Grandstream HT701 ATA Configuration Guide for Freephoneline

Software Version: 1.0.8.2
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.

Get Started and Signed In

1. Plug all the appropriate wires to your Grandstream HT701 device.

- Internet cable (Internet connection), typically from the ATA to a router
- Phone line (attached to the phone)
- Power AC adapter

2. From the phone attached to your Grandstream HT701 device:

- Dial '***'
- Then dial '02'
- Write down your IP address.

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.).

4. In the URL bar, enter the IP address that you wrote down.

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

The web interface for the HT701 appears.

5. Sign in using the default password **admin**

Configure Basic Settings

1. Go to the **BASIC SETTINGS** tab and fill in the following settings:

- **Time Zone**: Using self-defined Time Zone
- **Self-Defined Time Zone**: [Choose the appropriate time zone, based on where you are located]
  - **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

2. Click **Apply** at the bottom.
After you click **Apply**, the device should ask you to reboot.

- If so, click **Reboot** to apply the changes you have made.
- If it doesn't,
  - Go back to the **BASIC SETTINGS** tab and click **Reboot** at the bottom.
  - Wait while the device reboots.

Configure FXS Port Settings

Before you begin:

- Go to https://www.freephoneline.ca/showSipSettings and sign in.
- Write down your SIP username and password. You will enter these for the **SIP User ID** and **Authenticate Password** values.

1. Go to **FXS PORT** tab and change the following settings:

- For **Primary SIP Server**, enter one of the following:
  - voip.freephoneline.ca
  - voip2.freephoneline.ca
  - voip4.freephoneline.ca:6060 for Rogers ISP or if your router has faulty SIP ALG settings.
- **NAT Traversal**: Keep-Alive
- **SIP User ID**: [Your FPL number 1xxxxxxxxxx]
- **Authenticate Password**: [Your SIP password]
- **Name**: [Your first and last name]
- **Outgoing Call without Registration**: No
- **SIP Registration Failure Retry Wait Time**: 120
- **Enable SIP Options Keep Alive**: Yes
- **Use Random SIP Port**: Yes
- **Use Random RTP Port**: 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**: RFC2833
  - **Priority 2**: In-audio
  - **Priority 3**: SIP INFO
- **Enable Call Features**: No
- **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

2. Click **Apply** at the bottom.

You might not be prompted to reboot. In this case, after you applied changes:

- Go back to the **FXS Port** tab and click **Reboot** at the bottom.
- Wait while the device reboots.
- Ensure that all lights are green on the ATA box.

Enjoy your Freephoneline!

Troubleshoot Common Issues

Credit goes to Lipton for this information (and Mango for item #5).

1. Ensure whatever modem/router combo your ISP gave you is in bridge mode.
2. If you can't get the router combo from your ISP in bridge mode, then either see if you can disable SIP ALG in it--or for your Primary SIP Server in your ATA, use voip4.freephoneline.ca:6060
3. Submit a support ticket requesting a "forced registration" for your account: https://support.fongo.com/hc/en-us/requests/new
4. Enable QoS in your router for your ATA. Give your ATA highest priority and all other devices on your LAN, lower priority: https://www.asus.com/support/FAQ/1008717/
5. You can try Asuswrt-Merlin third party firmware: http://asuswrt.lostrealm.ca/

   Navigate to Tools-->Other settings in your router.

   The following conditions should be met:

   UDP Unreplied Timeout (in your router) < NAT Keep-alive Interval (in your ATA) < UDP Assured Timeout (in your router) < SIP Registration Failure Retry Wait Time (in your ATA)

   "<" means less than

   If you properly configured your ATA for FPL, NAT Keep-alive interval is 20, and SIP Registration Failure Retry Wait Time is 120.

   So, after installing Merlin firmware, navigate to Tools-->Other Settings.

   Change UDP Timeout: Unreplied to 10 and UDP Timeout: Assured to 100.

Thanks to Mango, many of us now understand that in order for ATAs to remain registered and working properly with a VoIP SIP provider like Freephoneline, in particular after power failures, the following conditions must be met:

UDP Unreplied Timeout (in your router) < NAT Keep-alive Interval (in your ATA; for Obihai ATAs this is X_KeepAliveExpires) < UDP Assured Timeout (in your router) < SIP Registration Failure Retry Wait Time (or RegisterRetryInterval in Obihai ATAs)

“<“ means less than.

When a modem leases a new IP address, a problem can arise where prior associations using the old IP address are maintained in the router. When the ATA attempts to communicate using the old IP address, the response is unreplied, and then if the UDP Unreplied timeout is greater than the Keep Alive Interval (and UDP Unreplied timeout is often set to 30 by default in consumer routers) a problem arises where the corrupted connection persists. If UDP Unreplied timeout is, for example, 10, and the NAT Keep Alive Interval is 20, then the corrupted connection will timeout or close. A new connection will be created, and everything will work fine.

Another problem can occur when the Keep-Alive interval is greater than UDP Assured Timeout (often 180 by default in consumer routers): the NAT hole will close due to the ATA not communicating frequently enough with the SIP server. In turn, incoming calls may, intermittently, not reach the ATA.

Getting access to both UDP Unreplied Timeout and UDP Assured Timeout settings in consumer routers may be difficult, if not impossible. Asuswrt-Merlin, third party firmware for Asus routers, does offer easy access to these two settings, which are found under Tools-->Other settings. In part, for this reason, I tend to use Asus routers. However, my understanding is that third party Tomato firmware has these two settings as well. So if your router supports Tomato firmware, that may be another option.

The keep alive interval for FPL is 20. The SIP Registration Failure Retry Wait Time is 120. I use 10 for UDP Unreplied Timeout and 100 for UDP Assured Timeout.

6. You shouldn't have to do the following at all with Asus routers, and port fowarding is a security risk, so only do the following if all else fails first. You could port forward from your router to your ATA, the RTP ports your ATA uses (these are UDP ports). But you shouldn't have to do that at all.