



**GXW410x** is a next generation IP voice & video gateway that features full interoperability with leading IP-PBXs, SoftSwitches and SIP server platforms. The gateway series includes the GXW4104 (4-port) and GXW4108 (8-port) model, each offering superb voice and video quality, traditional PBX telephone functionality, and easy deployment. The GXW410x series also offers a video surveillance port and is the only small business gateway in the industry that provides a video surveillance option.

## Features

- 4-and 8-FXO-port media gateways
- Video surveillance port
- Two RJ-45 ports (switched or routed)
- Supports PSTN/PBX analog trunk lines and SIP trunk interface
- Multiple SIP accounts and profiles (3 per system)
- Audio Codecs: G.711, G.723, G.729, GSM, G.726
- G.168 ECHO Cancellation
- Real time H.264 video codec (CIF resolution, up to 30fps)
- T.38 Fax

## Benefits

- Interconnect remote location legacy phone lines across the Internet.
- Web management for easy configuration and installation.
- Provides fax, voice, video surveillance, and modem support.
- Video surveillance port to enhance premise security.
- Can be a stand-alone gateway for IP Communications.

# GXW410x

## Technical Specifications

	GXW4104	GXW4108
<b>Interfaces</b>		
Voice Ports	4 ports	8 ports
Telephone Interface	FXO, RJ11	
Network Interface	Dual 10/100 Mbps RJ45 ports	
LED Indicators	Power, Video, Network, and Line LEDs	
On/Off Switch	Yes	
<b>Voice, Fax, Modem, Video Surveillance</b>		
Voice over Packet Capabilities	G.168 Line Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711	
Voice Compression	G.711, G.723, G.729A/B, GSM, G.726 (pending)	
Video Surveillance	real-time base line H.264 (CIF resolution, up to 30fps)	
DHCP Client/Server	Yes, NAT Router or Switched Mode	
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kbps and auto-switch to G.711 for Fax Pass-through (pending)	
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging	
IP Transport	RTP/RTCP & PPPoE	
<b>Signaling</b>		
PSTN Signaling	FXO Loop start	
DTMF Method	flexible DTMF transmission method, User interface of In-audio, RFC2833, and SIP Info	
IP Signaling	SIP (RFC 3261)	
Multiple Accounts Per Box	Up to three(3) distinct SIP accounts and profiles	
Provisioning	TFTP and HTTP; Round-robin port scheduling to ensure available lines to access PSTN	
<b>Security</b>		
Media	SRTP	
Management	Syslog support, HTTPS and telnet (pending), remote management using Web browser	
<b>Physical</b>		
Power	Output: 12VDC, Input: 100 - 240 VAC/ 50-60 Hz	
Environmental	1. Operational: 32-104°F / 0-40°C / 2. Storage: 14-140°F / -10-60°C / 3. Humidity: 10 -90% Non-condensing	
Dimensions (LxWxH)	225mm x 135mm x 35 mm	
Mounting	Rack mount, Wall mount, Desktop	
<b>Additional Features</b>		
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID	
Polarity Reversal / Wink	Yes	
<b>Homologation</b>		
EMC	FCC Part 68 & Part 15B, CE (EN55022, EN55024)	
Safety	UL	

