Grandstream
Innovative IP Voice & Video

GXW4024 is a high-density SIP based analog telephone VoIP gateway that is fully interoperable with leading IP-PBX and Softswitch systems. It features 24 telephone ports, superb voice quality, rich telephony functionalities, easy provisioning, flexible dialing plans, and advanced security protection. 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.

Key Features & Applications:

- 24 telephone FXS ports with both RJ11 and 50-pin Telco connector
- Up to 2 SIP server profiles per system and independent account per port
- Supported voice codecs include G.711, G.723, G.726, G.729 A/B/E, iLBC , T.38 Fax
- Carrier grade G.168 line echo cancellation
- Ideal IP enabler for analog phones, faxes, and legacy PBX systems

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Telephone Interfaces	24 x RJ11 FXS ports and 1 x 50-pin Telco connector
Network Interface	1 x 10M/100Mbps auto-sensing RJ45 port
LED Indicators	Power, Ready, LAN, Telephone Ports
Voice-over-Packet Capabilities	G.168 carrier grade line echo cancellation, dynamic jitter buffer, modem detection & auto-switch to G.711
Voice Compression	G.711, G.723, G.726 (40/32/24/16), G.729A/B/E, iLBC
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through
Telephony Features	Caller ID display or block, call waiting, blind or attended call transfer, call forward, do not disturb, 3-way conference, last call return, paging, message waiting indicator LED and stutter tone, auto dial
QoS	DiffServ, TOS, 802.1P/Q VLAN tagging
Network Protocols	TCP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, TELNET, PPPoE, STUN
DTMF Method	Flexible DTMF transmission methods including In-audio, RFC2833, and/or SIP INFO
Signaling	SIP (RFC 3261) over UDP/TCP
SIP Server Profiles and Accounts Per System	Up to two(2) distinct SIP server profiles per system and independent SIP account per telephone port
Provisioning	TFTP, HTTP, HTTPS
Security	SRTP, TLS/SIPS
Management	Syslog, HTTPS, Web browser, TELNET, voice prompt, TR-069(pending)
Universal Power Supply	Output: 12VDC Input: 100-240VAC, 1.0A, 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	445mm x 269mm x 44.5mm (1U)
Weight	2.8kg
Mounting	Desk top & rack mount with front brackets
Short and Long Haul	2 REN: Up to150 ft on 24 AWG line
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Polarity Reversal/Wink	Yes
Safety	UL
Compliance	FCC Part 15B; CE, EN55022/EN55024; C-Tick