

SIP VoIP Gateway



High Quality yet Affordable for All Businesses

PLANET VGW-x20FS enterprise-class SIP VoIP Gateways with 4, 8, 16, 24 and 32 FXS ports provide added flexibility during migration to Unified Communications by supporting the traditional analog devices. These devices include analog phones, fax machines, modems, voicemail systems and speakerphones.

Enhanced, Full-Featured Business Gateway

The VGW-x20FS Gateways, compliant with IETF SIP RFC 3261 standard, provide a total solution for integrating voice-data networks with built-in SIP trunk and TLS/SRTP security. They support up to 4, 8, 16, 24, 32 concurrent connections. Voice communications can be established from anywhere around the world, and they not only provide quality voice communication, but also offer secure, reliable Internet sharing capabilities for daily voice and Internet communications.



SIP Applications

- · IETF SIP RFC3261 based on UDP/TCP/TLS
- Their 4, 8, 16, 24, and 32 FXS ports connect to analog phone sets or PABX systems.
- · Fax over T.38 and pass-through
- ITU-T G.711 A-law, G.711 μ-law, G.723.1 and G.729 voice coding
- In-band/out-of-band DTMF (RFC4733, RFC2833 / SIP INFO)
- Echo cancellation exceeding ITU-T G.168, up to 128ms tail length
- · SIP trunk and caller ID: DTMF/FSK CLI Presentation

Internet Features

- SNMP v1/v2/v3
- VLAN 802.1P and 802.1Q
- · Layer 3 QoS and DiffServ
- · STUN (RFC 3489) and outbound proxy
- · TR069 and auto provisioning
- TLS/SRTP security

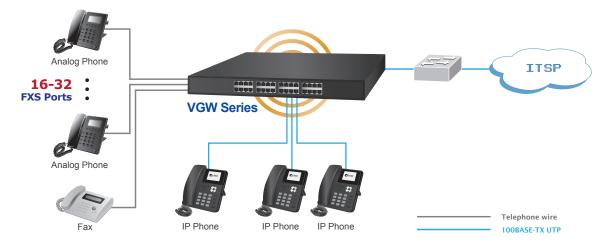
Call Features

- Call waiting, transfer (Blind transfer, attend transfer)
- · Call hold, quick pick
- · Call forwarding unconditional
- · Call forwarding on no reply
- · Hotline, speed dial, direct IP call
- Do not disturb (DND), 3-way conferencing



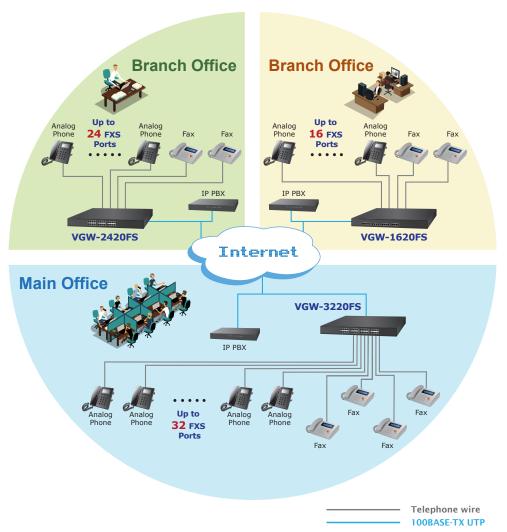
Distributed VoIP Network Infrastructure

PLANET VGW-x20FS Series is easy to use for all types of businesses as they deliver high-quality voice communication and real-time fax data over IP networks. Thus, they do not need human resources to deploy a VoIP network. With the optimized SIP architecture, the VGW series is the ideal choice for P2P/SIP proxy (IP PBX) voice chat, and ITSP cost-saving solution.



Applications

The x20FS Series offers all the essential features you need for business-class voice communication 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-420FS	VGW-820FS	VGW-1620FS	VGW-2420FS	VGW-3220FS	
Hardware						
VAN	1 x 10/100/1000BASE	T RJ45 port	-	-	-	
_AN	1 x 10/100/1000BASE	1 x 10/100/1000BASE-T RJ45 port		2 x 10/100/1000BASE-T RJ45 port		
XS Ports	4	8	16	24	32	
Console	-	-	1 x RS232, 115200bp	DS		
Veight	300g	300g	2700g	3200g	3200g	
Dimensions (W x D x H)	194x110x28 mm	194x110x28 mm	440x230x44 mm	440x250x44 mm	440x250x44 mm	
Power Requirements	100-240VAC, 50-60 H	Z				
Power Consumption	5W	18W	30W	40W	40W	
rotocols and Standard	• • • • • • • • • • • • • • • • • • • •		700			
Enclosure	■ Pulse: 10 and 20 PP ■ Caller ID: DTMF/FSI ■ Max. Cable Length: ■ Reverse Polarity	■ Dial Mode: DTMF and Pulse ■ Pulse: 10 and 20 PPS ■ Caller ID: DTMF/FSK CLI Presentation ■ Max. Cable Length: 3KM ■ Reverse Polarity ■ Programmable Call Progress Tone				
/oice & Fax	 Silence Suppression Comfort Noise Gene Voice Activity Detect Echo Cancellation (C Adaptive (Dynamic) Hook Flash Programmable Gain T.38/Pass-through Modem/POS DTMF mode: Signal VLAN 802.1P and 80 	■ Programmable Gain Control ■ T.38/Pass-through				
VoIP	■ IETF Session Initiation Protocol (SIP) v2.0 (UDP/TCP) ■ RFC 3261 and Session Description Protocol (SDP) ■ RTP (RFC 2833), RFC 3262, RFC 3263, RFC 3264, RFC 3265, RFC 3515, RFC 2976 and RFC 3311 ■ RTP/RTCP, RFC 2198 and RFC 1889 ■ RFC 4028 Session Timer ■ RFC 3266 IPv6 in SDP ■ RFC 2806 TEL URI ■ RFC 3581 NAT and rport ■ Primary/Backup SIP Server ■ Outbound Proxy ■ DNS SRV/A Query/NATPR Query ■ SIP Trunk ■ Early Media/Early Answer ■ NAT:STUN, Static/Dynamic NAT					
Supplementary Service	 Call Forward on No I Unconditional Call Forward Warm/Immediately For Call Hold Do-not-disturb 3-way Conferencing 	■ Blind Transfer ■ Attend Transfer ■ Call Forward on Busy ■ Call Forward on No Reply ■ Unconditional Call Forward ■ Warm/Immediately Hotline ■ Call Hold ■ Do-not-disturb				



	■ Hunting Group
	■ Web ACL
	■ Telnet ACL
	■ Action URL
Software Features	■ PPPoE/IPv4/IPv6
	■ Digitmap
	■ Bandwidth Optimization
	■ Routing Rules based Prefixes
	■ Caller/Called Number Manipulation
	■ SNMP v1/v2/v3
	■ TR069
	■ Auto provisioning
	■ Web/Telnet
	■ Configuration backup/restoration
	■ Firmware upgrade via web
Management	■ CDR
	■ Syslog
	■ Ping and tracert test
	■ Network capture
	■ Outward test (GR909)
	■ NTP and daylight saving time
	■ IVR local maintenance
Standards Conformance	
Emission	CE, FCC

Ordering Information

VGW-420FS	4-Port SIP VoIP Gateway (4 FXS)
VGW-820FS	8-Port SIP VoIP Gateway (8 FXS)
VGW-1620FS	16-Port SIP VoIP Gateway (16 FXS)
VGW-2420FS	24-Port SIP VoIP Gateway (24 FXS)
VGW-3220FS	32-Port SIP VoIP Gateway (32 FXS)

Related PoE Products

ICF-2000	7" Smart Media Android SIP Conference Phone with Full HD Camera and Touch Screen
HDP-1261PT	1080p SIP Single-unit Vandalproof Door Phone with RFID and PoE
HDP-5261PT	1080p SIP Multi-unit Vandalproof Door Phone with RFID and PoE
VTS-700WP	7-inch SIP Indoor Touch Screen PoE Video Intercom with Built-in Wi-Fi
IPX-2200	Internet Telephony PBX System (200 user registrations)

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VGW-420FS/VGW-820FS VGW-1620FS/VGW-2420FS/VGW-3220FS