

## CONFIGURACIÓN DE GXW410X CON FREEPBX

Realmente la configuración es bastante simple, pero la primera vez que configure uno de estos equipos me tomo un par de horas, a pesar de buscar bastante información en Internet, así que voy a explicar el procedimiento que utilice y básicamente lo que se tiene que hacer para configurar este equipo para llamadas salientes y entrantes usando el freePBX

Ingresamos via web:

**Grandstream Device Configuration**

[Status](#)   [Basic Settings](#)   [Advanced Settings](#)   [FXO Lines](#)   [Channels](#)   [Dial-plan](#)   [Profile 1](#)   [Profile 2](#)   [Profile 3](#)

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**FXO Termination**

1. Enable Current Disconnect:  No  Yes (Default Yes)  
 If enabled, use threshold:  (default 100, normally 100 ~ 800ms)

2. Enable Tone Disconnect:  No  Yes (If set yes, reorder tone is used as disconnect signal. Default No)

3. Enable Polarity Reversal:  No  Yes (Default No. Check with your PSTN carrier before set to Yes)

4. AC Termination Impedance:  (default 600)

5. Silence Timeout(X1s):  (default 60s)

**Channel Dialing**

1. DTMF Digit Length(X10ms):  (1-200, default 10)

2. DTMF Digit Volume(dB):  (-31-0, default -11)

3. DTMF Dial Pause(X10ms):  (1-200, default 10)

4. Wait for Dial-Tone(Y/N):  (default Yes - dial upon dial-tone)

5. Stage Method(1/2):  (default 1 - 1 stage dialing)

6. Min Delay Before Dial PSTN:  (default 100, normally 100 ~ 800ms)

7. Unconditional Call Forward to VOIP:

User ID	Sip Server	Sip Destination Port
ch1-4:1105;	@ ch1-4:p1;	: ch1-4:5060;

**PSTN to VOIP Caller ID Setting**

1. Caller ID Scheme:  (1-5, default 1)  
 (1:Bellcore, 2:ETSI\_RING, 3:ETSI\_TAS, 4:DTMF, 5:NTT)

2. Caller ID Transport Type:  (1-4, default 1)  
 (1:Relay via SIP From, 2:Disable, 3:Send Anonymous, 4:Relay via SIP P-Asserted-Identity)

**T.38 Setting** (Syntax: ch x-y: mode=val,rate=val,ecm=val[...])

1. T.38 Setting:   
 (mode: 1:Relay(default), 2:Passthrough)  
 (rate: 2400, 4800, 7200, 9600(default), 12000, 14400)  
 (ecm: 1:Enable(default), 0:Disable)

    

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La imagen es solo referencial, por lo que se tendrán que hacer los siguientes cambios:

### *Channel Dialing*

Wait for Dial-Tone(Y/N): **ch1-4:Y;**

Stage Method(1/2): **ch1-4:1;**

En la parte de *Unconditional Call Forward to VOIP*.

**User ID**

ch1-4:1105; #esto significa que todas las llamadas que ingresen por los canales 1,2,3 y 4 serán direccionadas al anexo 1105 (aquí lo mejor será crear un anexo virtual al IVR)

**Sip Destination Port**

ch1-4:p1; #esto significa que los canales 1,2,3 y 4 son usados por el Profile 1

**User ID**

ch1-4:5060; #esto significa que los canales 1,2,3 y 4 usan el puerto 5060 (SIP)

Ahora en la pestaña **Channels** (aquí también la imagen es referencial):

Toda la parte de *Phone Number Settings* la dejamos en blanco

### Channel Specific Setting

DTMF Methods(1-7): ch1-4:2;

### En Port Scheduling Schema (Voip->PSTN)

Simplemente cambia de 99 a 0

Todo lo demás por defecto

Grandstream Device Configuration						
Status	Basic Settings	Advanced Settings	FXO Lines	Channels	Dial-plan	Profile 1 Profile 2 Profile 3
<b>Phone Number Settings</b>						
	Channel(s)	SIP User ID	Authenticate ID	Authen Password	Profile ID	
1.	1	101	101		Profile 1	▼
2.	2	102	102		Profile 1	▼
3.	3	103	103		Profile 1	▼
4.	4	104	104		Profile 1	▼
<b>Call Progress Tones</b> (Syntax: ch x-y: f1=val@vol,f2=val@vol,c=on1/off1-on2/off2-on3/off3;[...])						
1. Dial Tone:	ch1-4:f1=425@-10,f2=425@-10,c=0/0;					
2. Ringback Tone:	ch1-4:f1=425@-10,f2=425@-10,c=100/400;					
3. Busy Tone:	ch1-4:f1=425@-10,f2=425@-10,c=25/25;					
4. Reorder Tone:	ch1-4:f1=425@-10,f2=425@-10,c=25/25;					
5. Confirmation Tone:	ch1-4:f1=425@-10,f2=425@-10,c=25/25;					
<b>Channel Voice Setting</b>						
1. Tx to PSTN Audio Gain(dB):	ch1-4:2;		(-12-12, default 1)			
2. Rx from PSTN Audio Gain(dB):	ch1-4:1;		(-12-12, default 0)			
3. Silence Suppression(Y/N):	ch1-4:N;		(default Yes)			
4. Echo Cancellation(Y/N):	ch1-4:Y;		(default Yes)			
<b>Channel Specific Setting</b>						
1. DTMF Methods(1-7):	ch1-4:2;		(default 1) (1:in-audio, 2:RFC2833, 3:1+2, 4:SIP Info, 5:1+4, 6:2+4, 7:1+2+4)			
2. No Key Entry Timeout(X1s):	ch1-4:4;		(1-9, default 4)			
3. Local SIP Listen Port:	ch1-4:5060++;		(default ch1-8:5060++)			
4. SRTP Mode(1-3):	ch1-4:1;		(default 1) (1:disabled, 2:enabled but not forced, 3:enabled and forced)			
<b>Port Scheduling Schema (Voip-&gt;PSTN)</b>						
1. Round-robin and/or Flexible:	rr:1-3;rr:4;		(default rr:1-8) (Syntax: rr: port_group; [...]) (Default: rr:1-8; round-robin of all ports )			
2. Prefix to Specify Port(1 stage dialing method):	99		(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)			
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>						

La pestaña de **Dial-plan** la dejamos por defecto ( {X+} )

Ahora en el **Profile 1**:

Lo importante aquí es:

**SIP Server:** la ip de tu servidor asterisk

**SIP Registration:** No

**NAT Traversal (STUN):** No

Grandstream Device Configuration			
Status	Basic Settings	Advanced Settings	FXO Lines Channels Dial-plan Profile 1 Profile 2 Profile 3
<b>Activate Profile:</b>	<input checked="" type="radio"/> Yes <input type="radio"/> No		
<b>Profile Name:</b>	<input type="text" value="Mediador"/>		(Optional, name of your profile)
<b>SIP Server:</b>	<input type="text" value="192.168.2.99"/>		(Server domain name or IP address)
<b>Outbound Proxy:</b>	<input type="text" value="192.168.2.99"/>		(Domain name or IP address if in use)
<i>Use DNS SRV:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<i>User ID is phone number:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<i>SIP Registration:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No		
<i>Unregister On Reboot:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No		
<i>Register Expiration:</i>	<input type="text" value="60"/>	(in minutes. default 1 hour, max 45 days)	
<i>SIP Registration Failure Retry Wait Time:</i>	<input type="text" value="20"/>	(in seconds. Between 1-3600, default is 20)	
<i>NAT Traversal (STUN):</i>	<input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> Yes		
<i>Proxy-Require:</i>	<input type="text"/>		
<i>Early Dial:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)		
<i>Session Expiration:</i>	<input type="text" value="180"/>	(in seconds. default 180 seconds)	
<i>Min-SE:</i>	<input type="text" value="90"/>	(in seconds. default and minimum 90 seconds)	
<i>Caller Request Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Request for timer when making outbound calls)		
<i>Callee Request Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (When caller supports timer but did not request one)		
<i>Force Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Use timer even when remote party does not support)		
<i>UAC Specify Refresher:</i>	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)		
<i>UAS Specify Refresher:</i>	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)		
<i>Force INVITE:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Always refresh with INVITE instead of UPDATE)		
<i>Enable 100rel:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No		
<i>Refer-To Uses Target Contact</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes		
<i>INVITE Ring-no-answer Timeout:</i>	<input type="text" value="40"/>	(in seconds. default 40 seconds)	
<i>Preferred Vocoder:</i> (in listed order)	choice 1:	<input type="text" value="PCMU"/>	choice 5: <input type="text" value="GSM"/>
	choice 2:	<input type="text" value="PCMA"/>	choice 6: <input type="text" value="GSM"/>
	choice 3:	<input type="text" value="G.723.1"/>	choice 7: <input type="text" value="PCMU"/>
	choice 4:	<input type="text" value="G.729A/B"/>	choice 8: <input type="text" value="PCMU"/>
<i>Special Feature:</i>	<input type="text" value="Standard"/>		
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>			
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Con eso terminamos la configuracion del GXW410x

Ahora en el freePBX

Creamos un SIP Trunk

**Outgoing Settings**

Trunk Name:

PEER Details:

```
allow=ulaw
context=from-internal
disallow=all
host=192.168.2.100 =>IP del GXW4104
insecure=port
type=peer
```

**Incoming Settings**

USER Context:

USER Details:

```
allow=ulaw
context=from-internal
disallow=all
host=192.168.2.100 =>IP del GXW4104
type=user
```

En [Outbound Routes](#)

### Edit Route

 Delete Route GXW4104

Route Name: GXW4104

Route Password:

PIN Set:

Emergency Dialing:

Intra Company Route:

Music On Hold?:

Dial Patterns

Dial patterns wizards:

Trunk Sequence

0  

Con esto debería funcionar sin problemas las llamadas salientes (anteponiendo el 0) y entrantes

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