

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.

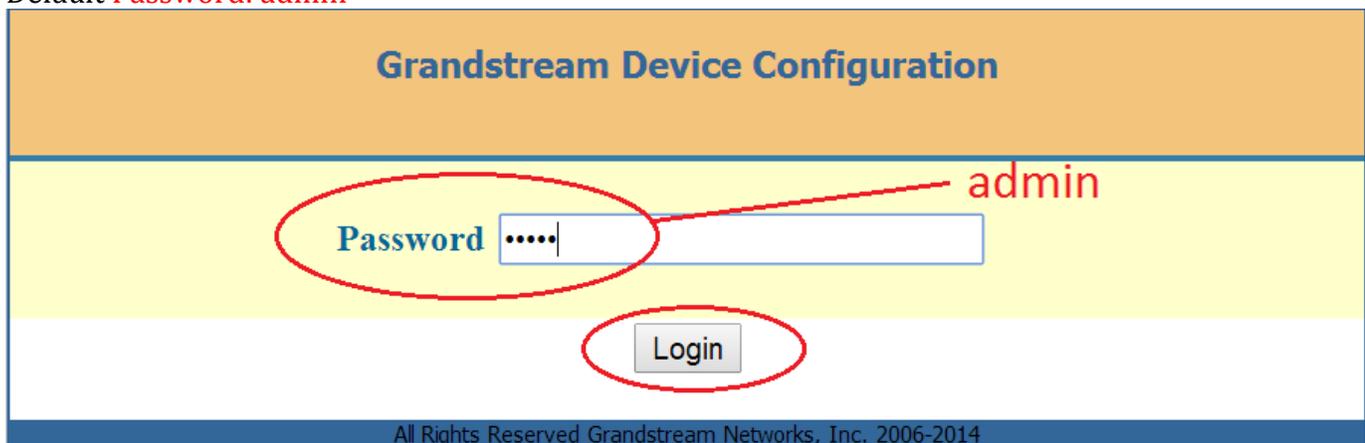
[Note: I am using 192.168.23.95 for my IP, your IP address might be different!]



Web-based utility will appear.

Now you have to login.

Default **Password: admin**

A screenshot of the Grandstream Device Configuration web interface. The page has a yellow header with the title 'Grandstream Device Configuration'. Below the header is a yellow background area containing a login form. The form has a 'Password' label followed by a text input field with four dots inside. 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: *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

1 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

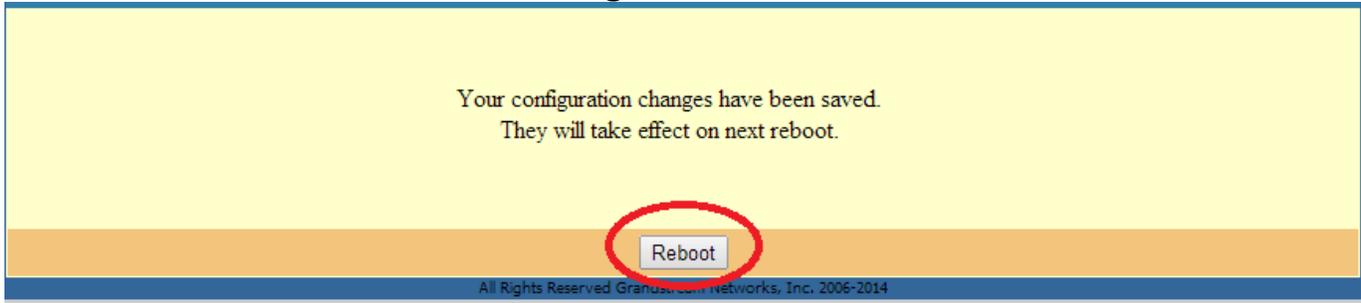
Language: ▼

Reset Type: ▼

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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.



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

Priority 1:	<i>RFC2833</i>
Priority 2:	<i>In-audio</i>
Priority 3:	<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):**

Choice 1:	PCMU
Choice 2:	G729
Choice 3:	PCMU
Choice 4:	PCMU
Choice 5:	PCMU
Choice 6:	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' tab and click 'Reboot' at the bottom. Wait while device is rebooting.

Enjoy your free phone line! 😊