

Grandstream HT503 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 HT503 device.

- Internet cable *[Note: Connect Ethernet cable to WAN port of your Grandstream HT503 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 HT503 WAN access is disabled.

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

From the phone attached to Grandstream HT503 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 HT503 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

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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]

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Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **FXS PORT** **FXO 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:

Preferred DNS server:

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 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)
 GMT-02:00 (Mid-Atlantic)

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Time Zone:

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

Language:

Then click 'Apply' button at the bottom

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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' tab and change the following settings:

Primary SIP Server: *voip.freephoneline.ca* OR *voip2.freephoneline.ca* [**Note:** **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 PORT
FXO PORT

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)

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 Use Configured IP

Primary IP:

Backup IP1:

Backup IP2:

Tel URI: Disabled

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

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

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

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

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

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

Use Random Port: No Yes

Refer-To Use Target Contact: No Yes

Transfer on Conference Hangup: No Yes

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

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Allow Incoming SIP Messages from SIP Proxy Only: Yes

Preferred DTMF method: (in listing order)

Priority 1:	RFC2833
Priority 2:	In-audio
Priority 3:	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):

Choice 1:	PCMU
Choice 2:	G729
Choice 3:	PCMU
Choice 4:	PCMU
Choice 5:	PCMU
Choice 6:	PCMU
Choice 7:	PCMU
Choice 8:	PCMU

Preferred Vocoder: (in listed order)

choice 1: PCMU ▼

choice 2: G729 ▼

choice 3: PCMU ▼

choice 4: PCMU ▼

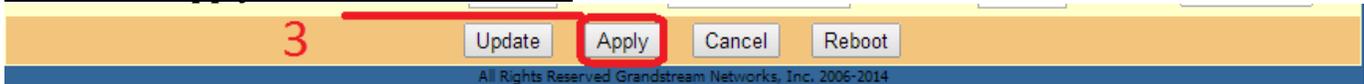
choice 5: PCMU ▼

choice 6: PCMU ▼

choice 7: PCMU ▼

choice 8: PCMU ▼

Then click 'Apply' button at the bottom

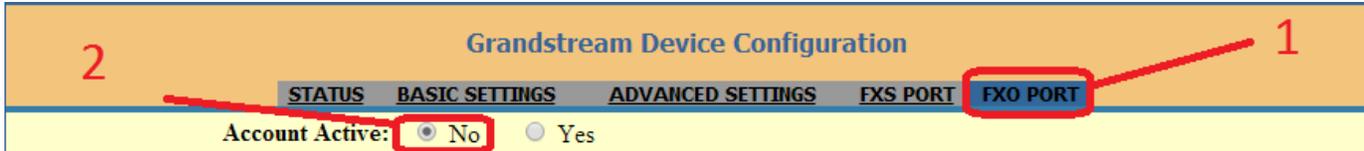


The device might not ask you for reboot. In this case – after you applied changes, go back to 'FXS PORT' tab and click 'Reboot' at the bottom. Wait while device is rebooting.

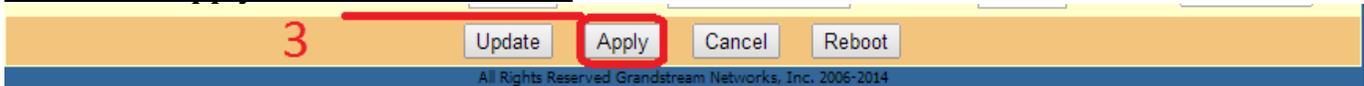
Step 6.

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

Account Active: No



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 'FXO PORT' tab and click 'Reboot' at the bottom. Wait while device is rebooting.

Enjoy your free phone line! 😊