

Grandstream Networks

Innovative IP Voice and Video

Grandstream Presents: The GXW/ HT Series

GXW Series Gateways and HT Series ATAs

GWV400x Series FXS Gateways

The GXW400x FXS Series is an ideal solution for businesses looking to connect one or more lines of a traditional PBX to a VOIP phone system or provider. The GXW400x features 4 or 8-port FXS interfaces for analog telephones, dual 10M/100M network ports with integrated router, PSTN life line in case of power failure, and an RS232 serial port for administration.

- Support for 2 SIP account profiles, caller ID for various countries/regions
- Support for T.38 fax, flexible dialing plans, security protection (SIPS/TLS), and comprehensive voice codecs



[GXV400X Product Brochure/Specifications](#)

[Configuring Skype for Business using Grandstream CPE Devices](#)



GWV4024 FXS Gateways

The GXW4024 gateway enables small and medium businesses to create a cost-effective hybrid IP and analog telephone systems and enjoy the benefits of VoIP communications while preserving investment on existing analog phones and traditional PBX systems.

- High density SIP based analog telephone VoIP gateway
- Fully interoperable with leading IP-PBX and Softswitch systems
- Features 24 telephone ports both RJ11 and 50-pin Telco connector



[GXV4024 Product Brochure/Specifications](#)

****New Feature****

The ability to handle 24 NEON light analog hotel phone



GWV410x Series FXO Gateways

The GXW FXO IP Analog Gateway series offers the small enterprise, SOHO, remote offices and multi-location enterprises a cost-effective, easy to deploy VoIP FXO solution. The GXW410x series allows any business to seamlessly connect multiple locations with up to 8 PSTN lines, to an IP PBX system, or with an existing traditional phone system.

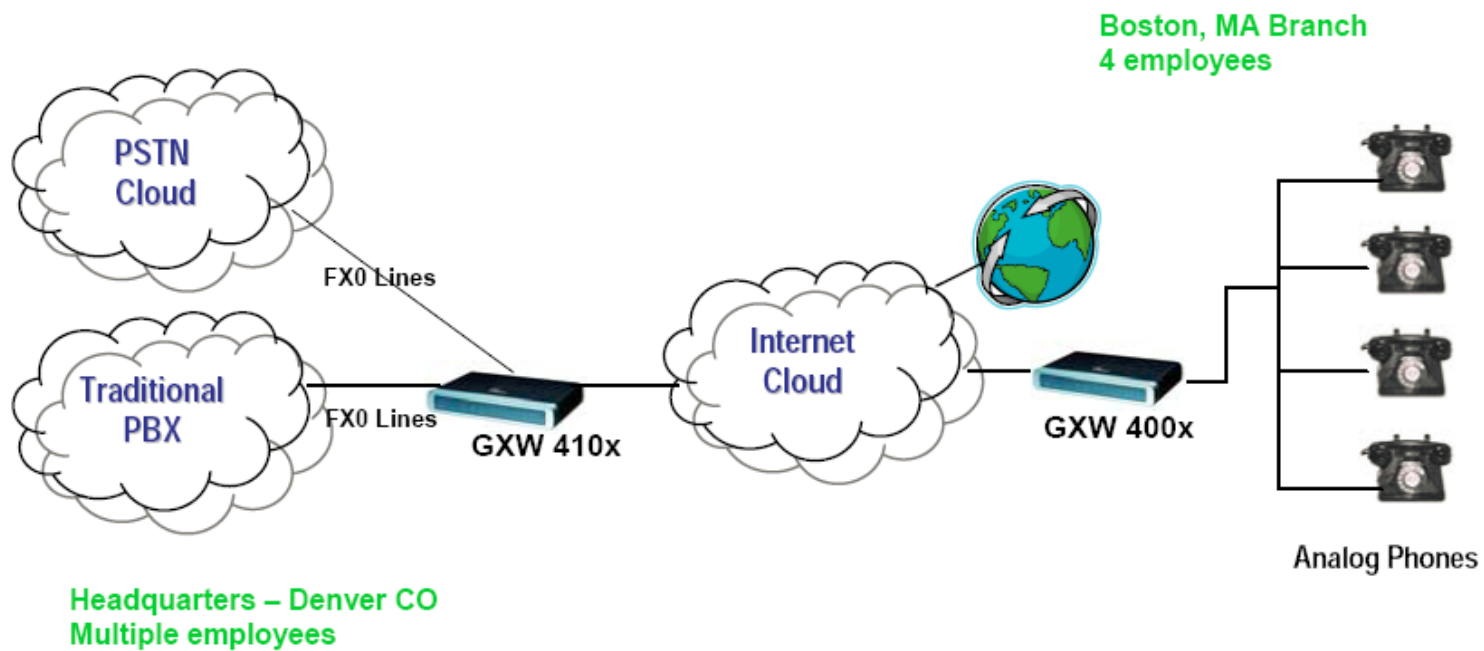
- The GXW series includes two models with 4 or 8 ports respectively
- Designed and tested for full interoperability with leading IP-PBXs, soft-switches and SIP-based environments

[GXV410X Product Brochure/Specifications](#)



[Using GXW-Series to extend analog lines without a SIP server](#)

Extending analog lines w/out a SIP Server



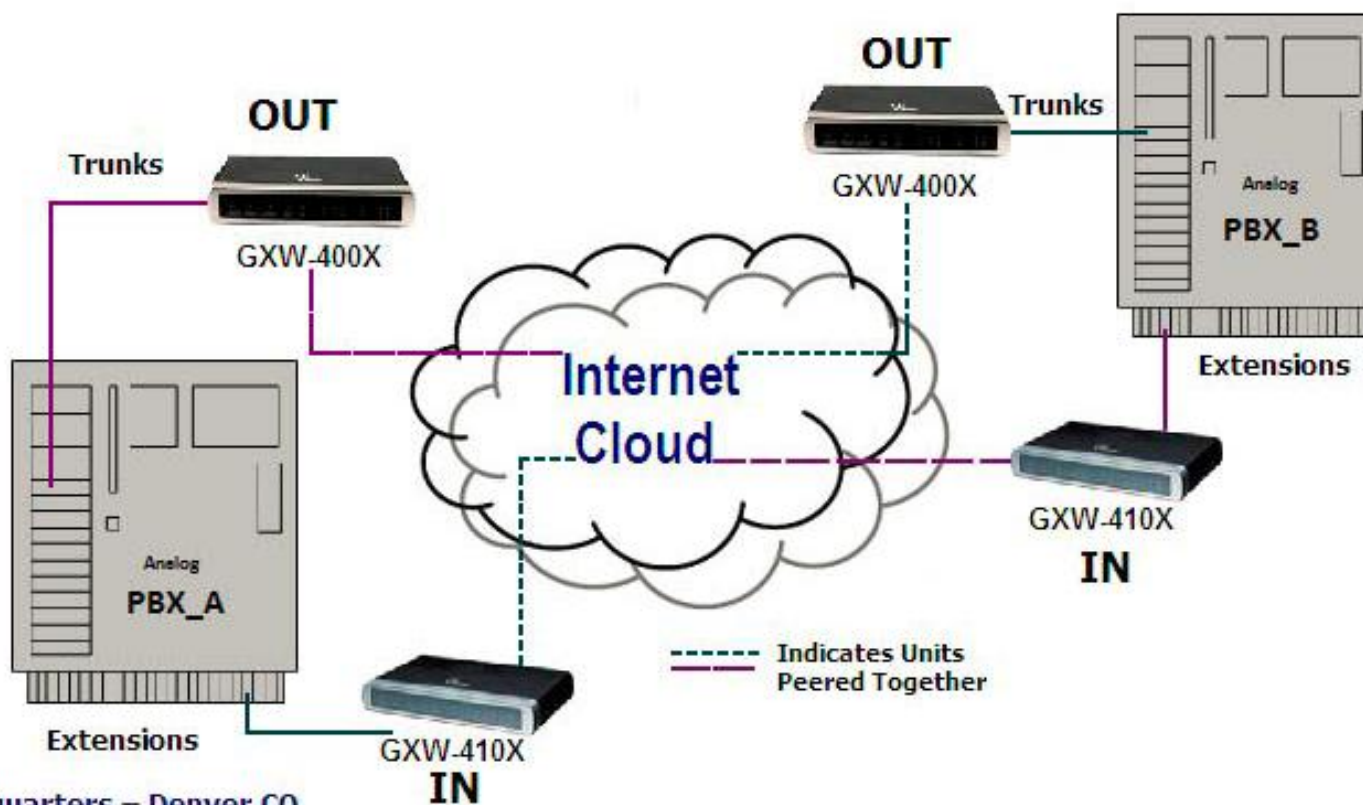
[Detailed configuration info](#)

Peering Legacy PBX using Grandstream Gateways

PEER TO PEER GATEWAY SCENARIO

(Integrating Two Traditional PBX using GXW Series w/o SIP Server)

Boston, MA Branch



[Detailed configuration info](#)

HT502/503



HT502 includes an integrated high performance NAT router and 10Mbps Ethernet WAN and LAN ports enables a shared broadband connection between multiple ethernet devices. In addition to being SIP 2.0 standard compliant, the product supports Universal Plug-in-Play (UPnP), **up to 2 SIP account profiles**, and advanced telephony features.

[GXV410X Product Brochure/Specifications](#)



HT503 includes 1 FXO and 1 FXS port enables remote call origination and termination to and from the PSTN line, known as "hop-on and hop-off" calling.

[HT503 Product Brochure/Specifications](#)



HT701

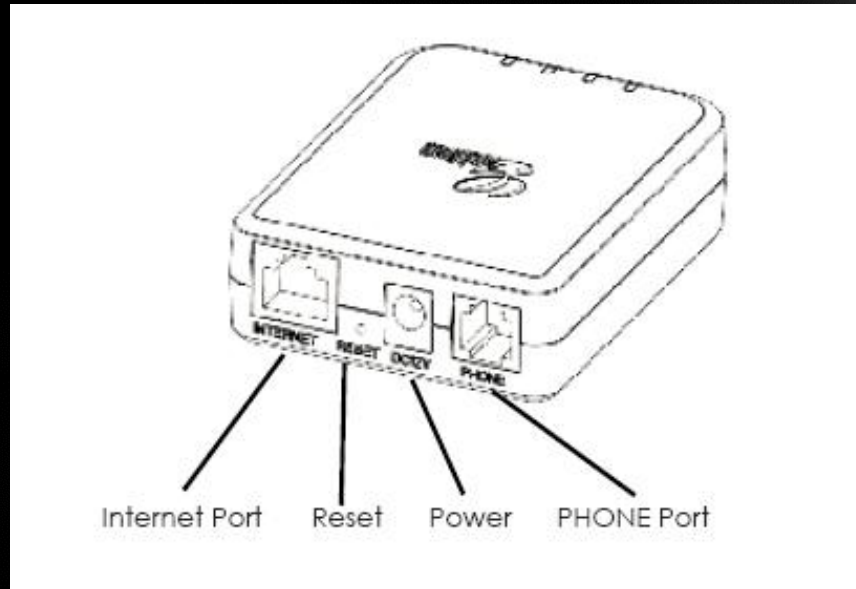


The HT701 is designed for single line analog phone users – residential, small-to-medium-sized business (SMB) and telecommuters – needing a compact, inexpensive ATA that converts legacy analog telephones, fax machines, conference telephones and other analog devices to access Internet-based telephone services. Feature highlights include:

- Ultra compact design (.98in H x 2.6in W x 3.38in L) and extremely affordable cost
- Support for 5 REN over up to 1km on 24 AWG line
- Advanced telephony features including Caller ID for various countries, call waiting, do-no-disturb, 3-way conference, transfer, forward, message waiting, T.38 fax, flexible dial plan, multi-language voice prompt, etc.
- Comprehensive voice codecs including G.711 with Annex I/II, G.729A/B, G.723.1, iLBC, G.726
- Strong security protection of voice/data privacy using TLS/SRTP/HTTPS, secure and automated provisioning using TRo69, HTTP/HTTPS/TFTP and AES encryption

Specifications	HT286	HT701
Telephone Interfaces	1 FXS port	1 FXS port
Network ports	1 (10 Mbps)	1 (10/100 Mbps)
Universal Power Supply	5V ; 1.2 A	12V ; 0.5A
Short and Long Haul	3REN, up to 1 Km on 24 AWG Lines	5REN , up to 1 Km on 24 AWG Lines
Voice Codec	PCMU, PCMA, G723.1, G729, G726-32, iLBC	PCMU, PCMA, G723.1, G729, G726-32, iLBC
Provisioning	TFTP/HTTP	TFTP/HTTP/ HTTPS, Encrypted XML and TR-069
Telephony Features	Hold, transfer, forward, 3-way conference, call waiting, DTMF via SIP INFO/RFC2833/IN-Audio, Dial Plan (HT701 only)	
Security	NO	SRTP, TLS/TCP, Secure Provisioning
Manageability	IVR, Web UI, Provisioning File	IVR, Web UI, Provisioning File, TR-069
Fax Mode	T.38	T.38
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, DTMF Based CID	Bellcore Type 1 & 2, ETSI, BT, NTT, DTMF Based CID
ADVANCED SPECIFICATIONS		
LEDs	1 Led (Power and status indication)	4 Leds (Power, Internet, Link/ACT, Phone) with advanced Indication
SIP Transport	UDP	UDP, TCP, TLS
Reset options	Full reset by IVR only	Full/ISP/VOIP reset by web, Full reset by IVR and reset button
Language	English	English, Chinese and Spanish (IVR)
Sip Sever Failover	DNS SRV	DNS SRV and Failover Sip Server
Security Control	n/a	TLS and SIPS
Mounting	Laying flat	Laying flat or wall mounting
Disconnect methods	Busy Tone, Polarity Reversal	Busy Tone, Polarity Reversal, Loop current

Easy Installation



1. Connect a standard touch-tone analog telephone (or fax machine) to PHONE port.
2. Insert the Ethernet cable into the Internet port of HandyTone and connect the other end of the Ethernet cable to an uplink port (a router, a modem, etc.).
3. Using the HandyTone embedded web server or IVR (Interactive Voice Prompt) menu, you can further configure the phone using either a static IP or DHCP.

THE FOLLOWING **PREVIEW** HAS BEEN APPROVED FOR
ALL AUDIENCES

HT702

- Dual FXS ATA
- Next generation of our popular HT502
- Same advanced telephony features as HT701

HT704

- Ultra-compact 4-Port ATA / gateway
- Next generation of our popular GXW4004
- Same advanced telephony features as HT701

TR-o6g Support

WHAT IS TRo6g?

It is a protocol for communication between CPE (Customer Premise Equipment) and an ACS (Auto Configuration Server) that provides secure auto-configuration as well as other CPE management functions within a common framework.

TR-o6g stands for a technical report defined by the Broadband Forum that specifies the CWMP “CPE WAN Management Protocol”. It commonly uses HTTP or HTTPS as transport for communication between CPE and the ACS. The message exchange is using SOAP (XML_RPC) for configuration and management of the device.

WHY USE TRo6g?

Service Providers, using TR-o6g, can have one common platform to manage all Grandstream devices and other CPEs, no matter the device type or the manufacturer.

This common application was not easily achieved before due to CPE vendor’s proprietary mechanisms for provisioning/management.

Grandstream Giveaway New HT701



