

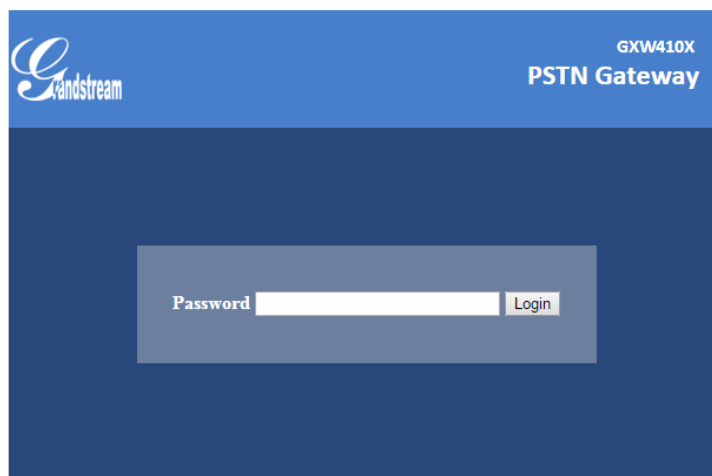


GUÍA SÚPER RÁPIDA DE PROGRAMACIÓN PARA EQUIPOS GXW4104 y GXW4004.

Configuraciones necesarias para enviar hasta 4 Extensiones/Troncales a un sitio remoto

GXW4104 (BASE- IP 192.168.1.2)

1. Conecte el equipo a un ruteador de Internet por el puerto WAN.
2. Acceda al ruteador de Internet y verifique la dirección IP asignada al equipo GXW4104 mediante la dirección MAC.
3. Ingrese al equipo mediante el navegador web con la contraseña “**admin**” a la IP asignada por el ruteador.



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4. Aparecerá la pantalla de bienvenida (**Overview**) del equipo.

GXW410X PSTN Gateway Logout Reboot

Grandstream Status Accounts Settings Networks Maintenance FXO Lines Line Analysis

Version: 1.4.1.5

Line Analysis **Overview**

Overview

[Auto Detect](#) **Overview of PSTN Line Interface**

For optimum functionality all analogue equipment needs to be appropriately tuned to its environment. This FXO Gateway also requires little tuning prior to configuring the VoIP settings

There are several aspects of PSTN line configurations that will enhance the performance of this device. They are related to call quality, reliability as well as functionality.

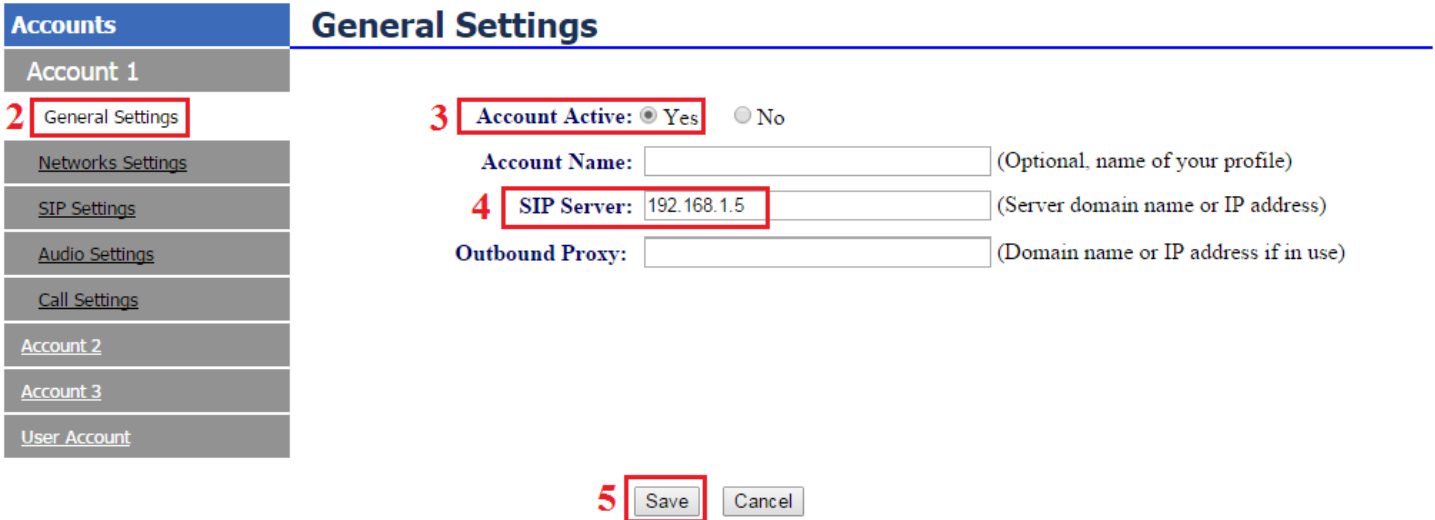
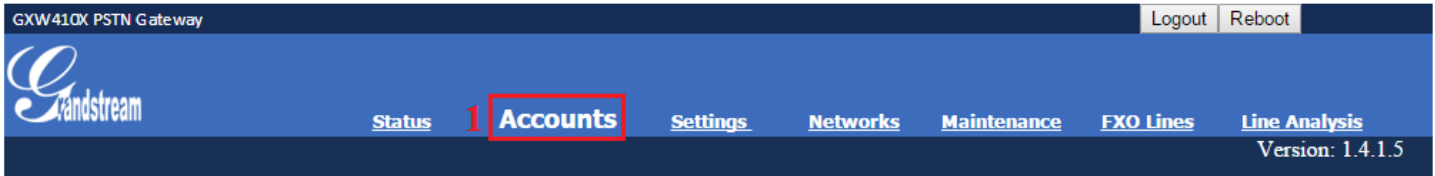
Call Quality:

In order to have better audio quality it is important that the impedance on the line is matched correctly on every port of the device. This PSTN gateway support 16 different impedance standards for world wide compatibility. Each port can be individually configured with its own impedance. If all of the PSTN lines come from the same provide chances are that they are the same impedance so a single impedance setting should be used for all lines. On the contrary there are cases when different PSTN lines are in use coming from different providers or even lines extensions from PBX systems or other FXS extension devices. In this case the lines may have different impedance therefore requiring different setting for each line.

Call Reliability:

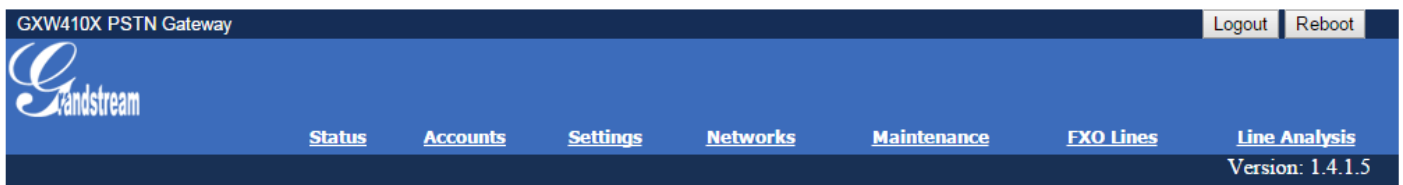
To achieve reliable connectivity it is important that the Gateway understands how to place and terminate calls on the PSTN line. Due to the large collection of termination signals and protocols around the world it is important that the correct method for call disconnection is configured in every region, in order to ensure reliable call termination, therefore avoiding hanging lines. This Gateway supports two call termination methods including Current Disconnect (aka Disconnect Supervision), and Busy tone detection (aka Call progress tone detection).

5. Ahí nos dirigimos hacia la pestaña “ACCOUNTS” y realizamos los siguientes cambios:



Nota: se asume que el equipo remoto será el 192.168.1.5 (GXW4004).

6. Después de guardar cambios en cada sección, el equipo solicitará reiniciar el equipo; el reinicio se deberá realizar al haber concluido todos los cambios necesarios para cada modelo.



Your configuration changes have been saved.
They will take effect on next reboot.

Reboot

7. En la misma pestaña “ACCOUNTS”, ahora vamos a “SIP SETTINGS” y cambiamos el parámetro indicado.

Accounts	SIP Settings
Account 1	
General Settings	
Networks Settings	
6 SIP Settings	SIP Registration: <input type="radio"/> Yes <input checked="" type="radio"/> No 7
Audio Settings	Unregister On Reboot: <input type="radio"/> Yes <input checked="" type="radio"/> No
Call Settings	Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)
Account 2	SIP Reg Failure Retry Wait: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)
Account 3	SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP
User Account	Session Expiration: <input type="text" value="180"/> (in seconds. default 180 seconds)
	Min-SE: <input type="text" value="90"/> (in seconds. default and minimum 90 seconds)
	Caller Request Timer: <input type="radio"/> Yes <input checked="" type="radio"/> No (Request for timer when making outbound calls)
	Callee Request Timer: <input type="radio"/> Yes <input checked="" type="radio"/> No (When caller supports timer but did not request one)
	Force Timer: <input type="radio"/> Yes <input checked="" type="radio"/> No (Use timer even when remote party does not support)
	UAC Specify Refresher: <input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)
	UAS Specify Refresher: <input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)
	Force INVITE: <input type="radio"/> Yes <input checked="" type="radio"/> No (Always refresh with INVITE instead of UPDATE)
	Enable 100rel: <input type="radio"/> Yes <input checked="" type="radio"/> No
	Refer-To Uses Target Contact <input checked="" type="radio"/> No <input type="radio"/> Yes
	Accept Proxy INVITE Only <input type="radio"/> No <input checked="" type="radio"/> Yes
	Special Feature: <input type="text" value="Standard"/> ▼
	8 <input type="button" value="Save"/> <input type="button" value="Cancel"/>

8. Ahora, en la pestaña “SETTINGS”, modificamos los parámetros indicados en “CHANNELS SETTINGS” y guardamos cambios.

GXW410X PSTN Gateway Logout Reboot

Grandstream Status Accounts **9 Settings** Networks Maintenance FXO Lines Line Analysis

Version: 1.4.1.5

Settings **Channels Settings**

General Settings

Call Settings SIP Channel Setting

Channels Settings **10**

DTMF Methods(1-7): (default 1)
(1:in-audio, 2:RFC2833, 3:1+2, 4:SIP Info, 5:1+4, 6:2+4, 7:1+2+4)

11 No Key Entry Timeout(Xls): (1-9, default 4)

Local SIP Listen Port: (default ch1-8:5060++;)

SRTP Mode(1-3): (default 1)
(1:disabled, 2:enabled but not forced, 3:enabled and forced)

Calling to VoIP

Unconditional Call Forward to Following:

12 User ID: (i.e ch1-2:223;ch3:224)

SIP Server: (ch1-2:p1;ch3:p2)

13 SIP Destination Port: (ch1-2:5060;ch2:7080)

T.38 FAX Settings

(Syntax: ch x-y: mode=val,rate=val,ecm=val:[...])

T.38 Settings:

(mode: 1:Relay(default), 2:Passthrough)

(rate: 2400, 4800, 7200, 9600(default), 12000, 14400)

(ecm: 1:Enable(default), 0:Disable)

14

9. En la pestaña “FXO LINES”, en “SETTINGS”, aplicamos los siguientes cambios

GXW410X PSTN Gateway Logout Reboot

Grandstream 15

[Status](#) [Accounts](#) [Settings](#) [Networks](#) [Maintenance](#) **FXO Lines** [Line Analysis](#)

Version: 1.4.1.5

FXO Lines **FXO Settings**

Settings 16

Dialing

Call Progress Tones

[Syntax: ch x-y: f1=val@vol,f2=val@vol,c=on1/off1-on2/off2-on3/off3; ...]

Note: f1,f2-frequency(Hz); vol-volume(dB); c-cadence(10ms, 0-continuous)

Dial Tone:

Ringback Tone:

Busy Tone:

Reorder Tone:

Port Voice Setting

Tx to PSTN Audio Gain(dB): (-12-12, default 1)

Rx from PSTN Audio Gain(dB): (-12-12, default 0)

Silence Suppression(Y/N): (default Yes)

Echo Cancellation(Y/N): (default Yes)

FXO Termination

17 **Enable Current Disconnect(Y/N):** (default Y=yes)

Current Threshold: if yes(5~65530,default 100ms)

18 **Enable Tone Disconnect:** (default No; Yes - busy tone)

Enable Polarity Reversal: (default No; Consult carrier)

Enable Call Supervision: (default No; Consult carrier)

Silence Timeout(X1s): (default 60s)

Incoming Ring Timeout(X1s): (2-10s, default 6s)

AC Termination Impedance: (0-15, default 0)

AC Termination Impedance Values (0-15, default 0)

- 0 - 600 Ohm (North American)
- 1 - 900 Ohm
- 2 - 270 Ohm + (750 Ohm || 150nF) and 275 Ohm + (780 Ohm || 150nF)
- 3 - 220 Ohm + (820 Ohm || 120nF) and 220 Ohm + (820 Ohm || 115nF)
- 4 - 370 Ohm + (620 Ohm || 310nF)
- 5 - 320 Ohm + (1050 Ohm || 230nF)
- 6 - 370 Ohm + (820 Ohm || 110nF)
- 7 - 275 Ohm + (78 Ohm || 150 nF)
- 8 - 120 Ohm + (820 Ohm || 110 nF)
- 9 - 350 Ohm + (1000 Ohm || 210nF)
- 10 - 0 Ohm + (900 Ohm || 30nF)
- 11 - 600 Ohm + 2.16 uF
- 12 - 900 Ohm + 1 uF
- 13 - 900 Ohm + 2.16 uF
- 14 - 600 Ohm + 1 uF
- 15 - Global complex impedance

Port Caller ID Setting

19 **Number of Rings Before Pickup:** (1-50, default 4)

Caller ID Scheme: (1-11, default 1)

- 1 - Bellcore/Telcordia
- 2 - ETSI-FSK during ringing
- 3 - ETSI-FSK prior to ringing with DTAS
- 4 - ETSI-FSK prior to ringing with LR
- 5 - ETSI-FSK prior to ringing with RP
- 6 - ETSI-DTMF during ringing
- 7 - ETSI-DTMF prior to ringing with DTAS
- 8 - ETSI-DTMF prior to ringing with LR
- 9 - ETSI-DTMF prior to ringing with RP
- 10 - SIN 227 - BT
- 11 - NTT - Japan

Caller ID Transport Type: (1-4, default 1)

- 1 - Relay via SIP From
- 2 - Disabled
- 3 - Send Anonymous
- 4 - Relay via SIP P-Asserted-Identity

20

10. En la misma pestaña “FXO LINES” pero en “DIALING”, aplicamos lo que se muestra a continuación

GXW410X PSTN Gateway Logout Reboot

Grandstream Version: 1.4.1.5

Status Accounts Settings Networks Maintenance **FXO Lines** Line Analysis

FXO Lines **Dialing**

Settings

Dialing **22**

Dialing to PSTN

23 **Wait for Dial-Tone(Y/N):** (default No)

24 **Stage Method(1/2):** (default 2 stage dialing)

Min Delay Before Dialing Out: (default 500ms, 50 ~ 65000ms)

Port Scheduling Schema

25 **Round-robin and/or Flexible:** (default rr:1-8;)

(Syntax: rr: port_group; [...]; Default: rr:1-8; round-robin of all ports)

Prefix to Specify Port: (1 stage dial, default 99)

(Syntax: prefix# + ch# + dialing# will request the ch# per call. Note that this code has to prefix dialplan number and prefix doesn't impact round-robin)

Use SIP User Account Scheduling: No Yes

(Note that Yes disables above 2 scheduling, which is not recommended)

26

******AQUÍ TERMINA LA CONFIGURACIÓN DEL GXW4004******

GXW4104 (REMOTO- IP 192.168.1.5)

1. Conecte el equipo a un ruteador de Internet por el puerto WAN.
2. Acceda al ruteador de Internet y verifique la dirección IP asignada al equipo GXW4004 mediante la dirección MAC.
3. Ingrese al equipo mediante el navegador web con la contraseña “admin” a la IP asignada por el ruteador.

4. Aparecerá la pantalla de bienvenida (**STATUS**) del equipo.

Port	Hook	Registration	DND	Forward	Busy Forward	Delayed Forward
FXS 1	On Hook	Not Registered	No			
FXS 2	On Hook	Not Registered	No			
FXS 3	On Hook	Not Registered	No			
FXS 4	On Hook	Not Registered	No			

5. En la pestaña “PROFILE 1”, es necesario realizar los siguientes cambios

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	PROFILE 1	PROFILE 2	FXS PORTS
<p>Profile Active: <input type="radio"/> No <input checked="" type="radio"/> Yes 2 1</p> <p>3 Primary SIP Server: <input style="border: 2px solid red;" type="text" value="192.168.1.2"/> (e.g., sip.mycompany.com, or IP address)</p> <p>Failover SIP Server: <input type="text"/> (Optional, used when primary server no response)</p> <p>Prefer Primary SIP Server: <input checked="" type="radio"/> No <input type="radio"/> Yes (yes - will register to Primary Server if Failover registration expires)</p> <p>Outbound Proxy: <input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)</p> <p>SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)</p> <p>NAT Traversal: <input checked="" type="radio"/> No <input type="radio"/> Keep-Alive <input type="radio"/> STUN <input type="radio"/> UPnP</p> <p>DNS Mode: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV <input type="radio"/> Use Configured IP</p> <p>Primary IP: <input type="text"/></p> <p>Backup IP1: <input type="text"/></p> <p>Backup IP2: <input type="text"/></p> <p>Tel URI: <input type="text" value="Disabled"/> ▼</p> <p>Use Request Routing ID in SIP INVITE Header: <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>4 SIP Registration: <input style="border: 2px solid red;" type="radio"/> No <input type="radio"/> Yes</p> <p>Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes</p> <p>Outgoing Call without Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes</p> <p>Ring Tones (Syntax: c=on1/off1-on2/off2-on3/off3;)</p> <p>Ring Tone 1: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 2: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 3: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 4: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 5: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 6: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 7: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 8: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 9: <input type="text" value="c=2000/4000;"/></p> <p>Ring Tone 10: <input type="text" value="c=2000/4000;"/> 5</p>					
<input type="button" value="Update"/> <input style="border: 2px solid red;" type="button" value="Apply"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>					
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Nota: se asume que el equipo remoto será el 192.168.1.2 (GXW4104).

6. En la pestaña “FXS PORTS”, debe quedar como lo siguiente

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
PROFILE 1
PROFILE 2
FXS PORTS

6

User Settings

Port	SIP User ID	Authenticate ID	Password	Name	Profile ID	Hunting Group	Request URI	Routing ID	Enable Port
1	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▾	None ▾	<input type="text"/>	<input type="text"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes
2	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▾	None ▾	<input type="text"/>	<input type="text"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes
3	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▾	None ▾	<input type="text"/>	<input type="text"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes
4	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	Profile 1 ▾	None ▾	<input type="text"/>	<input type="text"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes

Port	Offhook Auto-dial (e.g. 800123456)	Map to FXO Port# (e.g. valid line# 1-n)	Map to FXO Gateway IP (e.g. 192.168.1.22)	and Port (e.g. 5060)
1	<input type="text"/>	1	192.168.1.2	5060
2	<input type="text"/>	1	192.168.1.2	5062
3	<input type="text"/>	1	192.168.1.2	5064
4	<input type="text"/>	1	192.168.1.2	5066

7
8

9

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*****AQUÍ TERMINA LA CONFIGURACIÓN DEL GXW4004*****

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