

Segue os prints da nossa configuração atual.

Grandstream Device Configuration

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Hardware Revision: Main-- 2.2 Rev A Part Number-- 966-00002-22A

MAC Address: 00:0B:82:53:A1:5F

IP Address: 192.168.2.4

Product Model: GXW4108

Software Version: Program--1.3.4.13 Loader--1.1.3.4 Boot--1.1.3.2

System Up Time: 0 day(s) 0 hour(s) 6 minute(s)

Registered:

Phone Number 1: No

Phone Number 2: No

Phone Number 3: No

Phone Number 4: No

Phone Number 5: No

Phone Number 6: No

Phone Number 7: No

Phone Number 8: No

PSTN Lines:

Line 1: Connected, idle.

Line 2: Connected, idle.

Line 3: Connected, idle.

Line 4: Not Connected

Line 5: Not Connected

Line 6: Not Connected

Line 7: Not Connected

Line 8: Not Connected

PPPoE Link Up: disabled

detected NAT type is open Internet

Web Access: HTTP HTTPS

Web Port: (default for HTTP is 80 and HTTPS is 443)

End User Password: (purposely not displayed for security protection)

IP Address: dynamically assigned via DHCP (non-default) or PPPoE:
(will attempt PPPoE first if PPPoE setting is non-blank)

DHCP hostname (Option 12):

DHCP domain (Option 15):

DHCP vendor class ID (Option 60):

PPPoE account ID:

PPPoE account password:

PPPoE service name (option):

Preferred DNS server: . . .

statically configured (default) as:

IP Address: . . .

Subnet Mask: . . .

Default Router: . . .

DNS Server 1: . . .

DNS Server 2: . . .

Time Zone:

Allow DHCP Option 2 to override Time Zone setting:

No Yes

Daylight Savings Time: No Yes

Optional Rule:

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FXO Termination

1. Enable Current Disconnect(Y/N): (default Y=yes)
If enabled, use threshold: (default 100ms, range 5 ~ 65530 ms)
2. Enable Tone Disconnect(Y/N): (default No; If yes, use busy tone settings)
3. Enable Polarity Reversal(Y/N): (default No; Consult your carrier)
4. Enable Call Answer Supervision(Y/N): (default No; Consult your carrier)
5. Silence Timeout(X1s): (default 60s)
6. Incoming Call Ring Timeout(X1s): (2-10s, default 6s)
7. 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

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- 13 - 900 Ohm + 2.16 uF
- 14 - 600 Ohm + 1 uF
- 15 - Global complex impedance

Channel Dialing to PSTN

- 1. Wait for Dial-Tone(Y/N): (default No)
- 2. Stage Method(1/2): (default 2 - 2 stage dialing)
- 3. Min Delay Before Dialing Out: (default 500ms, range 50 ~ 65000ms)

Channel Dialing to VoIP

- 1. Unconditional Call Forward:
 - User ID: (i.e ch1-2:223;ch3:224)
 - Sip Server: @ (ch1-2:p1;ch3:p2)
 - Sip Destination Port: : (ch1-2:5060;ch2:7080)

PSTN to VOIP Caller ID Setting

- 1. Number of Rings Before Pickup: (1-50, default 4)
- 2. 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

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- 3. 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

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)

GXW4108 - Tronco configurado no Elastix.

type=friend

qualify=yes

secret=minhasenha

host=192.168.2.4 – **ip do GXW4108**

context=from-pstn

port=5060

dtmfmode=rfc2833

canreinvite=no

disallow=all

allow=ulaw

insecure=port

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Phone Number Settings

Channel(s)	SIP User ID	Authenticate ID	Authen Password	Profile ID
1. 1	GXW4108	GXW4108		Profile 1 ▼
2. 2	GXW4108	GXW4108		Profile 1 ▼
3. 3	GXW4108	GXW4108		Profile 1 ▼
4. 4	GXW4108	GXW4108		Profile 1 ▼
5. 5	GXW4108	GXW4108		Profile 1 ▼
6. 6	GXW4108	GXW4108		Profile 1 ▼
7. 7	GXW4108	GXW4108		Profile 1 ▼
8. 8	GXW4108	GXW4108		Profile 1 ▼

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:

Channel 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)

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Channel Specific Setting

- 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)
- No Key Entry Timeout(X1s): (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)

Port Scheduling Schema (Voip->PSTN)

- 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 dialing method): (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-

192.168.2.1 - IP Elastix.

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Activate Profile:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Profile Name:	<input type="text"/> (Optional, name of your profile)
SIP Server:	<input type="text" value="192.168.2.1"/> (Server domain name or IP address)
Outbound Proxy:	<input type="text" value="192.168.2.1"/> (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 checked="" type="radio"/> Yes <input 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>SIP Transport:</i>	<input checked="" type="radio"/> UDP <input type="radio"/> TCP
<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"/> (content for SIP Proxy-Require header)
<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 checked="" type="radio"/> Yes <input 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)

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<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>Accept INVITE from Proxy Only</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes
	choice 1: <input type="text" value="G.729A/B"/> choice 5: <input type="text" value="GSM"/>