

## Grandstream HT502 Configuration Guide for FreePhoneLine Software Version: 1.0.12.1

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 HT502 device.

- Internet cable *[Note: Connect Ethernet cable to WAN port of your Grandstream HT502 device and another end to your Internet rack/router]*
- Phone line (attached to the phone) *[Note: Use Phone1 port]*
- Power

### Step 2.

By default settings Grandstream HT502 WAN access is disabled.

You have to enable this setting before you access web-based utility.

From the phone attached to Grandstream HT502 device:

- Dial '\*\*\*' (to access IVR menu)
- Then dial '12' (to check your WAN Port Web Access status)
- Then press '9' (to toggle between enable/disable) *[Note: Your WAN Port Web Access should be enabled]*
- Hang up the phone

Now, you have to figure out IP address to access web-based utility. From your phone dial:

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

Write down your IP address.

### Step 3.

On the PC/Mac connected to the same network as your Grandstream HT502 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**

**Grandstream Device Configuration**

**Password**

.....

admin

Login

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## Step 4.

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

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

**1** **Grandstream Device Configuration**

**STATUS** **BASIC SETTINGS** **ADVANCED SETTINGS** **FXS PORT1** **FXS PORT2**

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:

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)

Language:  ▼

**2**

GMT-10:00 (US Hawaiian Time)  
 GMT-09:00 (US Alaska Time)  
 GMT-08:00 (US Pacific Time, Los Angeles)  
 GMT-08:00 (Baja California)  
 GMT-07:00 (US Mountain Time, Denver)  
 GMT-07:00 (Mountain Time (Arizona, no DST))  
 GMT-07:00 (Chihuahua, La Paz, Mazatlan)  
 GMT-06:00 (Central Time)  
 GMT-06:00 (Central America)  
 GMT-06:00 (Guadalajara, Mexico City, Monterrey)  
**GMT-05:00 (Eastern Time)**  
 GMT-05:00 (Eastern Time without daylight saving)  
 GMT-04:30 (Caracas)  
 GMT-04:00 (Atlantic Time)  
 GMT-04:00 (Atlantic Time (New Brunswick))  
 GMT-03:30 (Newfoundland Time)  
 GMT-03:00 (Greenland)  
 GMT-03:00 (Brazil, Sao Paulo)  
 GMT-02:00 (Argentina)

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

**3**

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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.

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

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## Step 5.

Go to 'FXS PORT 1' 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 Port:** *Yes*

**Transfer on Conference Hangup:** *Yes*

**Grandstream Device Configuration**

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT1	FXS PORT2
<p>Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes</p>				
<p>Primary SIP Server: <span style="border: 2px solid red; padding: 2px;">voip.freephoneline.ca</span> (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 type="radio"/> No <span style="border: 2px solid red; padding: 2px;"><input checked="" type="radio"/> Keep-Alive</span> <input type="radio"/> STUN <input type="radio"/> UPnP</p>				
<p>SIP User ID: <span style="border: 2px solid red; padding: 2px;">1xxxxxxxxx</span> (the user part of an SIP address)</p>				
<p>Authenticate ID: <input type="text"/> (can be identical to or different from SIP User ID)</p>				
<p>Authenticate Password: <input type="password" value="....."/> (purposely not displayed for security protection)</p>				
<p>Name: <span style="border: 2px solid red; padding: 2px;">Firstname Lastname</span> (optional, e.g., John Doe)</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>SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes</p>				
<p>Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes</p>				
<p>Outgoing Call without Registration: <span style="border: 2px solid red; padding: 2px;"><input checked="" type="radio"/> No</span> <input type="radio"/> Yes</p>				
<p>Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)</p>				
<p>Reregister before Expiration: <input type="text" value="0"/> (in seconds. Default 0 second)</p>				
<p>SIP Registration Failure Retry Wait Time: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)</p>				
<p>Local SIP Port: <input type="text" value="5060"/> (default is 5060 for UDP and TCP; 5061 for TLS)</p>				
<p>Local RTP Port: <input type="text" value="5004"/> (1024-65535, default 5004)</p>				
<p>Use Random Port: <input type="radio"/> No <span style="border: 2px solid red; padding: 2px;"><input checked="" type="radio"/> Yes</span></p>				
<p>Refer-To Use Target Contact: <input checked="" type="radio"/> No <input type="radio"/> Yes</p>				
<p>Transfer on Conference Hangup: <span style="border: 2px solid red; padding: 2px;"><input checked="" type="radio"/> Yes</span> <input type="radio"/> No</p>				
<p>Disable Bellcore Style 3-Way Conference: <input checked="" type="radio"/> No <input type="radio"/> Yes (Using star code *23 for 3-way conference)</p>				

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

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

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

**Enable Call Features:** No

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)

**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

Preferred Vocoder: (in listed order)

choice 1: PCMU ▼

choice 2: G729 ▼

choice 3: PCMU ▼

choice 4: PCMU ▼

choice 5: PCMU ▼

choice 6: PCMU ▼

choice 7: PCMU ▼

choice 8: PCMU ▼

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

Update  Cancel Reboot

**The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT 1' tab and click 'Reboot' at the bottom. Wait while device is rebooting.**

Step 6.

Go to 'FXS PORT 2' tab and change the following settings:

**Account Active:** *No*

The screenshot shows the 'Grandstream Device Configuration' interface. At the top, there are tabs for 'STATUS', 'BASIC SETTINGS', 'ADVANCED SETTINGS', 'FXS PORT1', and 'FXS PORT2'. The 'FXS PORT2' tab is selected and highlighted with a red box and a red arrow labeled '1'. Below the tabs, the 'Account Active' setting is shown with two radio buttons: 'No' (selected) and 'Yes'. The 'No' radio button is highlighted with a red box and a red arrow labeled '2'. At the bottom of the interface, there are four buttons: 'Update', 'Apply', 'Cancel', and 'Reboot'. The 'Apply' button is highlighted with a red box and a red arrow labeled '3'. The footer of the interface reads 'All Rights Reserved Grandstream Networks, Inc. 2006-2014'.

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

**The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT 2' tab and click 'Reboot' at the bottom. Wait while device is rebooting.**

Enjoy your free phone line! 😊