Grandstream GXW-4108 Analog FXO Gateway

Product Name: Grandstream GXW-4108 Analog FXO Gateway

Manufacturer: Grandstream

Model Number: GXW4108

Grandstream GXW-4108 Analog FXO Gateway
The Grandstream GXW-4108 offers an easy to manage, easy to configure IP communications solution for any small business or businesses with virtual and/or branch locations who want to leverage their broadband network and/or add new IP Technology to their current phone system. The Grandstream Enterprise Analog VoIP Gateway GXW410x series converts SIP/RTP IP calls to traditional PSTN calls. There are two models - the GXW4104 and GXW4108, which have either 4 and 8 FXO ports respectively.
Grandstream GXW-4108 Features

• 8 FXO analog port gateways
• Video surveillance
• Two RJ-45 ports (switched or routed)
• TFTP and HTTP firmware upgrade support
• Supports Audio Codecs: G711, G723, G729 and GSM
• T.38 compliant
• Web management for easy configuration and installation
• TFTP and HTTP firmware upgrade support
• Multiple SIP accounts, associated with physical line ports, each account corresponding to one of the multiple SIP profile
• Multiple SIP profiles, max of 3 profiles per system. Each profile hosts 0 to multiple number of SIP accounts, depending on user need
• One stage and two stage dialing
• Two stage dialing means when after dialing the number to the GXW, be it from VoIP to GXW or from PSTN to GXW, a second dial-tone prompts users to input the final destination number to finish final dialing.
• One stage dialing means user only hear dial-tone once and input a final destination number along with a pre-fix. One stage dialing need SIP server to support SIP call forward via a dial-plan.
• VoIP to PSTN call setup and teardown
• Channel configurable for one stage or two stage dialing, Default is 2 stage dialing.
• PSTN to VoIP call setup and teardown
• Channel configurable for one stage or two stage dialing, Default is 2 stage dialing. One stage dialing requires user to configure Off-Hook Auto Dial to a SIP Number.
• Support: G711, G723, G729, and GSM
• Line echo canceller g.168 support
• Flexible DTMF transmission method User Interface of In-audio, RFC2833, and SIP Info
• Round-robin port scheduling to ensure available lines to access PSTN networks

The installation is the same for either model. A SIP proxy server such as Asterisk or a SIP registrar server can be deployed with the GXW-4108 series. In this environment, the SIP server handles SIP registration and call control and the GXW4108 processes media conversion between IP and PSTN calls. By design, the system supports the North American call progress tones and signaling standards on PSTN sides.
Grandstream GXW-4108 Technical Specification

FXO Ports

• 8 FXO ports

Ethernet Ports

• 2 RJ45 10/100Mbps (LAN/WAN)

Video input

• Supports H.264 video codec (up to 30fps and CIF resolution, Hardware Version below 2.0 only)

Audio Codecs

• G711u/a
• G723
• G729
• GSM
• T.38 compliant

Configuration

• Web Based
• Remote TFTP/HTTP

Power Input

• 12 Vdc
• 1.25Amp

Compliance

• FCC/CE/C-Trick

Configurable channel dialing to improve dial-out reliability

• digit length: default 100ms
• digit volume: gain [-31,0]dB, default -11dB
• dial pause between digits: default 100ms
• wait for dial-tone: yes/no, default yes (1 for Yes, 2 for No)
• one-stage (use 1) or 2 stage (use 2) dialing: default of 2 stage dialing
• Syntax: ch (or chan or channel) x-y: val; ch

Configurable call progress/termination tones via pattern matching

• Dial-tone: f1/f2(350/440), v1/v2(-11/ -11), on1/off1(0/0), on2/off2(0/0)
• Ring back tone: f1/f2(default 440/480), on/off(default 2s/4s)
• Busy tone: f1/f2(480/620), on/off(0.5/0.5s), duration (8s)
• Re-order tone: f1/f2(480/620), on/off(25/25), duration (default 8s)
• Confirmation tone: f1/f2(350/440), on/off(0.1/0.1s), duration (default 8s)
• Usage Syntax: - ch x-y: f1(or freq1 or frequency1)=val1@vol1, f2 (or freq2 or frequency2) = val2@vol2, c (or cad or cadence) = on1/off1-on2/off2-on3/off3; ch3: - x,y - 0-9 digit.
• Configure Channel voice settings,
• Voice volume: gain control, [-31, 31], default 1 dB
• Audio input gain: [-31, 31], default 0 dB
• Silence Suppression: 1 - enabled, 2 - disabled, default is 1
• Line echo cancellation: 1 - enabled, 2 - disabled; default is 1

DTMF Method via : default value is in-audio
1 - in-audio 2 - RFC2833 3 - in-audio and RFC2833 4 - SIP Info 5 - in-audio and RFC2833

Price: £232.89