

## **Grandstream HT701 Configuration Guide for FreePhoneLine**

### **Software Version: 1.0.6.1**

Official FPL guidelines you can find at: <http://support.freephoneline.ca/entries/23120323-VoIP-Unlock-Key-Credentials>

**All the settings were default on the device before configuration.**

#### **Step 1.**

Plug all the appropriate wires to your Grandstream HT701 device.

- Internet cable (Internet connection)
- Phone line (attached to the phone)
- Power

#### **Step 2.**

From the phone attached to your Grandstream HT701 device:

- Dial '\*\*\*'
- Then dial '02'

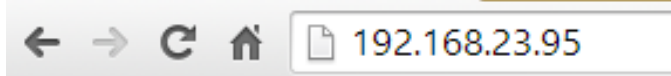
Write down your IP address.

#### **Step 3.**

On the PC/Mac connected to the same network as your Grandstream HT701 device, go to your browser (Internet Explorer; Chrome; Firefox; Opera; Safari, etc.).

In the URL bar put your IP address from Step 2.

*[Note: I am using 192.168.23.95 for my IP, your IP address might be different!]*



Web-based utility will appear.

Now you have to login.

Default **Password: admin**

A screenshot of the 'Grandstream Device Configuration' login page. The page has a yellow header with the title 'Grandstream Device Configuration'. Below the header is a yellow section containing a 'Password' label, a text input field with four dots, and a 'Login' button. A red oval highlights the password input field, and a red arrow points from the word 'admin' to it. Another red oval highlights the 'Login' button. The footer is a blue bar with the text 'All Rights Reserved Grandstream Networks, Inc. 2006-2014'.

## Step 4.

Go to 'BASIC SETTINGS' tab and fill in the following settings:

**Time Zone:** *Using self-defined Time Zone*

**Self-Defined Time Zone:** [Choose appropriate time zone, depending where you are located]

<u>Your time zone</u>	<u>What value to fill in</u>
EST	MTZ+5MDT+4,M3.2.0,M11.1.0
PST	MTZ+8MDT+7,M3.2.0,M11.1.0
MST	MTZ+7MDT+6,M3.2.0,M11.1.0
CST	MTZ+6MDT+5,M3.2.0,M11.1.0

Then click 'Apply' button at the bottom

**Grandstream Device Configuration**

**STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT**

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

Web Port:  (default for HTTP is 80)

Telnet Server: ☐ No ☒ Yes

IP Address: ☒ dynamically assigned via DHCP

DHCP hostname:  (optional)

DHCP vendor class ID:  (optional)

☐ use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

1st Preferred DNS server:

2nd Preferred DNS server:

3rd Preferred DNS server:

4th Preferred DNS server:

☐ statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone:  ▼

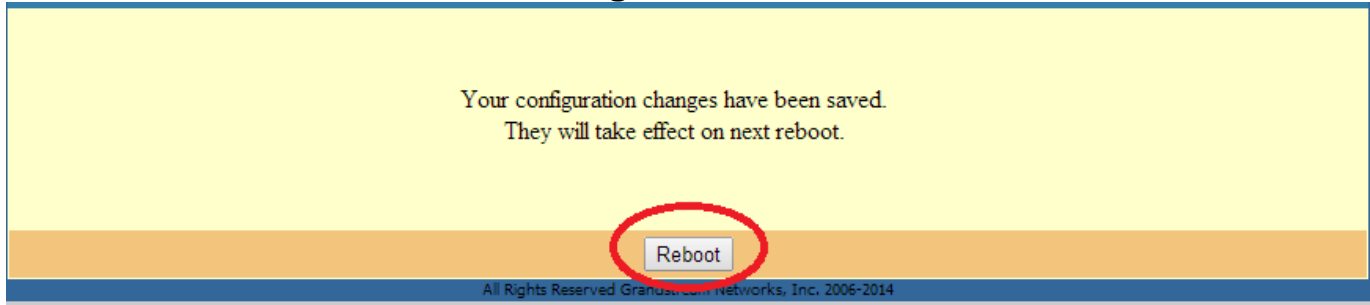
Self-Defined Time Zone:  (For example: MTZ+6MDT+5,M4.1.0,M11.1.0)

Allow DHCP server to set Time Zone: ☒ No ☐ Yes

Language:  ▼

Reset Type:

After you click 'Apply' button, device should ask you to reboot. If so – click on 'Reboot' button to apply the changes you have made. If it doesn't – go back to 'BASIC SETTINGS' tab and click 'Reboot' button at the bottom. Wait while device is rebooting.



#### Step 5.

Go to 'FXS PORT' tab and change the following settings:

**Primary SIP Server:** *voip.freephoneline.ca* **OR** *voip2.freephoneline.ca* **[Note: For ROGERS Internet provider customers use *voip4.freephoneline.ca:6060*]**

**NAT Traversal:** *Keep-Alive*

**SIP User ID:** *[Your FPL number 1xxxxxxxxx]*

**Authenticate Password:** *[Your SIP password]*

**Name:** *[Your first and last name]*

**Outgoing Call without Registration:** *No*

**Use Random SIP Port:** *Yes*

**Use Random RTP Port:** *Yes*

Grandstream Device Configuration	
STATUS	BASIC SETTINGS
<div> <div>Account Active:</div> <div> <input type="radio"/> No           <input checked="" type="radio"/> Yes         </div> </div>	
Primary SIP Server:	<div>voip.freephoneline.ca</div> <div>(e.g., sip.mycompany.com, or IP address)</div>
Failover SIP Server:	<div></div> <div>(Optional, used when primary server no response)</div>
Prefer Primary SIP Server:	<div> <input checked="" type="radio"/> No           <input type="radio"/> Yes         </div> <div>(yes - will register to Primary Server if Failover registration expires)</div>
Outbound Proxy:	<div></div> <div>(e.g., proxy.myprovider.com, or IP address, if any)</div>
Allow DHCP Option 120( override SIP server ):	<div> <input checked="" type="radio"/> No           <input type="radio"/> Yes         </div>
SIP Transport:	<div> <input checked="" type="radio"/> UDP           <input type="radio"/> TCP           <input type="radio"/> TLS         </div> <div>(default is UDP)</div>
NAT Traversal:	<div> <input type="radio"/> No           <input checked="" type="radio"/> Keep-Alive           <input type="radio"/> STUN           <input type="radio"/> UPnP         </div>
SIP User ID:	<div>1xxxxxxxxx</div> <div>(the user part of an SIP address)</div>
Authenticate ID:	<div></div> <div>(can be identical to or different from SIP User ID)</div>
Authenticate Password:	<div>.....</div> <div>(purposely not displayed for security protection)</div>
Name:	<div>Firstname Lastname</div> <div>(optional, e.g., John Doe)</div>
DNS Mode:	<div> <input checked="" type="radio"/> A Record           <input type="radio"/> SRV           <input type="radio"/> NAPTR/SRV         </div>
Tel URI:	<div>Disabled</div>
SIP Registration:	<div> <input type="radio"/> No           <input checked="" type="radio"/> Yes         </div>
Unregister On Reboot:	<div> <input checked="" type="radio"/> No           <input type="radio"/> Yes         </div>
Outgoing Call without Registration:	<div> <input checked="" type="radio"/> No           <input type="radio"/> Yes         </div>
Register Expiration:	<div>60</div> <div>(in minutes. default 1 hour, max 45 days)</div>
Reregister before Expiration:	<div>0</div> <div>(in seconds. Default 0 second)</div>
SIP Registration Failure Retry Wait Time:	<div>20</div> <div>(in seconds. Between 1-3600, default is 20)</div>
Layer 3 QoS:	<div>24</div> <div>SIP DSCP (Diff-Serv value in decimal, default 24)</div>
	<div>46</div> <div>RTP DSCP (Diff-Serv value in decimal, default 46)</div>
Local SIP port:	<div>5060</div> <div>(default is 5060 for UDP and TCP; 5061 for TLS)</div>
Local RTP port:	<div>5004</div> <div>(even number between 1024-65535, default 5004)</div>
Use Random SIP Port:	<div> <input type="radio"/> No           <input checked="" type="radio"/> Yes         </div>
Use Random RTP Port:	<div> <input type="radio"/> No           <input checked="" type="radio"/> Yes         </div>
Refer-To Use Target Contact:	<div> <input checked="" type="radio"/> No           <input type="radio"/> Yes         </div>

**Transfer on Conference Hangup:** Yes

**Allow Incoming SIP Messages from SIP Proxy Only:** Yes

**Transfer on Conference Hangup:** Yes

**Preferred DTMF method: (in listing order)**

<b>Priority 1:</b>	<i>RFC2833</i>
<b>Priority 2:</b>	<i>In-audio</i>
<b>Priority 3:</b>	<i>SIP INFO</i>

**Enable Call Features:** No

Transfer on Conference Hangup: ☐ No ☒ Yes

Disable Bellcore Style 3-Way Conference: ☒ No ☐ Yes (Using star code \*23 for 3-way conference)

Remove OBP from Route Header: ☒ No ☐ Yes

Support SIP Instance ID: ☐ No ☒ Yes

Validate Incoming SIP Message: ☒ No ☐ Yes

Check SIP User ID for incoming INVITE: ☒ No ☐ Yes (no direct IP calling if Yes)

Authenticate incoming INVITE: ☒ No ☐ Yes

Allow Incoming SIP Messages from SIP Proxy Only: ☐ No ☒ Yes (no direct IP calling if Yes)

Use Privacy Header: ☒ Default ☐ No ☐ Yes

Use P-Preferred-Identity Header: ☒ Default ☐ No ☐ Yes

SIP T1 Timeout: 0.5 sec ▼

SIP T2 Interval: 4 sec ▼

DTMF Payload Type: 101

Preferred DTMF method: (in listed order)  
 Priority 1: RFC2833 ▼  
 Priority 2: In-audio ▼  
 Priority 3: SIP INFO ▼

Disable DTMF Negotiation: ☒ No (negotiate with peer) ☐ Yes (use above DTMF order without negotiation)

Send Hook Flash Event: ☒ No ☐ Yes (Hook Flash will be sent as a DTMF event if set to Yes)

Enable Call Features: ☒ No ☐ Yes (if Yes, call features using star codes will be supported locally)

Offhook Auto-Dial:  (User ID/extension to dial automatically when offhook)

Offhook Auto-Dial Delay: 0 (0-60 seconds, default is 0)

Proxy-Require:

Use NAT IP:  (used in SIP/SDP message if specified)

Use SIP User-Agent Header:

Distinctive Ring Tone: Ring Tone 1 ▼ used if incoming caller ID is   
 Ring Tone 1 ▼ used if incoming caller ID is   
 Ring Tone 1 ▼ used if incoming caller ID is

Disable Call-Waiting: ☒ No ☐ Yes

Disable Call-Waiting Caller ID: ☒ No ☐ Yes

Disable Call-Waiting Tone: ☒ No ☐ Yes

Disable Receiver Offhook Tone: ☒ No ☐ Yes (ROH tone will not be played after offhook for 60 seconds)

Disable Reminder Ring for On-Hold Call: ☒ No ☐ Yes

**No Key Entry Timeout: 4**

**Preferred Vocoder: (in listed order):**

<b>Choice 1:</b>	PCMU
<b>Choice 2:</b>	G729
<b>Choice 3:</b>	PCMU
<b>Choice 4:</b>	PCMU
<b>Choice 5:</b>	PCMU
<b>Choice 6:</b>	PCMU

Ring Timeout:  (10-300, default is 60 seconds)  
 Delayed Call Forward Wait Time:  (Allowed range 1-120, in seconds.)  
 No Key Entry Timeout:  (in seconds, default is 4 seconds)  
 Early Dial: ☒ No ☐ Yes (use "Yes" only if proxy supports 484 response)  
 Dial Plan Prefix:  (this prefix string is added to each dialed number)  
 Use # as Dial Key: ☐ No ☒ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)  
 Dial Plan:   
 SUBSCRIBE for MWI: ☒ No, do not send SUBSCRIBE for Message Waiting Indication  
☐ Yes, send periodical SUBSCRIBE for Message Waiting Indication  
 Send Anonymous: ☒ No ☐ Yes (caller ID will be blocked if set to Yes)  
 Anonymous Call Rejection: ☒ No ☐ Yes  
 Special Feature:   
 Session Expiration:  (in seconds. default 180 seconds)  
 Min-SE:  (in seconds. default and minimum 90 seconds)  
 Caller Request Timer: ☒ No ☐ Yes (Request for timer when making outbound calls)  
 Callee Request Timer: ☒ No ☐ Yes (When caller supports timer but did not request one)  
 Force Timer: ☒ No ☐ Yes (Use timer even when remote party does not support)  
 UAC Specify Refresher: ☐ UAC ☐ UAS ☒ Omit (Recommended)  
 UAS Specify Refresher: ☒ UAC ☐ UAS (When UAC did not specify refresher tag)  
 Force INVITE: ☒ No ☐ Yes (Always refresh with INVITE instead of UPDATE)  
 Enable 100rel: ☒ No ☐ Yes  
 Add Auth Header On Initial REGISTER: ☒ No ☐ Yes  
 Use First Matching Vocoder in 200OK SDP: ☒ No ☐ Yes  
 Preferred Vocoder: (in listed order)  
 choice 1:   
 choice 2:   
 choice 3:   
 choice 4:   
 choice 5:   
 choice 6:   
 Voice Frames per TX:   
 G723 Rate: ☒ 6.3kbps encoding rate ☐ 5.3kbps encoding rate  
 iLBC Frame Size: ☒ 20ms ☐ 30ms  
 iLBC Payload Type:  (between 96 and 127, default is 97)

### Then click 'Apply' button at the bottom

Fax Mode: ☒ T.38 ☐ Pass-Through  
 Re-INVITE After Fax Tone Detected: ☒ Enabled ☐ Disabled  
 Jitter Buffer Type: ☐ Fixed ☒ Adaptive  
 Jitter Buffer Length: ☐ Low ☒ Medium ☐ High  
 SRTP Mode: ☒ Disabled ☐ Enabled but not forced ☐ Enabled and forced

SLIC Setting:   
 Caller ID Scheme:   
 DTMF Caller ID: Start Tone  Stop Tone   
 Polarity Reversal: ☒ No ☐ Yes (reverse polarity upon call establishment and termination)  
 Loop Current Disconnect: ☒ No ☐ Yes (loop current disconnect upon call termination)  
 Loop Current Disconnect Duration:  (100 - 10000 milliseconds. Default 200 milliseconds)  
 Enable Hook Flash: ☐ No ☒ Yes  
 Hook Flash Timing: In 40-2000 milliseconds range, minimum:  maximum:   
 On Hook Timing:  (In 40-2000 milliseconds range, default is 400)  
 Gain: TX  RX   
 Disable Line Echo Canceller (LEC): ☒ No ☐ Yes  
 Outgoing Call Duration Limit:  (0-180 minutes, default is 0 (No Limit) )

**Ring Tones** (Syntax: c=on1/off1-on2/off2-on3/off3;)  
 Ring Tone 1:   
 Ring Tone 2:   
 Ring Tone 3:   
 Ring Tone 4:   
 Ring Tone 5:   
 Ring Tone 6:   
 Ring Tone 7:   
 Ring Tone 8:   
 Ring Tone 9:   
 Ring Tone 10:

Update **Apply** Cancel Reboot

**The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT' tab and click 'Reboot' at the bottom. Wait while device is rebooting.**

Enjoy your free phone line! ☺