

## **Grandstream HT702 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 HT702 device.

- Internet cable (Internet connection)
- Phone line (attached to the phone) [*Note: Use 'Port 1'*]
- Power

### **Step 2.**

From the phone attached to your Grandstream HT702 device:

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

Write down your IP address.

### **Step 3.**

On the PC/Mac connected to the same network as your Grandstream HT702 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 web interface. The page has a blue header with the text 'Grandstream Device Configuration'. Below the header is a yellow section containing a login form. The form has a label 'Password' followed by a text input field containing five dots. A red oval highlights the input field, and a red arrow points from the word 'admin' to it. Below the input field is a 'Login' button, also highlighted with a red oval. At the bottom of the page is a blue footer 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:** [Choose appropriate time zone, depending where you are located]

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

**Grandstream Device Configuration**

**1** 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:

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:

**2** Time Zone: GMT-05:00 (Eastern Time)

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:

**3**

All Rights Reserved Grandstream Networks, Inc. 2006-2014

**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.

All Rights Reserved Grandstream Networks, Inc. 2006-2014

## 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 SIP Port:** *Yes*

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

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT1
FXS PORT2

Account Active:  No  Yes

Primary SIP Server: voip.freephoneline.ca (e.g., sip.mycompany.com, or IP address)

Failover SIP Server:  (Optional, used when primary server no response)

Prefer Primary SIP Server:  No  Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy:  (e.g., proxy.myprovider.com, or IP address, if any)

Allow DHCP Option 120( override SIP server ):  No  Yes

SIP Transport:  UDP  TCP  TLS (default is UDP)

NAT Traversal:  No  Keep-Alive  STUN  UPnP

SIP User ID: 1xxxxxxxxx (the user part of an SIP address)

Authenticate ID:  (can be identical to or different from SIP User ID)

Authenticate Password: ..... (purposely not displayed for security protection)

Name: FirstName LastName (optional, e.g., John Doe)

DNS Mode:  A Record  SRV  NAPTR/SRV

Tel URI: Disabled

SIP Registration:  No  Yes

Unregister On Reboot:  No  Yes

Outgoing Call without Registration:  No  Yes

Register Expiration: 60 (in minutes. default 1 hour, max 45 days)

Reregister before Expiration: 0 (in seconds. Default 0 second)

SIP Registration Failure Retry Wait Time: 20 (in seconds. Between 1-3600, default is 20)

Layer 3 QoS: 24 SIP DSCP (Diff-Serv value in decimal, default 24)

46 RTP DSCP (Diff-Serv value in decimal, default 46)

Local SIP port: 5060 (default is 5060 for UDP and TCP; 5061 for TLS)

Local RTP port: 5004 (even number between 1024-65535, default 5004)

Use Random SIP Port:  No  Yes

Use Random RTP Port:  No  Yes

Refer-To Use Target Contact:  No  Yes

**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)

2 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:

**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'. A red box and arrow labeled '2' point to the 'No' radio button. 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! 😊