



# 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 Configuration provisioning



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

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