Grandstream GXW-4104 Analog FXO Gateway

Product Name: Grandstream GXW-4104 Analog FXO Gateway

Manufacturer: Grandstream

Model Number: GXW4104

Grandstream GXW-4104 Analog FXO Gateway
The Grandstream GXW-4104 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-4104 Features

• 4 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-4104 series. In this environment, the SIP server handles SIP registration and call control and the GXW-4104 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-4104 Technical Specification
FXO Ports

• 4 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: £165.79