VGW-804

8-Port SIP VoIP Gateway (4 FXS + 4 FXO)

Cost-effective, High-performance VoIP Communication

To build high-performance VoIP communications at a low cost, PLANET now introduces the latest member of its gateway family, the VGW-804 enterprise-class 8-port SIP VoIP Gateway. The VGW-804 provides added flexibility during migration to Unified Communications by supporting the traditional analog devices, which include analog phones, fax machines, modems, voicemail systems, and speakerphones. It helps the enterprises to save money on long-distance calls; for example, the remote workers can dial in through a Unified VoIP Communication System just like an extension call but no long-distance call charge would occur. The VGW-804 also allows call to be transferred to anyone at any location within the voice system, which enables the enterprises to communicate more effectively and is helpful to streamline business processes.

SIP Standard Compliance

The VGW-804 supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VGW-804 is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Highlights

- Supports SIP 2.0 (RFC 3261)
- Supports IPv6 and IPv4 simultaneously
- Up to 24 SIP service domains and Caller ID
- Supports auto HTTP provision and fax feature
- Flexible routes plan, dial plan and SIP trunk
- Life-line for emergency calls

Internet Features

- IPv4 (RFC 791) and IPv6
- IPv6 auto configuration (RFC 4862)
- MAC clone setting
- Vendor Class ID
- DDNS (Planet DDNS, Easy DDNS, DynDNS)
- DNS client
- Firewall
- URL / IP / MAC / Port Filter
- Port forwarding (TCP, UDP or both)
- Bandwidth control (download and upload), maximum bandwidth priority setting

SIP Applications

- SIP Session Timer (RFC 8048)
- SIP Session Refresher: UAC or UAS
- SIP Encryption
- Supports Outbound Proxy / STUN NAT Traversal
- Supports Primary and Backup SIP Server

Call Features

- Supports peer to peer dialing
- 4-line FXO connects to PSTN line
- 4-line FXS connects to analog phone set or PABX
- Caller ID recognition DTMF (before/after 1st ring) and FSK (before 1st ring), ETSI and Bellcore
- DTMF Caller ID start and stop bit configurable
- T.38 fax volume configuration

FXO/FXS Line Configuration

- Line ID / Line phone number
- Polarity reversal detection or generation for call establishment and billing
- VoIP dial to FXO/PSTN Line: 1 stage dialing and 2 stage dialing
- Outgoing SIP Caller ID selection

Routing Plan

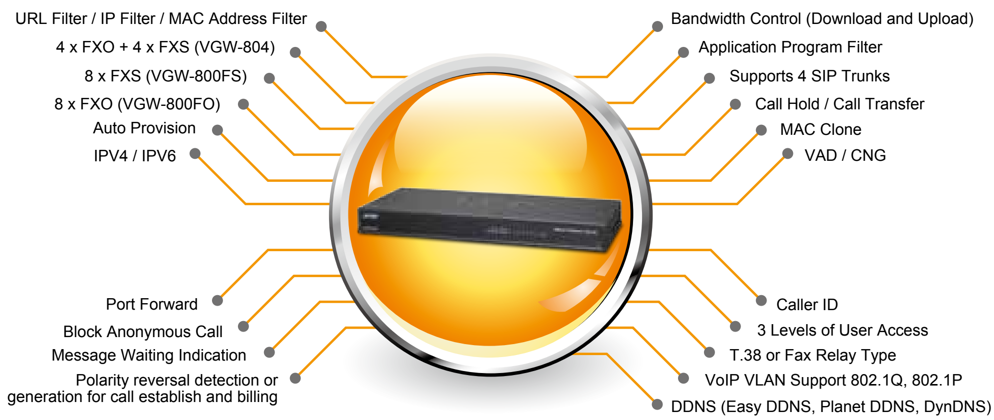
- Prefix match and length
- Priority / Cyclic / Simultaneous ring
- Programmable hunting cycle

Compliant with standard SIP RFC 3261

Enhanced, Full-Featured Business Gateway

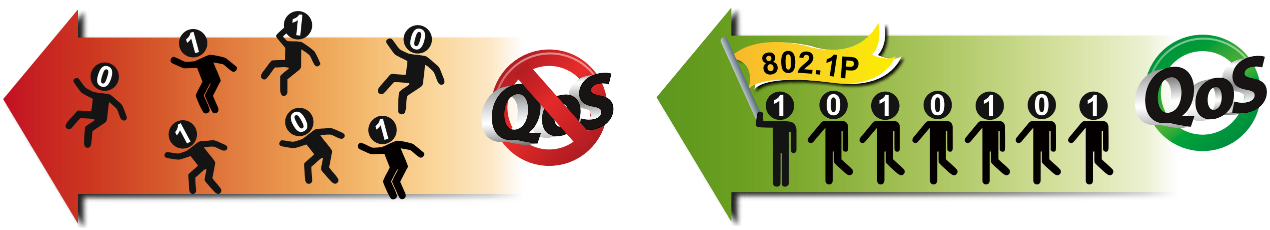
The VGW-804 is a full-featured enhanced business SIP Gateway that addresses the communication needs of the enterprises. It provides 4-line FXO plus 4-line FXS interface which allows connection with 4 analog PSTN telephone lines and with 4-line analog telephone set to make or receive VoIP call over Internet or VPN network. This device is suitable for office PABX to enable to have VoIP call without changing cabling, dial plan and extension number.

The VGW-804 supports all kinds of SIP-based gateway features and multiple contact filter functions, such as 24 SIP trunk accounts, both IPv6 and IPv4 protocols, flexible dial plan and route plan features, and switch of analog and VoIP signal to help both protocols to communicate efficiently.



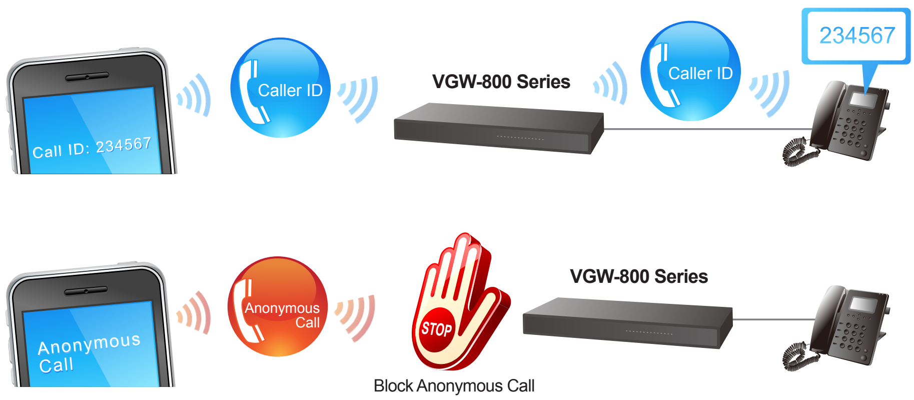
Secure, High-Quality VoIP Communication

The VGW-804 can effortlessly deliver secure toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service), 802.1Q VLAN tagging, and IP TOS (Type of Service) technology. Using voice and data VLAN can easily separate the data and voice, thus maintaining the best quality.



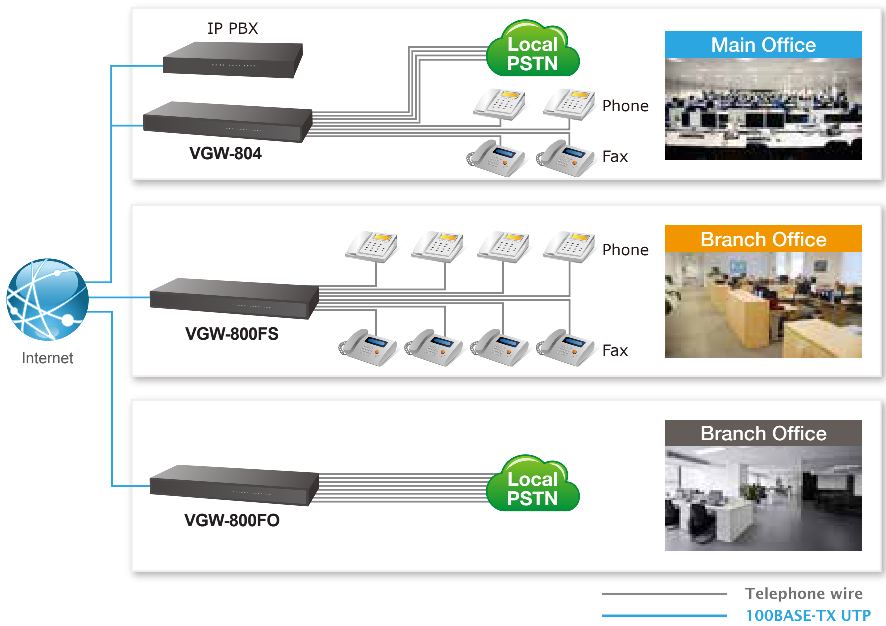
Supporting Caller ID

Both the FXS and FXO ports of the VGW-804 support caller ID function, helping users identify calling number and verify number easily. It also helps to block anonymous call by filtering strange calls. The FXS port transmits Caller ID, while the FXO port receives Caller ID. The Caller ID interoperates with analog phones, public switched telephone networks (PSTN) and private branch exchanges (PBXs).



Applications

The VGW-804 provides the essential features you need for business-class voice communications in an easy-to-manage solution. Designed for businesses with branch offices, it helps the enterprises to save money on long-distance calls.



Specifications

Product	VGW-804
Hardware	
WAN	1 x 10/100BASE-TX RJ45 port
LAN	1 x 10/100BASE-TX RJ45 port
Voice	8 x RJ11 connection (4 x FXS, 4 x FXO)
Protocols and Standard	
Data Networking	IPv4 (RFC 791) and IPv6 IPv6 auto configuration (RFC 4862) IPv6 only, IPv4 only or dual stack MAC address (IEEE 802.3) MAC clone setting Vendor Class ID IP / ICMP / ARP / RARP / SNTP Static IP DHCP Client (RFC 2131), WAN port DHCP Server, LAN port NAT Server (RFC 1631) PPPoE Client / DNS Client / TFTP Client DDNS (Planet DDNS, Easy DDNS, DynDNS) Firewall URL / IP / MAC / Port filter Application program filter Port forwarding (TCP, UDP or both) Bandwidth control (download and upload), maximum bandwidth priority setting UPnP server at LAN port Behind NAT, use DMZ for NAT traversal SNTP with time zone and daylight saving TCP/UDP (RFC 793/768), RTP/RTCP (RFC 1889/1890), IPV4 ICMP (RFC 792) VoIP VLAN supports 802.1Q, 802.1P VLAN ID range: 2 to 4094 VLAN priority: 0 to 7 (highest priority) QoS: DiffServ (RFC 2475), TOS (RFC 791/1394)
Voice Gateway	RFC 3261 compliance Supports up to 24 SIP trunks to register SIP UDP Protocol Supports SIP compact form Supports SIP Hold Type: Send only, 0.0.0.0 or inactive SIP Session Timer (RFC 4028) SIP Session Refresher: UAC or UAS SIP encryption MD5 digest authentication (RFC 2069/2617) Reliability of provision response PRACK (RFC 3262) Early/Delay media support Offer/Answer (RFC 3264) Message Waiting Indication (RFC 3842) Event notification (RFC 3265) REFER (RFC 3515) Supports outbound proxy Supports primary and backup SIP server Supports STUN NAT Traversal Supports “rport” parameter (RFC 3581) Configure SIP local port SIP QoS type: DiffServ or QoS Accept proxy only: Yes or No
Audio Codec	G.711 A-law/μ-law, G.729A, G.723.1 (6.3K, 5.3K) Select voice codec priority: Local or remote Voice payload size (ms) configuration Silence suppression VAD/CNG LEC: Line Echo Canceller Max Echo Tail Length (G.168): 32, 64 and 128ms Packet Loss Compensation Automatic Gain Control In-band / out of band DTMF (RFC 4733, RFC 2833 / SIP INFO) Adaptive/Configurable jitter buffer G.168 Acoustic Echo Cancellation Configure RTP basic port

	RTP QoS type: DiffServ or TOS Phone book (50 records) for peer to peer calls Dialing plan with drop, replace, insert dialing digits Selects first digit and inter digit timeout duration (Sec) Selectable Call Progress Tone Support Specified Line Calling
Functions	
Call Functions	Supports peer to peer dialing 4-line FXO connects to PSTN Line 4-line FXS connects to analog phone set or PABX. Caller ID recognition DTMF (before/after 1st ring) and FSK (before 1st ring), ETSI and Bellcore DTMF Caller ID start and stop bit configurable Current drop detection to release FXO port Disconnect tone recognition to release FXO port Tone Generation: Ring Back, Dial, Busy, Call Waiting, ROH, Warning, Holding, Stutter Dial Tone and Disconnected Tone Configure Tone Frequency, Cadence, Level and Cycle Select Tone specification by country name list Global Country based Tone Specification NAT Traversal supports STUN, UPNP and Behind NAT Out-of-Band DTMF with RFC 2833 and SIP info RFC2833 Payload type: 101 or 96 DTMF send out ON and OFF time configuration DTMF incoming recognition minimum ON and OFF time DTMF Relay Volume configuration T.38 Fax Volume configuration Flash Time transmit via SIP info (enable or disable) Message Waiting Indication (Stutter Tone Notice) Blocks Anonymous Call Call Hold, Call Transfer
FXO/FXS Line Configuration	Activates or deactivates: Line ID, line phone number Polarity reversal detection or generation for call establishment and billing Hot line to desired phone number Plays voice file to incoming call Repeats playing voice file counts Self-recorded voice files to upload Generates Flash Time to PSTN network T.38 or Fax Relay Type Incoming and outgoing dB value configurable Dialing Answer Delay time to establish call path Answers PSTN incoming call following the number of rings VoIP dial to FXO/PSTN Line: 1 stage dialing and 2 stage dialing Outgoing SIP Caller ID selection Supports 24 SIP trunk Accepts desired SIP Proxy incoming calls only
Flexible Routing Plan	Prefix match and length Priority ring Cyclic ring Simultaneous ring Programmable hunting cycle Backup routes with digit manipulation Default routes
Flexible Dial Plans	Retrieves transfer call from 3rd party by dial code (default: *#) Inter digit time out setting First digit dial out delay time setting End of dial keypad number Dial rule: Match dial prefix and maximum digits length (1-15) Phone book can be exported or imported
FXS Analog 2-wire Interface	Flash Time Detection: range from 80 to 800 ms ON-HOOK Voltage -48V DC Configures ring cadence, frequency and voltage Supports polarity reversal for billing Service up to 1km in distance to analog telephone set Generate Current Drop Time (Open Loop Disconnect time)
FXO Analog 2-wire Interface	Incoming Ring frequency recognition range: 10 to 70 Hz Incoming Ring ON time recognition range: 0 to 8000ms Incoming Ring OFF time recognition range: 0 to 8000ms Incoming Ring Level recognition range: 10 to 95Vrms Flash Time Detection: range from 80 to 800 ms Configures ring cadence, frequency and voltage

Management	Administrative Telnet CLI and HTTP, HTTPS HTTP provision through MAC address Multilingual Web user interface 3 levels of user access right with password protection with a different Web language (Administrator, Supervisor and User) HTTP/HTTPS service access limitation from WAN port Configures service ports at HTTP, HTTPS and Telnet services Phone debug module: Device Control, Call Control, DB, Verbose SIP debug module: Register, Call, SIP Message, Others SNTP debug module Device debug module DSP debug Provides system status logs Connect to external Syslog server Status display: Network, line, SIP trunk status Diagnostics (debug through Syslog event notice) Debug in real time by Telnet Auto provision via HTTP server SNMP v2 / Trap Configuration Backup/Restore Dual Firmware Image Backup Reset to factory default
Environments	
Power Requirements	12V DC, 3.33A
Operating Temperature	0 ~ 45 degrees C
Operating Humidity	10%~90% relative humidity, non-condensing
Weight (with package)	2.3kg
Dimensions (W x D x H)	440 x 110 x 45 mm
Emission	CE, FCC, RoHS
Connectors	Two 10/100BASE-TX RJ45 Ethernet ports Eight RJ11 ports DC power jack

Ordering Information

VGW-804	8-Port SIP VoIP Gateway (4 FXS + 4 FXO)

Related Products

VGW-800FO	8-Port SIP VoIP Gateway (8FXO)
VGW-800FS	8-Port SIP VoIP Gateway (8FXS)
VGW-402	4-Port SIP VoIP Gateway (2*FXS + 2*FXO)
VIP-1010PT	High Definition PoE IP Phone (1-line)
VIP-2020PT	Enterprise HD PoE IP Phone (2-line)
VIP-5060PT	Professional HD PoE IP Phone (6-line)
VIP-6040PT	Gigabit Color LCD HD PoE IP Phone (4-line)
VIP-8030NT	HD Voice Conference IP Phone with PSTN (3-line)
ICF-1800	HD Touch Screen Android Multimedia Conference Phone (6-line)
IPX-330	Internet Telephony PBX System (30 user registrations)
IPX-2100	Internet Telephony PBX System (100 user registrations)
IPX-2500	Internet Telephony PBX System (500 user registrations)
UMG-1000	Desktop Unified Office Gateway
UMG-2200	Unified Office Gateway (8-port FXO)
VIP-156	SIP Analog Telephone Adapter
VIP-156PE	802.3af PoE SIP Analog Telephone Adapter
VIP-157	1 FXS / 1 FXO SIP Analog Telephone Adapter
VIP-157S	2 FXS Analog Telephone Adapter
VIP-1680 Series	16-Port FXS H.323 / SIP VoIP Gateway
VIP-2480 Series	24-Port FXS H.323 / SIP VoIP Gateway

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