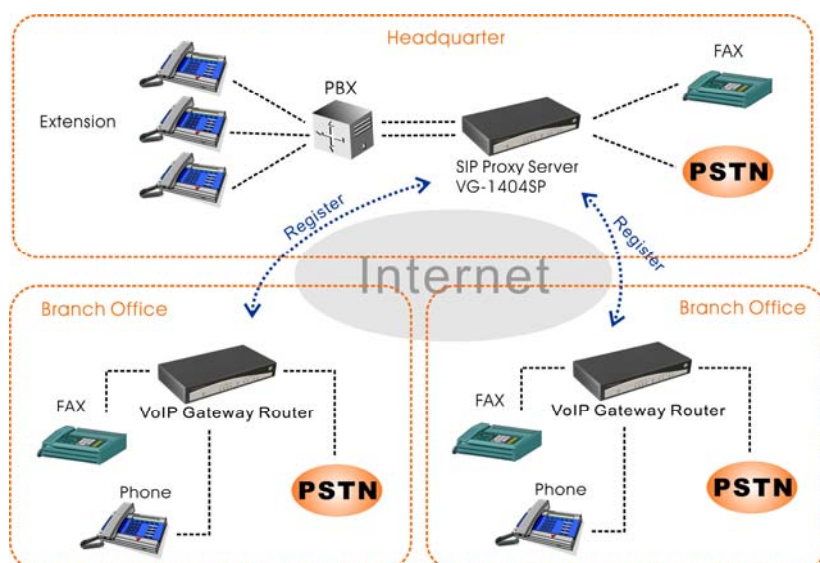SIP Proxy Server with VoIP Gateway Router

(4-Port RJ-11 FXS + 1 WAN Port)

As an increasing number of carrier deployment and enterprise solutions, the need for direct interconnection between these VoIP networks is becoming more demanding. The PHEENET VG-1404SP gives enterprises and system integrators a standalone SIP 2.0 (RFC3261) compliant proxy server for VoIP networks.

Based on the Session Initiation Protocol (SIP), the VG-1404SP SIP Proxy provides a MD5 authentication, call routing capabilities to maximize network performance in both corporation and Internet voice networks.

The VG-1404SP provides the SIP client registration, authentication, and administration for all the SIP compliant endpoints in your network, including gateways, IP phones and PC clients.

The PHEENET VG-1404SP provides the NAT transversal capability to serve both the public and private network environments. The product detects and configures the network address information of calling parties.

APPLICATION

Features

SIP Proxy Server Key Features:

1. 250 SIP endpoints scale: providing 250 SIP endpoints to register
2. Register Security Policy: providing MD5 authentication setting on your VoIP Network.
3. Real time call detail records and post call detail records report: Support Real Time CDR to monitor VoIP calls, including caller IP, callee IP, call date, call duration and other information. Also providing a CDR report to look up VoIP call record.
4. Top 20 list: it can list top 20 calls by call duration, caller number, calling number, caller IP or callee IP address.
5. Syslog: send syslog information to Syslog server.

VoIP Key Features:

6. Both support H.323 and SIP protocols: H.323 Gatekeeper Registration, SIP Registration, and Digest Authentication
7. Flash Time Programmable: Gateway support Flash detection and generation to switch VoIP Call
8. Smart VoIP Call Dialing Book
9. Smart QoS Guarantee: This bandwidth management feature provides good voice quality when user place VoIP call and access internet at the same time. The Gateway will start reserve bandwidth for voice traffic automatically when VoIP call proceeds
10. Voice Channels Status Display: this function display each port status like as on hook, off hook, calling number, caller's number, talk duration, codec
11. Domain name call & DDNS supports
12. User friendly management

Specification

Industrial Standard	
Standard	IEEE802.3 10BaseT Ethernet
	IEEE802.3u 100BaseTX Fast Ethernet
	SIP (RFC2543/3261), SDP (RFC2327)
Interface	LAN:4×10/100Base-TX Ethernet port
	Voice: 4×RJ-11 (2-pin modular jacks) for FXS
	WAN:1×10/100Base-TX Ethernet port
VoIP Specification	
Voice Codec Support	G.711A/µ-law
	G.723.1
	G.729A/B
Fax Support	T.30 real-time Fax complaint
	T.38 Fax relay
Simultaneous Connection	Up to 4 channels voice/fax
Voice Processing	Silence Suppression: VAD, CNG

	Echo Cancellation: G.168
	FXO Caller ID Detection: DTMF and FSK
	Dynamic Jitter Buffer
	Auto Dial for Speed
	Silence Compression
Protocol	LAN: NAT, Virtual Server, DHCP Server
	WAN: PPPoE Client, DHCP Client, Fix IP Address, DDNS Client
Management	Web-based configuration
	FTP Firmware Upgrade
	Backup and Restore Configuration file
Smart QoS	Guarantee the voice bandwidth
Hardware	
Environment	Operating Temperature: 0 to 45° C
	Storage Temperature: -10 to 60° C
	Humidity: up to 90% non-condensing
Power Supply	AC 100V-240V, DC12V/1.5A, 50~60Hz
LED Indicators	Power, LAN Link/ACT, Voice Ringing/ACT
Dimensions	260 (L) X 130(W) x 35(H) mm
Certificate	FCC, CE