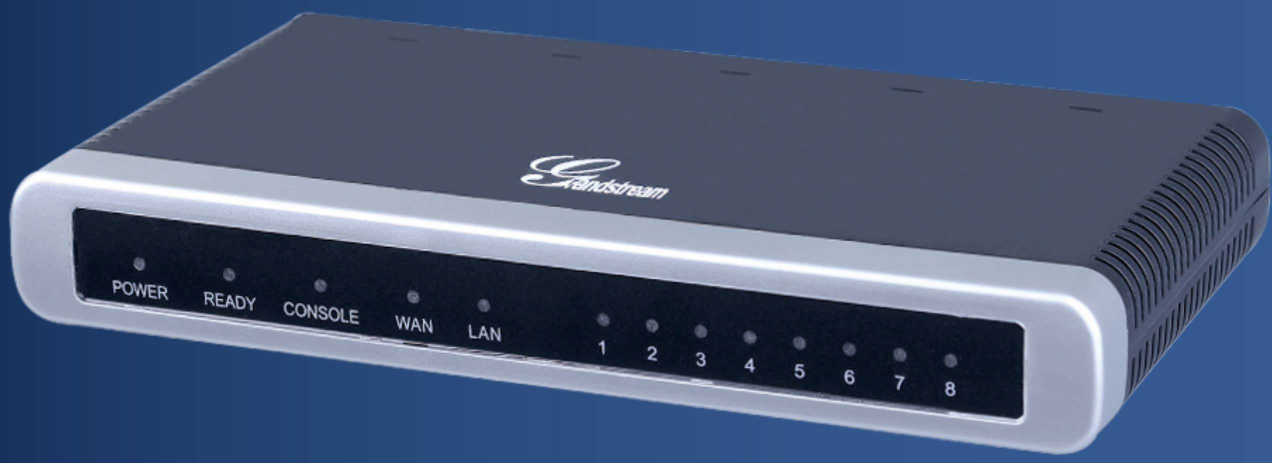
GRANDSTREAM
CONNECTING THE WORLD



GXW4104/4108

4/8 port FXO Gateways

The GXW410x FXO gateway series enables businesses of all sizes to create an easy-to-deploy VoIP solution. These FXO gateways offer the ability to seamlessly connect multiple locations and all devices within an office to any hosted or on premise IP PBX network to make deployments as easy as possible. The GXW410x series includes 4/8 FXO ports, 2 10/100 Mbps ports and supports SIP video through the H.264 codec. Advanced telephony features, easy automated provisioning and superb voice quality allow the GXW410x series to be the ideal VoIP gateway for businesses.

TLS and SRTP security encryption technology to protect calls and accounts

Automated provisioning options include TR-069 and XML config files

3 WAY
Supports 3-way voice conferencing

Failover SIP server feature automatically switches to secondary server if main server loses connection

Supports T.38 Fax for creating Fax-over-IP

Supports a wide range of caller ID formats

zero CONFIG
Use with Grandstream's UCM series of IP PBXs for Zero Configruartion provisioning

Supports advanced telephony features, including call transfer, call forward, call-waiting, do not disturb, message waiting indication, multi-language promtps, flexible dial plan and more

Telephone Interfaces	GXW4104: 4RJ11 FXO ports GXW4108: 8 RJ11 FXO ports
Network Interfaces	10M/100M/1000Mbps, dual RJ45 ports
LED indicators	Power, Network and Line LEDs
Voice-over-Packet Capabilities	G.168 compiant Echo Cancelation, Dynamic Jitter Buffer, Moderm detection & auto-switch G.711
PTSN Fail-over Life Line	PTSN failover on power failure
Voice Compression	G.711, G.723, G.726 (40/32/24/16), G.729A/B/E, G.728, iLBC
DHCP Client/Server	Yes, NAT Router or Switched Mode
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
QoS	Diffserve, TOS, 802.1P/Q VLAN tagging
IP Transport	RTP/RTCP
DTMF Method	flexible DTMF transmission method, user interface of in-audio, RFC2833, and/or SIP info
IP Signaling	SIP (RFC 3261)
SIP Server Profiles & Accounts Per System	Up to 2 distinct SIP server profiles per system and independent SIP account per port
Provisioning	TFTP, HTTP, HTTPS
Media	SRTP not available
Control	TLS/SIPS
Management	Syslog support, HTTPS, telnet, remote mamangement using web browser
Power	Output: 12VDC Input: 100-240VAC / 50-60Hz
Environmental	Operational: 32-104°F or 0-40°C Storage: 14-140°F or -10-60°C Humidity: 10-90% Non-condensing
Dimensions (LxWxH)	230mm x 135mm x 35mm
Mounting	Wall mount and desktop
Short & Long Haul	REN3: Up to 150ft on 24AWG line
Caller ID	Bellcore Type 1&2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal/Wink	Yes
EMC	EN55022/EN5504 and FCC part 15 Class B
Safety	UL